

SWITCHED SERVICES NETWORK

ELECTRONIC SWITCHED NETWORK,

GENERAL DESCRIPTION

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Reason for Reissue: This practice is reissued to incorporate changes to the **NARS/BARS** U-digit translation added in Meridian SL-1 Generic X11 Release 8 as well as to make other minor corrections. Revisions are shown by arrowheads in the margins.

1. INTRODUCTION

ELECTRONIC
SWITCHED NETWORK

1.01 The Electronic Switched Network (**ESN**) is a private communications network intended for use by large business customers with distributed operating locations.

1.02 This practice introduces the reader to the concepts of the **ESN**, with emphasis on the Meridian SL-1 switch element. The information contained in this practice will enable the reader to gain an appreciation of the enhanced quality of communications service and reduced communications costs provided by the ESN.

SWITCH ELEMENT

1.03 A prime element of the ESN is the Meridian **SL-1*** Integrated+ Services Network which, when arranged as a combined Private Branch Exchange (**PBX**) and network switch, is termed a Meridian SL-1 ESN Node. The Meridian SL-1 ESN Nodes are strategically located (Fig. 1-1) to concentrate on on-network traffic and access to off-network facilities efficiently and economically. An ESN can comprise a single ESN Node serving a few locations in a metropolitan area, or multiple ESN nodes serving up to 999 locations in a widely-dispersed national or North American network.

1.04 The Meridian SL-1 ESN Nodes function to direct calls from a switch in one geographical location to a switch in any other geographical location in a cost-efficient and easy-to-use manner by:

- eliminating long, complex dialing plans, and replacing them with an abbreviated Uniform Dialing Plan (**UDP**) common to all switches which are part of the ESN,
- providing a means of controlling the number and type of trunks that are available to each network caller, and a method of controlling the time of day that access to a trunk (or group of trunks) is allowed,
- | selecting automatically the least-cost trunk route available to complete a call between network switches,
- | providing uniform network access to stations served directly by the ESN Node, and stations served at other switches (Meridian SL-1 ESN Mains, Conventional Mains) connected via tie trunks to an ESN Node as shown in Fig. 1-1,
- providing the call originator with the option to either accept or refuse that a call be completed over an expensive trunk route, if no cheaper trunks are currently available,
- providing **optional** queuing features which enable the call originator (when all trunks are currently busy) to either remain off-hook until a trunk becomes idle, or hang up and receive a callback from the system when a trunk becomes idle.

1.05 To accomplish the efficient call-handling operations in an ESN, each Meridian SL-1 ESN Node utilizes some or all of the following unique ESN software features which are described in this practice:

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- | Network Class of Service (**NCOS**)
- | Network Alternate Route Selection (**NARS**)
(or Basic Alternate Route Selection (BARS))
- Network Signaling (**NSIG**)
- Network Traffic Measurements (**NTRF**)
- Off-Hook Queuing (**OHQ**)
- Call-Back Queuing (**CBQ**)
- | Coordinated Call-Back Queuing (**CCBQ**)
- | Call-Back Queuing to Conventional Mains (CBQCM)
- Free Calling Area Screening (**FCAS**)
- | Coordinated Dialing Plan (**CDP**)
- Network Authorization Codes (**NAUT**)
(or Basic Authorization Codes (**BAUT**)).

SWITCH DEFINITIONS

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1.06 In the context of this practice, the following definitions of the various switch types that can be part of an ESN network are applied.

- (a) Meridian SL-1 ESN Node. A Meridian SL-1 switch that is equipped with BARS or NARS features.
- (b) Meridian SL-1 ESN Main. A Meridian SL-1 switch that is equipped with features as identified in Table 1-A and that is connected via tie trunks to a single Meridian SL-1 ESN Node. A Meridian SL-1 ESN Main can also be equipped with the Basic Alternate Route Selection (BARS) feature to provide alternate route selection capabilities for calls placed to satellite switches that are homed on the ESN Main. (BARS is described in 553-2751-100.)
- (c) Conventional Main. A switch (Meridian SL-1 or any other type) that is connected to a Meridian SL-1 ESN Node and equipped with none of the features listed in Table 1-A. Other switch types can include Step-by-Step (**SXS**), Electronic Tie Network (**ETN**), DIMENSION, etc.

Table 1-A
 FEATURE PACKAGE **REQUIREMENTS FOR**
 MERIDIAN SL-1 ESN NODES AND MERIDIAN **SL-1** ESN MAINS

	ESN NODE	ESN MAIN
Network Alternate Route Sel or Basic Alternate Route Sel	X	X(opt)
Off -Hook Queuing	X(opt)	
Call-Back Queuing	X	
Coordinated Call-Back Queuing	X(opt)	X(opt)
Call-Back Queuing to Conventional Mains	X(opt)	
Network Signaling	X(opt)	X
Free Calling Area Screening	X	
Network Class of Service	X	X
Coordinated Dialing Plan	X(opt)	X(opt)
Network Authorization Code or Basic Authorization Code	X(opt)	X (opt)
Network Traffic Measurements	X(opt)	X (opt)

Note 1: opt = optional.

Note 2: A Meridian SL-1 switch is termed a Conventional Main if none of the above feature packages are equipped at the switch.

NETWORK
 ENGINEERING
 SUPPORT ELEMENT

1.07 In addition to the comprehensive support services currently provided for Meridian SL-1 systems, Northern Telecom provides network engineering support for ESN applications. This support includes:

- network design
- | network engineering assistance
- network engineering practices.

1.08 A three-phased approach to network design and engineering assistance is employed by Northern Telecom for ESN applications. In the first phase, preliminary network design alternatives are assessed. This assessment is based on estimated traffic statistics and cost/performance objectives supplied by the customer, and is used to determine the economic viability of an ESN network for a particular application.

1.09 Once the viability of the proposed network is established, further analysis is undertaken to generate a detailed network configuration suitable for implementation. This analysis constitutes the second phase.

1.10 The third phase consists of follow-up refinements to the initial network configuration, based on an analysis of traffic and network calling data accumulated at the Meridian SL-1 ESN Nodes and Meridian SL-1 ESN Mains when the network is fully operational. This analysis also ensures that the potential benefits of ESN are actually achieved.

1.11 Additional technical considerations are needed to successfully plan, implement, and maintain a telecommunications network over those required to support regular Meridian SL-1 PBX services. These technical considerations are discussed in the following network engineering practices:

309-3001-180 Signaling Guidelines

309-3001-181 Transmission Guidelines

2. NETWORK CLASS OF SERVICE

2.01 The NCOS feature is an integral part of network control and routing control in an ESN. The NCOS feature provides the means to control:

- | which trunk routes are eligible to be accessed to attempt call completion
- | whether or not queuing is offered to a call originator
- | whether or not the originator of a network call receives a warning tone when an expensive trunk is selected to complete the call
- | whether or not the user is allowed to access the Network Speed Call (**NSC**) feature.

2.02 A Meridian SL-1 ESN Node (and Meridian SL-1 ESN Main) can accommodate 16 (O-15) NCOS groups (with **NARS**) and 8 (0-7) NCOS groups (with **BARS**) - each group with different network-access characteristics. Once each NCOS group is defined through Meridian SL-1 service change, then line, trunk, and attendant groups connected directly to the switch are assigned to the NCOS group which serves best the particular requirements of that line, trunk, or attendant. The NCOS to which each group is assigned, is independent of the regular Meridian SL-1 class of service (**553-2311-105**) assigned to them. Incoming tie trunks which connect another switch (Meridian SL-1 ESN Main, Conventional Main, ETN switch) to the Meridian SL-1 ESN Node are also assigned to an NCOS group which determines their level of access to the network facilities at the Meridian SL-1 ESN Node.

FACILITY RESTRICTION LEVEL

2.03 Included as part of each NCOS group is a Facility Restriction Level (**FRL**) number which ranges from 0 (low-privilege) to 7 (high-privilege). The FRL is used by the Meridian SL-1 software to determine the alternate route selection choices available for specific network call attempts by a line or trunk within an NCOS group.

2.04 For example, a station user assigned in an NCOS group having an FRL of 3 would be allowed access only to alternate route selection choices assigned an FRL of 3 or less; access to trunks with an FRL greater than 3 would be denied. Thus, by assigning low-privilege network users to an NCOS group having a low FRL, and high-privilege network users to an NCOS group having a higher FRL, the customer can control worker/management access to all network facilities.

EXPENSIVE ROUTE WARNING TONE

2.05 In some instances, expensive trunk routes may be assigned to an NCOS group with an FRL which would allow them to be accessed by some network users. When this occurs, the originator of the network call may be sent an optional Expensive Route Warning Tone (**ERWT**).

2.06 The ERWT tone alerts the caller that an expensive route has been selected to complete the call, and provides the caller with the option of either accepting or rejecting that the call be completed over the expensive route. Eligibility for ERWT is allowed or denied to individual lines and incoming trunk groups on an NCOS group basis.

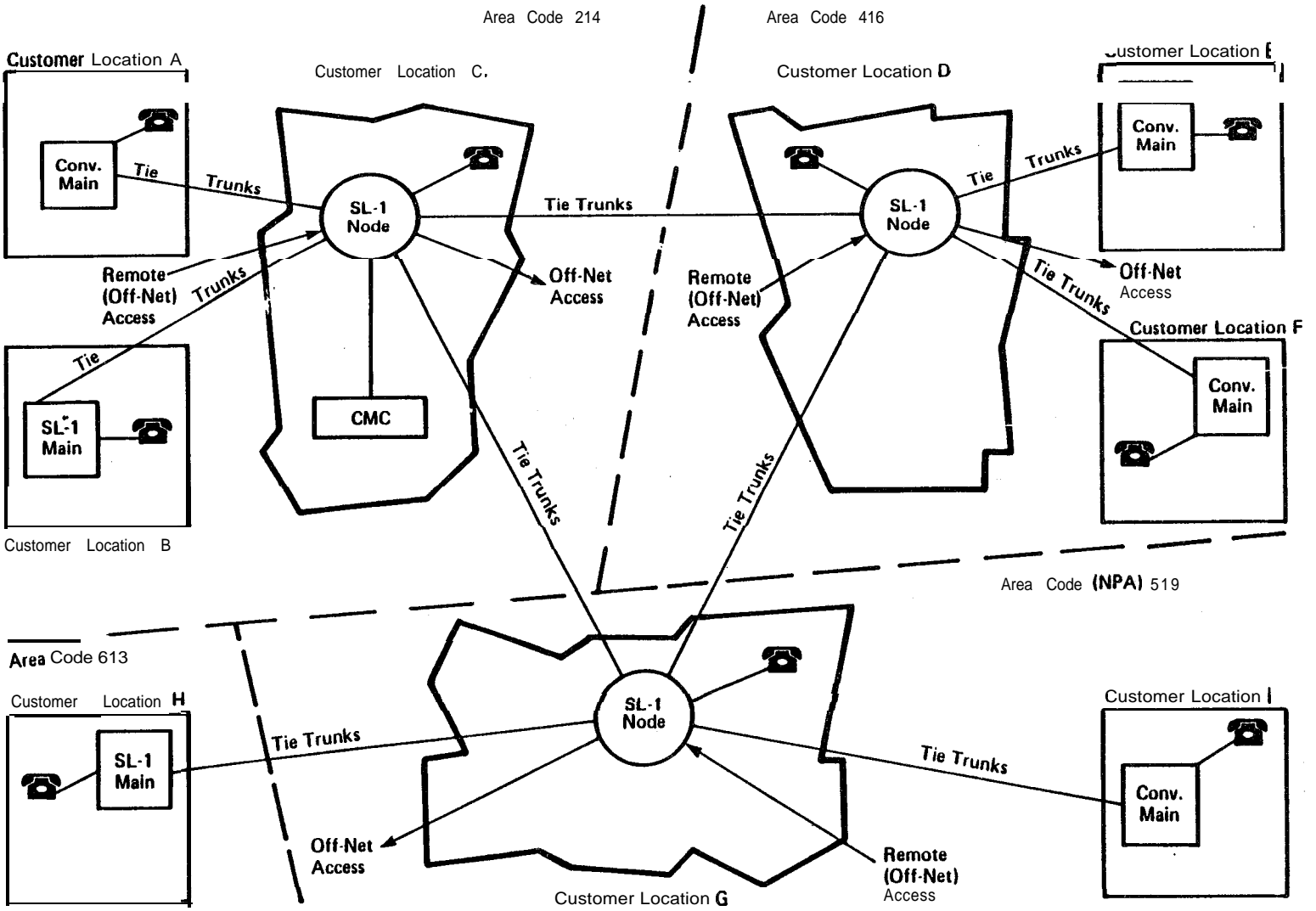


Fig. 1-1
Example of an Electronic Switched Network

QUEUING

2.07 Each NCOS group also defines whether or not the various queuing features are available to lines or trunks assigned to the group.

3. NETWORK ALTERNATE ROUTE SELECTION

3.01 The **NARS** feature provides a comprehensive and flexible networking package that can be configured to satisfy the specific requirements of a customer's network. The NARS feature provides benefits to users through simplified dialing plans and reduced communications costs. Prime elements of the NARS feature are:

- | simple network access codes
- | a **UDP**
- | dialing transparency
- | automatic least-cost routing
- | Time-of-Day (**TOD**) routing
- | automatic on-network (on-net) to off-network (off-net) overflow
- | network controls through Network Class of Service (**NCOS**), Travelling Class of Service (**TCOS**), Facility Restriction Levels (**FRL**), and regular Meridian SL-1 COS
- | routing control through NCOS alterations based on a special **TOD** schedule ↗
- 1-4 digit translation and 1-7 digit restriction (**X11 Release 4**)
- | 1-4 digit translation, 1-7 digit restriction, and 1-7 digit recognition (**X11 Release 5+**)
- | 1-11 digit translation, restriction, recognition (**X11 Release 8+**) ↘
- | Free Calling Area Screening (**FCAS**)
- | Expensive Route Warning Tone (**ERWT**)
- | Network Call Detail Recording (**CDR**)
- | Network Speed Calling (**NSC**).

NARS ACCESS CODES

3.02 To access NARS, the user at a Meridian SL-1 ESN Node, Meridian SL-1 ESN Main, or Conventional Main dials either one of two customer-assigned network Access Codes (AC), AC1 or AC2. These access codes are typically '8' for on-net and long distance calls (**AC1**), and '9' for of f-net and local calls (**AC2**). However, any 1-digit or 2-digit access code can be used, provided the access code assigned for AC1 is different from that assigned for AC2, and there is no conflict with any other part of the dialing plan.

Note: Only TIE trunks allow digit insertion of the AC1 code. DISA (CO) trunks require the user to dial the AC1 code.

3.03 Dialing a NARS access code triggers NARS to perform the necessary call-processing and routing using a specified set of network translation tables. This mechanism is used to implement the UDP for private networks. (NARS dial tone may or may not be provided to the caller after an access code is dialed at the option of the customer.)

↳ **11-DIGIT NARS
TRANSLATION (X11
RELEASE 8+)**

3.04 11-Digit Translation eliminates possible conflicts between translatable codes (NPA, NXX, LOC, SPN). By allowing translation of more than four leading digits, unique unconflicted routing to a destination is possible. More than one Route List Index can exist for each specific code of a type. For example, the NXX 727 could only translate into one Route List Index previously. With 11-Digit Translation, as many Route Lists as are needed to eliminate code conflict or achieve network requirements can be defined by extending translation deeper into the dialed code. The following table compares the number of digits that can be translated prior to Generic X11, Release 8 with the present capability.

TYPE	BEFORE X11 RELEASE 8	X11 RELEASE 8 ONWARD
LOC	3	3-7
HLOC	3	3-7
NPA	3-4	3-11
HNPA	3-4	3-11
NXX	3-4	3-8
SPN	1-4	1-11

↳

**UNIFORM DIALING
PLAN**

3.05 The UDP enables users at a Meridian SL-1 ESN Node, Meridian SL-1 ESN Main, or Conventional Main to dial all calls in a uniform manner, regardless of the location of the calling party or the route which the call will take.

**UDP For On-Net
Calling**

3.06 An on-net call is one which terminates at a customer-owned location. To reach any on-net location, the user dials the on-net access code (AC1), followed by seven digits. The format for this call would be:

AC1 * LOC + XXXX

where:

AC1 = the 1-digit or 2-digit on-net access code

* = pause for NARS dial tone (optional)

LOC = a 3-digit location code assigned for the destination location.

XXXX = the extension number of the party to be reached at the destination location.

3.07 Each switch which is part of the network (including the Meridian SL-1 ESN Nodes) is identified by a unique 3-digit location (LOC) code assigned at the Meridian SL-1 ESN Node. There must be no conflict between the location code number assigned for a switch and all Number Plan Area (NPA) codes.

3.08 A customer-owned location can be either physically connected to the network (via private trunk facilities) or virtually connected to the network (via public facilities). If a location is virtually connected to the network, the dialed LOC code is translated and converted by NARS translation (at the Meridian SL-1 **ESN** Node) into the public number for the virtual location: i.e., the Direct Distance Dialing (**DDD**) number or the Direct Inward Dialing (**DID**) number.

3.09 Example. Figure 3-1 illustrates an ESN network with a typical UDP. A user at LOC 776 (Conventional Main. location **I**) wishing to call extension number 3283 at LOC 777 (Meridian SL-1 **ESN** Main, location **H**) would first dial 8 (**AC1**), pause for the optional NARS dial tone from Meridian SL-1 ESN Node at location G, then dial 777-3283. A user at any other customer location which is part of the network would dial these same numbers to reach extension 3283.

UDP For Off-Net
Calling

3.10 An off-net call is one which does not terminate at a customer-owned location, even though some on-net facilities may be used to complete a portion of the call routing. Referring to Fig. 3-1, a call would be termed off-net if a user at LOC 776 called a station number associated with CO **758-XXXX** in the foreign area code 214.

3.11 Table 3-A lists the dialing formats for the various types of UDP calls.

Fig. 3-1
Example of an ESN With a Typical Uniform Dialing Plan

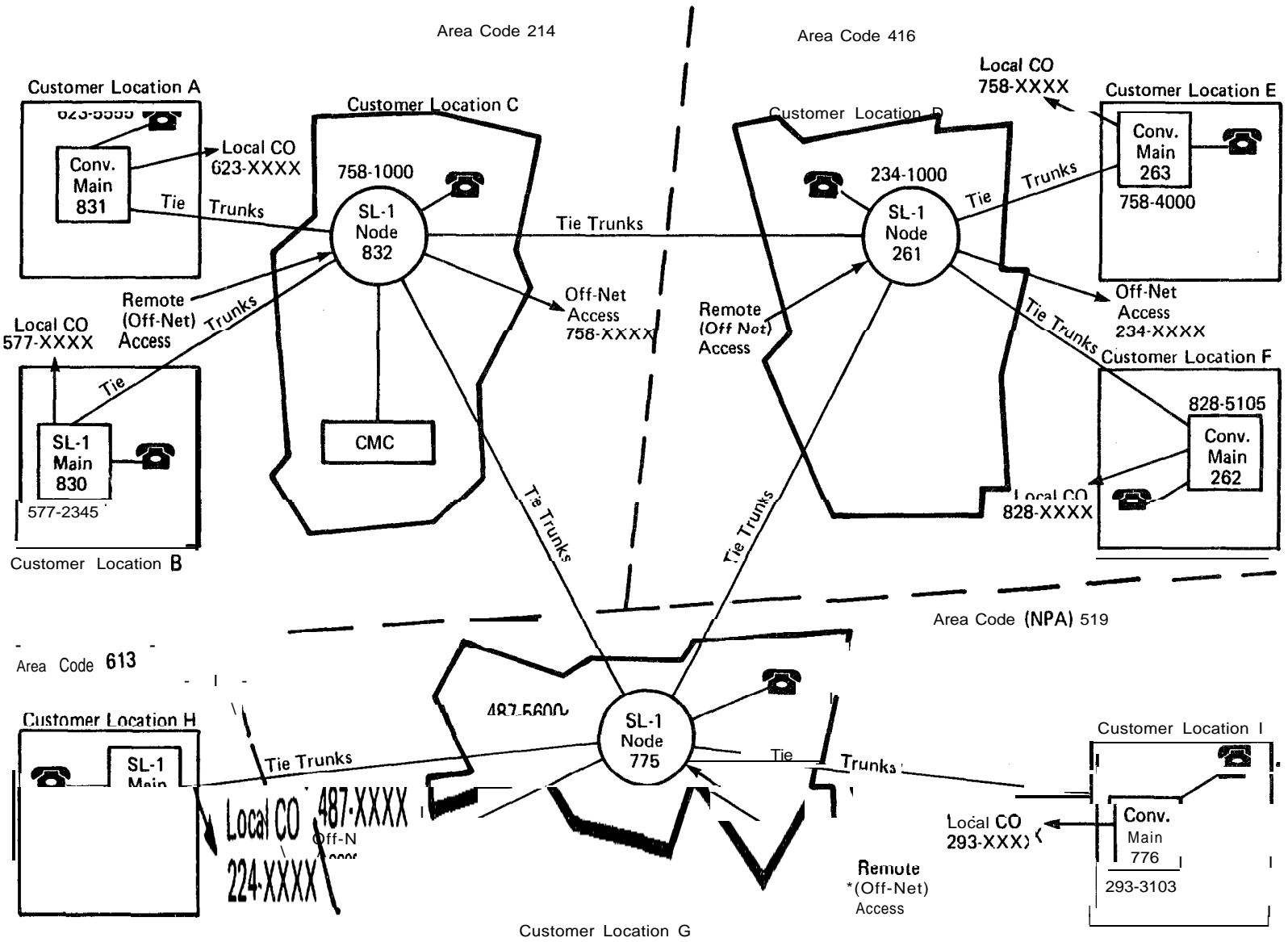


Table 3-A
DIALING FORMATS FOR NARS UDP CALLS

CALL TYPE	DIALING FORMAT	CODE CAPACITY
On-Net (Notes 1.2 and 3)	AC1 * LOC + XXXX	640
DDD FNPA (Note 1)	AC1 * 1 + NPA + NXX + XXXX	160
Network Speed Call	AC1 * LA + LN	1
Operator-assisted DDD	AC1 * 0 + NPA + NXX + XXXX	160
International DDD	AC1 * 011 + CC + NN	99
Operator-assisted International DDD	AC1 * 01 + CC + NN	99
DDD HNPA (Note 1)	AC1 or AC2 * 1 + NXX + XXXX	1 ←
DDD Operator	AC2 * 0	1
Local Calls (Note 1)	AC2 * NXX + XXXX	640
Special Local Services	AC2 * SPN	8
Toll-free Calls	AC2 * 800 + NXX + XXXX	1
Toll-free Calls (Note 1)	AC2 * 1 + 800 + NXX + XXXX	1
Toll Special Numbers	AC2 * 900 + NXX + XXXX	1
Toll Special Numbers (Note 1)	AC2 * 1 + 900 + NXX + XXXX	1

Note 1: If I-I-Dialing is used, the On-Net and Local Calls code capacities are increased to 800 and 792 respectively.

Note 2: If the code **1XX** is reserved for future **1+Dialing** use, and not for Network Speed Call codes, then the location code capacity will be reduced to 639 if a three digit NSC code is used, 632 if a two digit NSC code is used, or 560 if a one digit NSC code is used.

Note 3: When **1+Dialing** is used, Network Speed Call access will be in the form of **2XX-9XX** as a subset of the location codes utilized in the UDP. The location code capacity will be reduced to 799 if a three digit NSC code is used, 792 if a two digit NSC code is used, or 720 if a one digit NSC code is used.

LEGEND

AC1 Access code for on-net, long distance and Network Speed Calls. Typically the digit '8' but can be either one or two digits in length.

Table 3-A Continued
DIALING FORMATS FOR NARS UDP CALLS

AC2	Access code for local calls. Typically the digit '9' but can be either one or two digits in length.
*	Symbol meaning wait for NARS dial tone (optional).
NPA	Numbering Plan Area (NPA) code. Any number of the form NOX or N1X.
HNPA	Home Numbering Plan Area (HNPA) code. Any number of the form NOX or N1X.
FNPA	Foreign Numbering Plan Area (FNPA) code. Any number of the form NOX or N1X.
c c	Country code. Any one, two or three digits of 2 to 9.
NN	National Number. Depends on national dialing plan: maximum 12 digits including the Country Code.
N	Any of the digits 2 to 9.
X	Any of the digits 0 to 9.
LA	List access code. Any one, two or three digits of 0 to 9.
LN	List element number. Any one, two or three digits of 0 to 9 up to a maximum of 1000 element numbers.
LOC	Three-digit location code for each UDP network location.
NXX	Local Exchange Code.
XXXX	Four-digit directory (extension) number.
SPN	Special numbers e.g., 411, 611 etc. or may be XXXX.

DIALING
TRANSPARENCY

3.12 Extending the UDP to a remote-access switch (Meridian SL-1 ESN Main or Conventional Main) is accomplished by forming a single tie trunk access group between the remote switch and the Meridian SL-1 **ESN** Node. Users at this remote switch access the trunk group to the Node by dialing the on-net access code (**AC1**). The Meridian SL-1 ESN Node is arranged to insert the digit for AC1 on each incoming call from the switch, thus enabling access for on-net and long distance calling in a transparent fashion. Local calling is arranged through conventional dial '9' CO trunks at the remote access switch.

Note: If a Meridian SL-1 ESN Node replaces a tandem switch in a Tandem Tie Trunk Network (TTTN), other tandem switches in the network can 'tandem through' the Meridian SL-1 ESN Node using the same access codes as before. This requires that there be no dialing conflicts between the access codes for the TTTN trunks and the dialing plan implemented at the Meridian SL-1 ESN Node.

AUTOMATIC LEAST-COST ROUTING

3.13 For each on-net or off-net call translated at a Meridian SL-1 ESN Node, NARS selects a route from a list of up to eight outgoing alternate routes to complete the call. A list of alternate routes to a particular destination is called a route list, and each route specified in the list is termed an entry.

3.14 Any combination of trunks, e.g., Central Office (CO), Foreign Exchange (FX), or TIE, can be specified in a route list. Typically, the first entries (routes) in a route list are the least-cost routes to a destination and comprise the 'initial set' of routes in the list, whereas the last entries in the list are the most expensive and comprise the 'extended set' of routes in the list. An initial set 'marker', defined through Meridian SL-1 service change, determines which routes comprise the initial route set. There can be a maximum of 256 (O-255) route lists defined at each Meridian SL-1 ESN Node.

Note: For BARS, there can be a maximum of 128 route lists+ defined at each SL-1 ESN Node. ←

3.15 Associated with each entry in a route list is information relevant to:

- the route number (O-127)
- the minimum FRL required for access
 - | the time of day the route can be accessed
- whether or not CBQ or OHQ is allowed on the route
- whether or not the route is to receive expensive route warning tone treatment
- a digit manipulation table index number (O-255)
- a FCAS table index number (O-255)
 - | activation/deactivation of conversion from an on-net call to an+ of f-net call. ←

ROUTE ELIGIBILITY

3.16 NARS translates a dialed LOC, NPA, NXX, or SPN (special+ number, 3-11 digit.9 into a route list, and searches the list sequentially+ for an available route. Route eligibility for a given call is based on the caller's NCOS, the NCOS-defined FRL, the current TOD, and Meridian SL-1 cos.

3.17 Because each entry in a route list has a minimum FRL required for access and all network users are assigned an FRL through their NCOS, the network communications manager can restrict the type of calls allowed to particular users.

3.18 Example. If the minimum FRL for all calls is 1, except for special local services numbers which are assigned an FRL of 0, then a user assigned to an NCOS group with an FRL of 0 would only be able to make calls to the special numbers. In addition, the communications manager can restrict the use of high-cost facilities by assigning a high FRL to the expensive routes in a route list and a lower FRL to a user's NCOS.

DIGIT MANIPULATION

3.19 As mentioned previously, any trunk type can be specified in a route list. However, when certain trunk types are accessed, the digits dialed by the user must be manipulated to conform to the dialing requirements of the trunk. To do this, NARS uses digit manipulation tables to modify the dialed digits. There can be a maximum of 256 digit manipulation tables, each referenced by a digit manipulation index number, defined at each Meridian SL-1 ESN Node (Fig. 3-2). Digit manipulation can delete up to 15 leading digits, and insert up to 20 leading digits.



3.20 Example. A user at Conventional Main location I (Fig. 3-1) dials 8-613-596-9084 to reach an off-net station in the 613 NPA associated with Meridian SL-1 ESN Main, location H. At the Meridian SL-1 ESN Node, NARS selects the appropriate route list for call completion to NPA 613, and finds that the only available route to that NPA is a local CO trunk which requires the insertion of the leading digit '1' for long distance calls. The route list entry for this route specifies a digit manipulation index number (O-255: 0 means no digit manipulation is required). NARS references the digit manipulation table indicated by the index number, deletes digits as specified in the table (none in this case), and inserts the required digits ('1' in this case), and completes the call on this route.

TIME-OF-DAY ROUTING

3.21 NARS provides for up to eight (O-7) TOD schedules. Each entry (route) in a route list is assigned to the TOD schedule which specifies the hour(s) that the particular entry can be accessed. Thus, based on the current time of day, the most cost-effective route alternatives can be specified. A typical TOD schedule is shown in Table 3-B.

Table 3-B
A TYPICAL TOD SCHEDULE

TOD SCHEDULE	TIME PERIOD
2	00:00 to 07:44 17:30 to 23:59
1	07:45 to 08:59 12:00 to 13:14 16:00 to 17:29
0	0900 to 11:59 13:15 to 15:59

Note: A TOD schedule can be associated with any number of arbitrarily selected 15-minute periods. However, any one 15-minute period can appear only in one TOD schedule.

3.22 Based on the TOD schedule in Table 3-B, a route list entry assigned to TOD schedule 2 would be accessed only between the hours of 00:00 to 07:44 and 17:30 to 23:59. Access to the route at any other time would be denied. TOD schedules can be turned 'on' or 'off' through Meridian SL-1 service change as traffic conditions warrant. A TOD schedule is turned on for an entry by turning off all other TOD schedules.

AUTOMATIC ON-NET
TO OFF-NET
OVERFLOW

3.23 If all on-net facilities to a location are busy or blocked, NARS can convert a dialed UDP number to the Listed Directory Number (LDN) or DID number of the destination location, and use off-net facilities to complete the call.

3.24 Example. A user at Conventional Main location I (Fig. 3-1) dials S-777-3283 to reach a party with extension number 3283 at Meridian SL-1 ESN Main location H. At the Meridian SL-1 ESN Node, NARS translates the dialed LOC number (777) into a route list, and searches all eligible routes in the list. Failing to find an available tie trunk route, NARS then seizes local off-net facilities and, to complete the call, outpulses either:

- (a) 224-3283, if location H is arranged for DID, or
- (b) 224-5600, if location H is not arranged for DID.

3.25 Limitations.

Prior to Generic X11 Release 8, only one LDN may be defined per location code (LOC).

Prior to Generic X11 Release 5, only one contiguous DID DN range can be defined per location. DN's which lie outside the range are converted to the LDN.

Note: The capability of this feature has been enhanced in Generic X11 Release 5 software by multiple DID Office Code Screening and in X11 Release 8 by 11-digit Translation.

MULTIPLE DID OFFICE
CODE SCREENING (X11
R5+)

3.26 Multiple DID Office Code Screening is an enhancement to the On-Net to Off -Net Overflow capability of the NARS feature. This enhancement permits on-net calls which are routed through the public network using on-net to off-net conversion, to terminate at any DN which has been defined in the Location Code data block of memory. For each LOC defined, Multiple DID Office Code Screening will:

- 1 allow the definition of multiple NXX codes
- 1 allow the definition of multiple ranges of DN within each NXX



3.27 The following arrangements of Multiple office codes (NXX) and multiple DN ranges are possible:

- single office code with a single DN range
(the only alternative prior to generic X11 release 5)
- single office code with multiple DN ranges
 - | multiple office codes, each with a single DN range
- multiple office codes with multiple DN ranges

3.28 Limitations.

- | Only one NPA per LOC code is allowed.
- Ranges defined within a LOC code must be unique. Overlapping or duplication of ranges is not permitted.
- The number of digits in each DID range must be 4.
- A maximum of 20 DID ranges may be defined per location code, regardless of the number of office codes.

INCOMING TRUNK
GROUP EXCLUSION
(X11 R5+)

3.29 Incoming Trunk Group Exclusion is an enhancement to the NARS feature which blocks calls from Main users who use the network to reach destinations in the home NPA, or other restricted NPAs, NXXs, LOCs and SPNs. When the feature is implemented, users cannot use the network to circumvent the restrictions. They are forced to dial off-net instead and become subject to whatever restrictions are imposed at the Main.

3.30 Standard call blocking is applied on outgoing calls to specific NPA, NXX, SPN or LOC at the ESN node if the call is from a specific incoming trunk group.

- (a) This prevents loop-back routing through the caller's home switch (i.e., home NPA, NXX). Calls which should have been made off-net from the caller's home switch are blocked outgoing at the node.
- (b) Main users are prevented from using ESN to make calls to certain NPA, NXX, SPN or LOC that they are restricted from making at the home switch.

3.31 Customers define sets of restricted trunk routes to specific NPA, NXX, SPN or LOC. There is one Incoming Trunk Group Exclusion Index (maximum 255) for each defined NPA, NXX, SPN or LOC. Each index points to an Incoming Trunk Group Exclusion (ITGE) table. A maximum of 128 restricted routes can be defined in each ITGE table. Incoming Trunk Group Exclusion provides full 10-digit restriction for NPA and SPN codes, 7-digit restriction for NXX codes and 3-digit restriction for LOC codes.



OFF-NET NUMBER
RECOGNITION (X11
RELEASE 5+)

3.32 When an ESN call is received, NARS tests to see if the dialed code⁷ is a restricted type (Supplemental-Digit Restriction). If it is, NARS checks whether or not it has an ITGE restriction and if there is an ITEI number (1 -255) associated with it. If an ITEI is defined, the appropriate ITGE table, corresponding to the dialed code is searched. If the incoming trunk route is a member of the ITGE, the NARS process is terminated and the call is blocked. Otherwise, call processing continues.

3.33 Off-Net Number Recognition eliminates the need of using two extra CO trunks, when a subscriber, using the private network, dials a DID or DDD number that terminates at an ESN Location. Calls are routed directly to the dialed DN (DID calls) or to the LDN (DDD calls), rather than being switched from the terminating ESN switch to the CO and back again.

3.34 Off-Net Number Recognition parameters for local and remote DDD and DID locations are defined by the customer in the Network Translation tables and Supplemental Digit Recognition/Restriction blocks (SDRR). Recognition of up to 10 digits can be defined.

CALL TYPE	NETWORK TRANSLATION TABLE (# of digits)	SDRR BLOCK (# of digits)
NPA	3	1 - 7
1NPA	4	1 - 7
NXX	3	
1NXX	4	1 - 4
SPN	4	1 - 7

3.35 With 11-Digit Translation, available on Generic X11 release 8 and onwards, up to 11 digits can be defined as follows:

CALL TYPE	NETWORK TRANSLATION TABLE (# of digits)	SDRR BLOCK (# of digits)
NPA	3 - 10	1 to (10-N)
1NPA	4 - 10	1 to (11-N)
NXX	3 - 7	1 to (7-N)
1NXX	4 - 8	1 to (8-N)
SPN	1 - 11	1 to (11-N)

3.36 Up to 512 SDRR blocks can be defined. Each table can contain up to 64 entries.

3.37 Off-Net numbers are recognized at the last intelligent NARS/BARS switch. Translation of the NPA, NXX or SPN identifies the method of treatment for the call. If the data type is SDRR and the index is an SDRR table index, supplemental digit recognition/restriction is applied by comparing the dialed digits with the numbers declared in the SDRR block.



- (a) If no match is found in the SDRR, route selection is called, call processing resumes and the call is routed to the CO of the terminating off-net number.
- (b) If a match is found and the number is in the 'denied' block, standard call blocking takes place.
- (c) If a match is found and the number is recognized as a terminating number at the local switch (i.e., the last intelligent NARS/BARS switch), the call is terminated at the station DN (DID calls) or at the attendant DN (DDD calls).
- (d) If a match is found and the dialed number is a recognized number terminating at a remote conventional main, route selection is called, the appropriate digit manipulation takes place and the call is routed directly to the conventional main. DID calls terminate at the dialed station and DDD calls terminate at the attendant DN.

4

→ DIGIT TRANSLATION/
RESTRICTION/
RECOGNITION

3.38 The ESN provides a 1-digit through 11-digit translation/restriction/recognition capability through the use of network translation tables. There are two network translation tables; one associated with each of the network access codes (AC1 and AC2).

3.39 Normal Meridian SL-1 translation mechanisms translate the dialed network access code (e.g., 8 or 9), determine that the call is to be processed by NARS, and selects the appropriate network translation table (Fig. 3-2). The NARS translation determines the method to be used to process the call and applies digit restriction or recognition, where required. The result of translation is to invoke either route selection with a specified route list, or standard call blocking. More information on digit recognition can be found under 'Off-Net Number Recognition'.



3.40 Information contained in the network translation tables is as follows:

- (a) For each NPA (area code) excluding the Home NPA (HNPA):
 - a route list index number (O-255) which indicates which route list to use in processing a call to this NPA entry,
 - whether or not there are telephone numbers within this NPA to which network calls are to be blocked; i.e., denied, (a list of up to 64, 1- to 7- digit numbers that are to be blocked or recognized in this NPA)
 - whether or not there are telephone numbers within this NPA to which network calls are to be blocked because of ITGE restrictions (X11 Release 5+)
 - whether or not there are numbers under this NPA that are to be recognized as DID or DDD codes to an On-Net location (X11 Release 5+)



- (b) For each NXX office code:
 - a route list index number (O-255) which indicates which route list to access in processing a call to this NXX entry,

- whether or not there are telephone numbers within this NXX entry to which network calls are to be blocked,
- whether or not there are telephone numbers within this NXX entry to which network calls are to be blocked because of ITGE restrictions (X11 Release 5+).
- whether or not there are numbers under this NXX that are to be recognized as DID or DDD codes to an On-Net location (X11 Release 5+)
- a list of up to 64, 1- to 4-digit numbers that are to be blocked or recognized in this NXX (X11 Release 8+).

(c) For each LOC location code, excluding the Home LOC code:

- a route list index number (O-255) which indicates which route list to access in processing a call to this LOC entry,
- ┆ the Listed Directory Number (LDN) to which the LOC entry number is to be converted when using off-net DDD facilities.
- the ranges of DID numbers to which the LOC number can be converted when using DID facilities.
- whether or not there are LOC numbers to which network calls are to be blocked because of ITGE restrictions (X11 Release 5+).

(d) For each SPN (special number):

- a route list index number (O-255) which indicates which route list to access in processing a call to this SPN
- ┆ whether or not there are SPN numbers to which network calls are to be blocked,
- ┆ whether or not there are SPN numbers to which network calls are to be blocked because of ITGE restrictions (X11 Release 5+).
- ┆ whether or not there are SPN numbers that are to be recognized as codes to an On-Net location (X11 Release 5+)
- a list of up to 64, 1-to 11-digit SPN numbers that are to be blocked or recognized.

SUPPLEMENTAL DIGIT
RESTRICTION

3.41 Supplemental digit restriction blocks (Fig. 3-2) enable the customer to block (deny) access to certain telephone numbers; e.g., dial-a-joke. There can be a maximum of 256 supplemental digit restriction blocks at a Meridian SL-1 ESN Node. One block can be assigned per NPA, NXX, or SPN. Each block can restrict up to 16 numbers. The customer can also specify (through Meridian SL-1 service change) the treatment that blocked calls receive; e.g., overflow tone, intercept to attendant, or recorded announcement.

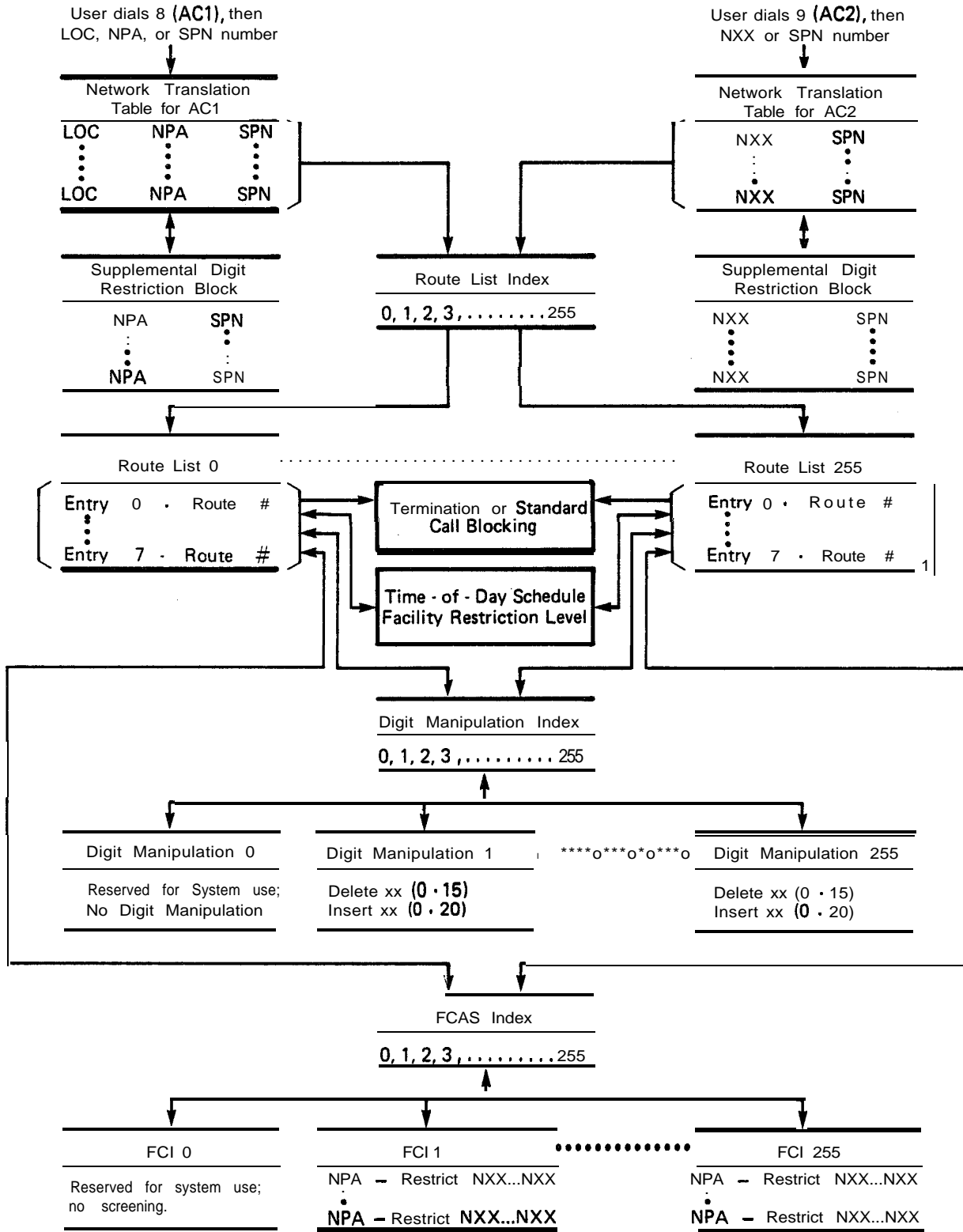


Fig. 3-2
NARS Elements Accessed at a Meridian SL-1 ESN Node to Process a Network Call

SUPPLEMENTAL DIGIT
RESTRICTION/
RECOGNITION
(X11 RELEASE 5+)

3.42 Supplemental Digit Restriction/ Recognition blocks (Fig. 3-2) enable the customer:

- to block (deny) access to certain telephone numbers (e.g., dial-a-joke)
- | to recognize Off-Net calls dialed to On-Net locations (Off-Net Number recognition)
- | to prevent routing of calls to the home switch of the originating trunk group by either on-net or off-net facilities (ITGE).

3.43 There can be a maximum of 512 supplemental digit recognition/restriction blocks at a Meridian **SL-1** ESN Node. One block can be assigned per type (e.g., DENY, LDID, DDD, etc). Each block can recognize or restrict up to 64 numbers. The customer can also specify (through Meridian SL-1 service change) the treatment that blocked calls receive; e.g., overflow tone, intercept to attendant, or recorded announcement.

FREE CALLING AREA
SCREENING

3.44 The FCAS is a BARS or NARS feature which provides the customer with the capability of full six-digit (NPA-NXX) screening to determine the route choice for completion of off-net calls. With FCAS, a customer can allow calls to NXX codes within the 'free calling area' surrounding a particular on-net location, and restrict (deny) calls to those NXX codes that would incur long distance charges.

3.45 The method used for FCAS is similar to the method used for digit manipulation (i.e., through FCAS tables). There can be up to 255 FCAS tables defined with NARS (127 with BARS) at a Meridian SL-1 ESN Node. Each table can contain up to 15 NPA codes with NARS (7 with BARS), and up to 800 NXX codes can be restricted (allowed or denied) within each NPA code. Each FCAS table is referenced by a Free Calling Index (FCI) number (0-255), where FCI=0 is a system default meaning no Free Calling Area Screening is required. The appropriate FCI number is then assigned to the applicable route list entries.

3.46 Whenever a route list entry is being considered for an off-net call (e.g., 8-NPA-NXX-XXXX), NARS checks to see if there is an FCI number (other than 0) referred to by the entry. If an FCI number other than 0 is defined, the appropriate FCAS table for the dialed NPA is found and used for NXX screening. If the dialed NXX is denied in the table, NARS will not use the route list entry for call completion, but will continue to search for another eligible route list entry. If the dialed NXX is not denied in the table, the route list entry is eligible for the call. (Calls converted to the LDN of a location are screened only if the NPA is included as part of the LDN.)

EXPENSIVE ROUTE
WARNING TONE



3.47 This feature enables the network communications manager to select certain users to receive an Expensive Route Warning Tone (ERWT). Eligibility for this tone is based on the user's NCOS. The tone (three ~~256-ms~~ bursts of 440 Hz) notifies the user that NARS or BARS has selected facilities designated as expensive to complete the call. Upon receipt of ERWT, the user has the choice of either allowing the call to complete over the expensive facilities, or going on-hook to avoid the increased expense or queuing on the I set routes. (The user must make this choice within a programmable time of 0 to 10 s.) If the call originator is located at a Meridian SL-1 ESN Node or Meridian SL-1 ESN Main and the Meridian SL-1 Ring Again feature is defined for the user and the user is **CBQ(a)** eligible, then ring again may be activated to queue the call - see the various queuing features. If the Meridian SL-1 ESN Node is equipped for Call Detail Recording (CDR), acceptance of an expensive route after ERWT is received is noted in the CDR record.

NARS/BARS BYPASS
CONTROL

3.48 A customer can, if desired, allow selected users to bypass the **NARS/BARS** feature for call completion between any two locations; e.g., two locations which share a high community of interest. To do this, routes and trunks are set up between the two locations, and assigned an access code distinct from the AC1 and AC2 codes used to access NARS. The normal Meridian SL-1 trunk controls; e.g., Trunk Group Access Restriction (**TGAR**), class of service, code restriction, etc., are then used to enable access only to the selected users - all other users are denied access to the trunk group and, thus, forced to use NARS for all calls.

NETWORK SPEED CALL



4



3.49 The NSC feature enables a user at a Meridian SL-1 ESN Node, Meridian SL-1 ESN Main, or Conventional Main who is normally restricted from making certain types of **NARS/BARS** calls, to make such a call if the destination is a company-approved number defined in a System Speed Call (**SSC**) list. This feature requires that the System Speed Call feature (**553-2001-105**) be equipped.

3.50 Access to the NSC feature is allowed after the **NARS/BARS** access code is dialed. Upon receipt of **NARS/BARS** dial tone (optional), the user dials a customer-defined Network Speed Call access code of 1-3 digits. (The NSC access code must be unique from all LOC, NPA and NXX codes, and special numbers defined in the translator for the NARS access code.)

3.51 The NSC access code is associated with a previously-defined System Speed Call list (O-253) through Meridian SL-1 service change. If the SSC list has its length (size) changed, the list access code and list number must be deleted and reentered into the **NARS/BARS** translator. Associated with the SSC list is an NCOS number (0-7 with BARS, 0-15 with **NARS**). The NCOS assigned to the SSC list is applied to the call only if the FRL (O-7) is greater than that associated with the call originator's assigned NCOS.

3.52 If **1+** Dialing is specified for an NPA, **NXX** or SPN number in a translator, the digit '1' must not be used as the leading digit for Network Speed Call list codes in that translator.

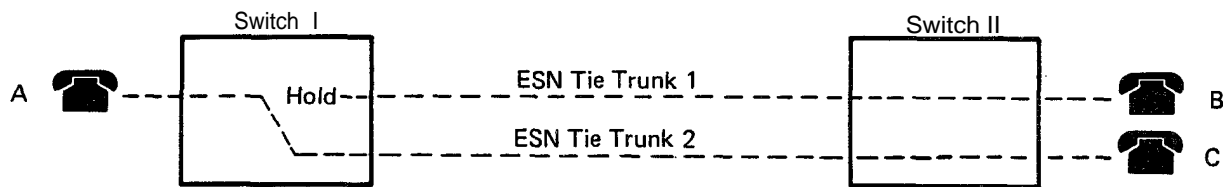


Fig. 3-3
Connection during Network Call Transfer

3.53 The user then dials the number of the desired entry in the SSC list. (Entry numbers can range from 0 to 9, 00 to 99, or 000 to 999 depending upon the number of elements allocated when the list was defined through service change.) Upon completion of dialing, the digits defined for the list entry are passed to NARS translation for processing. Route and feature (OHQ, CBQ) eligibility for call completion are based on the NCOS associated with the SSC list, only if the FRL of the user's assigned NCOS is lower.

NETWORK CALL TRANSFER (NXFER)

3.54 This feature improves the operation of the existing Call Transfer (XFER) feature between two Meridian SL-1 switches when a call is transferred back to the originating switch. The regular XFER feature requires two TIE trunks to complete the call. With NXFER, if the call is transferred back to the originating switch via a TIE trunk of the same trunk group, the originating switch completes the transfer within itself and the TIE trunks are dropped. Refer to 553-2751-100 for a detailed description.

Note: Both SL-1 switches must be equipped with NSIG and* NXFER software for this feature to operate. ←

3.55 The benefits derived from the NXFER feature include:

- minimal use of access TIE lines.
- improved transmission performance since TIE lines are not used for the completed connection.
- operation of NXFER is the same as for the existing XFER feature.

3.56 Application. Fig. 3-3 and 3-4 assume that station A receives an incoming trunk call from B and transfers it to C.

- (a) As shown in Fig. 3-3, the NXFER feature allows station A at one ESN switch (I) to transfer the tie trunk call from station B (switch II) to a third party, station C (switch II), via a tie trunk of the same trunk group.
- (b) If the transfer is allowed, stations B and C are connected on switch II and the ESN tie trunks are dropped when the transfer is completed. See Fig. 3-4.

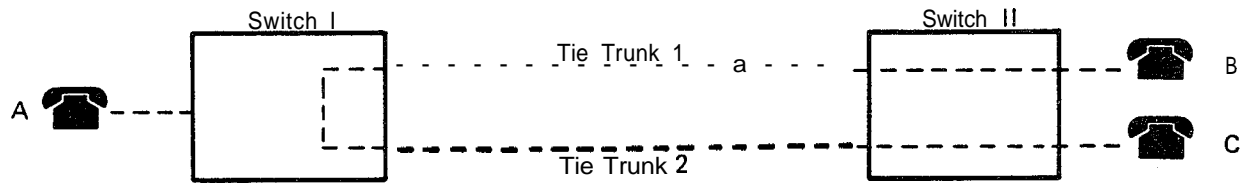


Fig. 3-5
Connection after Call Transfer

(c) In comparison, regular Meridian SL-1 Call Transfer (XFER) requires two tie trunks and both switches to connect stations B and C. See Fig. 3-5.

1+ DIALING

3.57 Translation Tables. With 1+ Dialing, both NARS translation tables and the single translation table for BARS are equipped for 4-digit translation (based on the first 1, 2, 3 or 4-digits), or 11-digit translation with X11 Release 8 and on, thus allowing NARS/BARS customers the option of dialing 1+ after the NARS/BARS access code for long distance calls. See Table 3-A for dialing formats for NARS Uniform Dialing Plan (UDP) calls.

4

3.58 Code Ambiguity. The 1+ Dialing capability also eliminates ambiguity between identical 3-digit NPA, NXX and LOC codes so that the NARS customer can route calls to an NPA, NXX or LOC code which conflicts with one of the customer's 3-digit codes.

3.59 Limitation. If 1+ Dialing is specified for an NPA, NXX or SPN number in a translator, the digit '1' must not be used as the leading digit for Network Speed Call list codes in that translator.



Fig. 3-4
Connection after Network Call Transfer

4. NETWORK CONTROL

4.01 Network control is an enhancement to the NCOS feature that extends NCOS controls to users located at Meridian SL-1 ESN Main switches. Network control requires that the Meridian SL-1 ESN Main and serving Meridian SL-1 ESN Node be equipped with the NSIG feature.

MERIDIAN SL-1 ESN
MAIN NCOS

4.02 Users (lines, trunks, attendants) at a Meridian SL-1 ESN Main are assigned an NCOS (O-15) with NARS or (O-7) with BARS which is **used** ← to determine their level of access to network facilities at the serving Meridian SL-1 ESN Node. When a user at a Meridian SL-1 ESN Main initiates a call to (or through) a Meridian SL-1 ESN Node, the user's assigned NCOS is transmitted to the serving Meridian SL-1 ESN Node. The transmitted NCOS overrides the NCOS assigned to the incoming tie trunk group at the node, and is used to determine the user's eligibility for **network** resources/features at the Meridian SL-1 ESN Node. Thus, a user at a Meridian SL-1 ESN Main has the same **network-access** capabilities as a user at the Meridian SL-1 ESN Node who is assigned the same NCOS.

Note: If the user at the Meridian SL-1 ESN Main enters a valid authorization code prior to placing a **NARS/BARS** call, the NCOS associated with the authorization code is transmitted to the Meridian SL-1 ESN Node, in place of the user's assigned NCOS.

4.03 Calls from a Conventional Main to the Meridian SL-1 ESN Node are controlled by the NCOS assigned to the incoming trunk group at the ESN Node.

MERIDIAN SL-1 ESN
NODE TCOS

4.04 Network Control at a Meridian SL-1 ESN Node provides a Traveling Class of Service (**TCOS**) mechanism which controls route access and Of f-Hook Queuing (**OHQ**) eligibility for calls placed to (or through) another Meridian SL-1 ESN Node or an associated Meridian SL-1 ESN Main, and enables the Meridian SL-1 ESN Node to interface with switches that are part of an ETN.

4.05 The Traveling Class of Service is, in effect, the FRL of a user's assigned NCOS. When a user at a Meridian SL-1 ESN Node initiates a call to another Meridian SL-1 ESN Node (or a Meridian SL-1 ESN Main) the TCOS (i.e., the FRL of the user's assigned NCOS) is transmitted to the other Meridian SL-1 ESN Node. At the receiving Meridian SL-1 ESN Node, the TCOS (O-7) replaces the FRL of the NCOS assigned to the incoming trunk group. Route access and OHQ eligibility for the call are, therefore, based on the NCOS of the incoming trunk group with the modified FRL (i.e., **TCOS**).

Note: The Network Control (**NCTL**) data block (overlay 87) is used to define OHQ eligibility on a per-FRL (**TCOS**) basis (e.g., if **FRL 4** is defined as OHQ eligible, then all users with an NCOS with an FRL of 4 are eligible for OHQ on calls placed to another Meridian SL-1 ESN Node or to an associated Meridian SL-1 ESN Main).

ETN SWITCH
COMPATIBILITY

4.06 If a user at a Meridian SL-1 ESN Main or Conventional Main initiates a call that tandems through the serving Meridian SL-1 ESN Node to another Meridian SL-1 ESN Node or Meridian SL-1 ESN Main, TCOS applies to the call as if the call originated at the serving Meridian SL-1 ESN Node.

4.07 The Meridian SL-1 TCOS is equivalent to the Traveling Class Mark (**TCM**) used at ETN switches. (See Technical Publication 42709, 'Tie Trunk Signaling Compatibility for Connecting to a DIMENSION PBX', July 1979.)

4.08 When a **7-digit/10-digit** UDP call or Distance Steering Code (**DSC**) CDP call is made from an ESN Node to an ETN switch, the dialed digits, together with the TCOS number (**0-7**), are sent to the connected ETN switch. At the ETN switch, the TCOS number received from the Meridian SL-1 ESN Node is used as a TCM to determine route access and off-hook queuing eligibility at the ETN switch.

4.09 Similarly, when a call is made from an ETN switch to a Meridian SL-1 ESN Node, the dialed digits, together with the TCM number (**0-7**), are sent to the connected Meridian SL-1 ESN Node. The Meridian SL-1 ESN Node interprets the received TCM number as a TCOS number. The received TCM (i.e., **TCOS**) replaces the FRL of the NCOS assigned to the incoming trunk group from the ETN switch. This new **FRL** (i.e., TCM) is then used to determine route access and off-hook queuing eligibility for the call.

4.10 However, if a DSC CDP call is terminated on a Meridian SL-1 switch as a Local Steering Code (**LSC**) call, the TCOS value transmitted by the connected switch will not be collected and saved by this switch.

5. NETWORK SIGNALING

5.01 The NSIG feature provides the required signaling protocol to interface Meridian SL-1 ESN Nodes with Meridian SL-1 ESN Mains, Meridian Sk-1 ESN Nodes with other Meridian SL-1 ESN Nodes, Meridian SL-1 ESN Nodes with Conventional Mains, and Meridian SL-1 ESN Nodes with ETN switches. This feature is required at any Meridian SL-1 ESN Nodes connected to the SL-1 ESN Mains.

5.02 When the NSIG feature is equipped at a switch, there are options available (route data block, OVL 16) to define the signaling arrangements between that switch and any other switch that may be connected to it via tie trunks. These options define what call information is to be transmitted to a connected switch and what call information is to be received from a connected switch. The option selected depends on the type of connected switch (ESN node, ESN main, conventional main, ETN) and the options (e.g., CCBQ, CBQCM) that are available to the connected switch.

5.03 The signaling options are: STD (standard), ESN, ESN2, ESN3 (Electronic Switched Network), and ETN (Electronic Tie Network).

- STD. Arranges the tie trunk group for transmission/reception of the called number between switches.
- ESN. (X11 Release 2 only.) Arranges the tie trunk group for transmission/reception of the call type, NCOS/TCOS, and called number between switches, and is required on systems equipped with the CCBQ / CBQCM feature.
- **ESN2.** (X11 Release 3.) Arranges the tie group as described for ESN in X11 Release 2.
- **ESN3.** (X11 Release 3 or 4.) Arranges the tie group as described for ESN in X11 Release 2 and is required on systems equipped with the NXFER or Satellite Link Control features.
- **ESN5.** (X11 Release 5.) Arranges the tie trunk group as described for ESN in X11 Release 5.
- ETN. Arranges the tie trunk group for transmission/reception of the called number and TCOS/TCM between switches.

5.04 Application. Following is a description of how these options would be applied to accommodate the different switch types that can be connected to a Meridian SL-1 ESN Main or Meridian SL-1 ESN Node that is equipped with the NSIG feature.

- (a) Meridian **SL-1** ESN Node. A Meridian SL-1 ESN Node can be connected via tie trunks to: another Meridian SL-1 ESN Node, a Meridian SL-1 ESN Main, a Conventional Main, and/or an ETN Switch.
- If the Meridian SL-1 ESN Node connects to another Meridian SL-1 ESN Node, both ends of the connecting tie trunk group are defined with the ETN option.

- If the Meridian SL-1 ESN Node is equipped with Generic X11 Release 2 and connects to a Meridian SL-1 ESN Main equipped with Generic X11 Release 2, both ends of the connecting tie trunk group are defined with the ESN option.
 - | If the Meridian SL-1 ESN Node is equipped with Generic X11 Release 3 and the Meridian SL-1 ESN Main is equipped with Generic X11 Release 3, both ends of the connecting tie trunk group are defined with the **ESN3** option.
 - If the Meridian SL-1 ESN Node is equipped with Generic X11 Release 2 and the Meridian SL-1 ESN Main is equipped with Generic X11 Release 3, the Node end of the connecting tie trunk group is defined with the ESN option, and the Main end is defined with the **ESN2** option.
 - If the Meridian SL-1 ESN Node is equipped with Generic X11 Release 3 and the Meridian SL-1 ESN Main is equipped with Generic X11 Release 2, the Node end of the connecting tie trunk group is defined with the ESN2 option, option and the Main end is defined with the ESN option.
 - If the Meridian SL-1 ESN Node connects to a Conventional Main, the node-end of the tie trunk group is defined with the STD option.
 - If the Meridian SL-1 ESN Node connects to an ETN switch, the node-end of the tie trunk group is defined with the ETN option.
- (b) Meridian SL-1 ESN Main. A Meridian SL-1 ESN Main can be connected via tie trunks to a Meridian SL-1 ESN Node and satellite switches.
- | If the Meridian SL-1 ESN Main is equipped with Generic X11 Release 2 and connects to a Meridian SL-1 Node equipped with Generic X11 Release 2, both ends of the connecting tie trunk group are defined with the ESN option.
 - | If the Meridian SL-1 ESN Main is equipped with Generic X11 Release 3 and the Meridian SL-1 ESN Node is equipped with Generic X11 Release 3, both ends of the connecting tie trunk group are defined with the **ESN3** option.
 - | If the Meridian SL-1 ESN Main is equipped with Generic X11 Release 2 and the Meridian SL-1 ESN Node is equipped with Generic X11 **Release** 3, the Main end of the connecting tie trunk group is defined with the ESN option, and the Node end is defined with the **ESN2** option.
 - | If the Meridian SL-1 ESN Main is equipped with Generic X11 Release 3 and the Meridian SL-1 ESN Node is equipped with Generic X11 Release 2, the Main end of the connecting tie trunk group is defined with the ESN2 option, and the Node end is defined with the **ESN** option.
 - If there are satellite switches (non-ESN) connected to the Meridian SL-1 ESN Main, the main-end of the tie trunk groups from the satellite switches are defined with the STD option.

5.05 Limitations. The following limitations apply.

- A Meridian SL-1 ESN Main can connect to only one Meridian SE-1 ESN Node. Both switches must be equipped with the NSIG feature.
- Tie trunks between Meridian SL-1 ESN Nodes and Meridian SL-1 ESN Mains must be arranged for DTMF sending/receiving, wink-start operation and have answer supervision.
- Meridian SL-1 ESN Node **compatibility** with ETN switches is limited to 7-digit on-network, **10-digit** off-network and DSC CDP calls.

SATELLITE LINK
CONTROL

5.88 Tandem trunk calls, when connected through more than one communications satellite trunk, are subject to transmission distortion due to propagation to and from communications satellites. The Satellite Link Control feature ensures that the configuration of a call does not include more than one communications satellite trunk.

5.07 Limitations.

- This feature only applies to ESN network calls (**NARS/BARS/CDP**).
 - | ESN Proprietary Signalling is required among ESN switches.
- Routes which receive digits from satellites or send digits to satellites have to be marked as SATELLITE routes.

MERIDIAN SL-1 TONE
DETECTOR

5.08 The Meridian SL-1 Tone Detector (**SL1TD**) is a peripheral equipment circuit pack designed to detect various tones used in automatic route selection features (**NARS, BARS, CDP**) interfacing with Special Common Carriers (**SCC**) and Tandem Tie Trunk Networks (**TTTN**). Call progress tone detection is used as an aid for the Meridian SL-1 system to control digit outpulsing to **SCC** and **TTTN**. A full description of the **SL1TD** feature is given in 553-2001-191.

5.09 First Tone Detection. Software requests the **SL1TD** to compare the first signal received on a trunk with one or more specified tones. The message sent to software identifies the tone detected as one of the following

- | Precise dial tone
- Precise busy/overflow tone
- SCC dial tone
- Busy tone
- Overflow tone
- Ringback tone
- Tone with a duration of 100 ms or more
- Tone with a duration greater than 300 ms
- Tone with a duration less than 300 ms
- Unidentified tone

- | Silence (reported while the unit is active and a specific tone is not yet detected).

5.10 Active mod I request identifies all the foregoing tones while active mod II request identifies the same tones except for unidentified tone.

5.11 Limitations.

- | A tone detector should not be designated when dial tone is not provided.
- | The SLITD does not apply to manual **500/2500** or Meridian SL-1 dialing.
- | When a route list entry specifies that a tone detector is required and none is available, that entry is not eligible for Network Queuing.

5.12 Special Common Carriers. When a Meridian **SL-1** connects to an SCC, a start-dial indication is required from the SCC. The SLITD detects the start-dial (SCC dial tone) and sends a message to software indicating that the SCC is ready to accept digits, or that a network busy or SCC busy tone has been encountered allowing the Meridian SL-1 to select an alternate route.

5.13 TTTN Dialing When tandeming through a tie trunk network, pauses are required to wait for dial tone from each trunk in the chain. Once dial tone is given, further digits can be accepted. Since in the Tandem Tie Trunk Network (TTTN) application of the SLITD the digits are dialed automatically, asterisks are inserted in the digit stream to allow for the pauses required. By using the SLITD to detect tones in the call sequence, call set-up reliability is improved and unnecessary delays are avoided. The SLITD can detect trunk blocking at tandem switches allowing alternate routing of calls.

6. ROUTING CONTROL

6.01 The Routing Control feature provides a mechanism for reducing or raising a user's network-access capabilities when a special TOD schedule is in effect.

NCOS MAP

6.02 With the NARS or BARS feature, TOD schedule 7 is the special TOD schedule. Associated with the special TOD schedule is a NCOS map. The NCOS map lists all NCOS numbers for NARS (0-15) and BARS (0-7). Associated with each listed NCOS is an alternate NCOS number (the same or smaller) which replaces the original NCOS number when the special TOD schedule is in effect. Table 6-A illustrates a typical NCOS map.

Table 6-A
A TYPICAL NCOS MAP FOR SPECIAL TOD SCHEDULE 7

ORIGINAL NCOS	ALTERNATE NCOS (Note)	ORIGINAL NCOS	ALTERNATE NCOS (Note)
0	0	8	2
1	0	9	3
2	0	10	3
3	1	11	4
4	1	12	4
5	2	13	5
6	2	14	5
7	2	15	5

Note: When TOD schedule 7 is in effect, the alternate NCOS replaces the user's original NCOS.

INVOKING ROUTING CONTROL

6.03 The alternate NCOS numbers associated with special TOD schedule 7 are normally invoked when the time specified for TOD schedule 7 corresponds to the time in the Meridian SL-1 system clock. Additionally, the alternate NCOS numbers can be scheduled for implementation (through Meridian SL-1 service change) for the full 24-hr period of specified days of the week. This capability enables network-access capabilities to be changed automatically on weekends or company holidays.

6.04 The Meridian SL-1 attendant can also manually invoke the special TOD schedule through use of a routing control (RTC) key on the console. Pressing the RTC key lights the associated Light-Emitting Diode (LED) lamp, and invokes the special TOD schedule. To deactivate routing control, the RTC key is pressed again. The associated LED goes dark and normal TOD schedules are once again in effect.

Note: Authorization code can be used to override the restrictions imposed through routing control. If a user enters a valid authcode, the NCOS number associated with the authcode is applied for the duration of the call.

7. OFF-HOOK QUEUING

7.01 The OHQ is a software feature that can be equipped at a Meridian SL-1 ESN Node and/or ESN Main. The feature enables a call originator to remain off-hook for a short time (customer programmable) until a network facility for call completion becomes available at the Meridian SL-1 ESN Node, or Meridian SL-1 ESN Main.

ELIGIBILITY

7.02 Network calls may be placed in an OHQ if all trunk routes (entries) in the initial route set of a route list (see Note) are busy, and the following criteria are met.

- | At least one of the trunk routes in the initial route set of a route list is defined as being eligible for OHQ.
- | The NCOS of the call originator (at a Meridian SL-1 ESN Node or a Meridian SL-1 ESN Main) is defined to permit OHQ.
- | The incoming trunk group at the Meridian SL-1 ESN Node or Meridian SL-1 ESN Main is defined in software to permit OHQ for incoming calls.
- | The TCM received at the Meridian SL-1 ESN Node from an ETN switch is compatible with a FRL that is defined to permit OHQ.
- | The TCOS received at the Meridian SL-1 ESN Node from another Meridian SL-1 ESN Node is compatible with an FRL that is defined to permit OHQ.
- | The probability that the call will timeout in the OHQ before a trunk becomes available is below a specific threshold.
- The OHQ feature is enabled for the customer.

Note: A Meridian SL-1 defined initial set 'marker' determines which route list entries are in the initial route set. Typically, the initial route set contains the cheaper routes to a particular destination. The remaining routes in the route list (if any) comprise the extended route set and are usually the more expensive routes to the destination. Only routes in the initial route set should be defined to allow OHQ. OHQ will not be offered by routes in the extended route set, even if they are defined to allow OHQ.

7.03 Calls which do not meet the preceding requirements for OHQ eligibility, may be offered CBQ at this point.

AVAILABILITY

7.04 The OHQ software performs an 'availability' test to prevent calls from entering the OHQ when there is a high probability that the call will timeout before a trunk becomes available. The following procedure is used to make this test.

- (1) For each trunk route, a counter is maintained which reflects the current number of calls with the maximum queue priority of 3 queued against it. This includes all calls in OHQ, Ring Again and those CBQ calls that are currently at priority 3.



Note: Priority Queuing. A maximum priority (0, 1, 2 or 3) and a starting priority (0, 1, 2 or 3) are assigned to each user through Overlay 87. Zero is the lowest priority while three is the highest. Calls are queued according to their starting priority and move to a higher priority up to their maximum priority according to a programme promotion timer.



- (2) Each trunk route has a threshold value which indicates the maximum number of priority 3 calls that can be queued against it before OHQ timeout becomes a high probability. Before a call is placed in the CBQ, the current queue count is compared with the threshold value for each eligible trunk route in the initial set of routes. If at least one of the trunk routes has a count which is less than or equal to the threshold value, the call is allowed to OHQ against all OHQ-eligible routes.

OFFER

7.05 If all eligibility and availability checks were successful, the call originator is given a distinctive OHQ offer tone (a 1-s burst of 440-Hz tone) when the call enters the OHQ. The call originator either accepts the OHQ offer by remaining off-hook, or declines the offer by going on-hook.

7.06 OHQ calls are placed in a priority-ordered queue with all other calls waiting for trunks. OHQ calls are assigned the maximum priority (3), since other network facilities may be held while the call is queued. A timer with an initial value equal to the software-defined OHQ time limit is started to limit the duration of the OHQ. (The OHQ time limit is service changeable within the range of 2 to 60 s.)

7.07 The queue is examined whenever a trunk becomes idle in a trunk route on which one or more calls are queued. If a call is found which can be terminated on an idle trunk, the available trunk is seized, and the call terminated.

7.08 If the OHQ timer expires before the call can be terminated, the call is removed from the OHQ. At this time, the remaining eligible routes in the route list (the extended set) are examined, and the call is either processed or given standard blocking treatment if no facilities are available. (CBQ will not be offered at this point as OHQ was already offered.) The ERWT is not given to calls which have been queued, even if terminated on expensive facilities.

7.09 OHQ can be offered to call originators located at a Meridian SL-1 ESN Node, Meridian SL-1 ESN Main, Conventional Main or ETN switch. Also, as a call progresses through the network, OHQ can be offered to the call originator from any of the Meridian SL-1 ESN Nodes or Meridian SL-1 ESN Mains that are used to process the call (i.e., OHQ can be offered more than once for a given call).

FEATURE INTERACTIONS

7.10 The OHQ feature has the following interactions with existing Meridian SL-1 features:

- (a) Call Modification. Station users are not allowed to activate call modification features (hold, call transfer, conference) while waiting in the OHQ. Switchhook flash used to activate features from 500/2500 sets is ignored. Similarly, operation of Meridian SL-1 set feature keys is ignored.

- (b)** Camp-On, Call Waiting. If the Meridian SL-1 attendant extends a call to a station that is in the OHQ, the call-waiting tone is not offered to the station. If the attendant releases, the call is camped-on the OHQ station, but no warning tone is given. When the camp-on call is recalled to the attendant console, the attendant can repeat the camp-on procedure. Once the OHQ call is in an established state, the camp-on tone is provided.
- (c)** Attendant Functions:
- | The Meridian SL-1 attendant cannot barge-in during trunk seizure for OHQ calls.
 - | If the Meridian SL-1 attendant extends a network call for a station user and the call is offered OHQ, the attendant must inform the caller of the OHQ offer before releasing from the call.
 - | The attendant is not allowed to operate the release key or another loop key if the source call is in conference and the destination call is in the OHQ. Operation of the release destination key is permitted, however, and causes the OHQ call to be abandoned.

8. CALL-BACK QUEUING

8.01 The CBQ is a Meridian SL-1 ESN Node feature which provides queuing for network calls which encounter busy or blocked facilities at the Meridian SL-1 ESN Node. CBQ enables the calling party to go on-hook, after activation of the Meridian SL-1 Ring Again feature (553-2001-105), and receive a callback from the Meridian SL-1 ESN Node when a network facility becomes available.

8.02 The CBQ feature is available only to stations located at a Meridian SL-1 ESN Node. Unlike OHQ, CBQ is offered only at the originating Meridian SL-1 ESN Node. Access to CBQ is accomplished through the existing Meridian SL-1 Ring Again feature.

OPTIONS

8.03 Two options for CBQ eligibility are defined by the call originator's NCOS. The first option, **CBQ(i)**, means that CBQ may be offered after only the initial route set of a route list has been examined for an available route. The second option, **CBQ(a)**, means that CBQ may be offered after both the initial and extended (i.e., all) route sets of a route list have been examined. In either case, a call offered CBQ is queued initially against only the initial route set.

ELIGIBILITY

8.04 Before offering CBQ to a call originator, the following eligibility tests are performed.

- (a) At least one of the routes in the initial route set is defined as CBQ eligible.
- (b) The user's NCOS is defined as permitting CBQ; either **CBQ(i)** or **CBQ(a)**.
- (c) The call is not eligible for OHQ. Calls that are eligible for both OHQ and CBQ will be offered OHQ.
- (d) The user's telephone is allowed access to the Ring Again feature and does not have another CBQ or Ring Again call already in the queue.
- (e) The CBQ feature is enabled for this customer.

8.05 **CBQ(i)** Eligible. For call originations by a caller defined as **CBQ(i)** eligible, the system searches the initial route set for an available route. If no available route is found, CBQ is offered to the caller subject to the CBQ eligibility tests.

8.06 **CBQ(a)** Eligible. For call originations by a caller defined as **CBQ(a)** eligible, the system examines the initial route set for an available route. If no available route is found, the extended route set is then searched for an available route. If an available route is not found in the extended set, then CBQ is offered, subject to the CBQ eligibility tests. However, if an available route in the extended route set is found that is designated as expensive, and the user's NCOS allows ERWT, the tone is given and the system delays terminating the call. During this delay the user has the following options.

- (a) Refuse the expensive route by abandoning the call.

- (b) Wait, and allow the call to complete over the expensive route.
- (c) Activate the Ring Again feature (feature key or access code) to place the call in the CBQ. The user must be **CBQ(a)** eligible; otherwise, operation of the Ring Again feature is ignored.

OFFER

8.07 The CBQ offer consists of an optional recorded announcement, followed by overflow tone. If the station user wishes to accept the CBQ offer, Ring Again must be activated within 30 s. Ring Again activation follows present feature operation for Meridian SL-1 and **500/2500** sets (553-2001-105). The CBQ offer can be refused by going on-hook. If the station user neither accepts nor refuses the CBQ offer within 30 s, the call is force disconnected.

8.08 CBQ calls are placed in a priority-ordered trunk queue (together with OHQ calls, if any) with a starting priority and maximum priority (see Overlay 87) defined by the call originator's NCOS. (Refer to Of f-Hook Queuing for additional information on priority queuing.) At the same time, two timers are started - a queue promotion timer and a route advance timer - each with values defined through the originator's NCOS. At intervals defined by the queue promotion timer, the priority of the call is incremented until it reaches its maximum priority. Each time the call priority is incremented, its position in the CBQ is advanced. If the route advance timer reaches its maximum value before the call can be terminated on a route in the initial set, the extended set or routes is added to the routes that the call is currently queued against.

8.09 Expensive route warning tone is not given to calls which have been queued, even if terminated on expensive facilities. Unless cancelled by the call originator, CBQ calls remain in the queue until they have been offered a trunk: there is no time limit on CBQ calls.

CALLBACK

8.10 When a trunk becomes available for a CBQ call, it is seized to prevent incoming originations during the CBQ callback period. Outpulsing of digits (either those originally dialed by the user or those required as a result of digit manipulation) is started at a slow, fixed rate. The number or digits to be outpulsed determines how long the trunk can be held while the CBQ callback is being offered to the originating station. The system computes this time by allowing 10 s before the first digit is outpulsed and 2.56 s between subsequent digits.

8.11 The originator of the call is alerted to the CBQ callback by either tone buzzing and a winking Ring Again feature lamp (Meridian SL-1 set), or short bursts of ringing (**500/2500** set). The Meridian SL-1 set user must accept the callback within the computed value of outpulse time or the service-changeable CBQ time limit of 10 to 30 s, whichever is less. A user with a **500/2500** set must accept the callback within 6 s. Acceptance of the CBQ callback is performed with present Ring Again operating procedures. (When a CBQ callback is answered at a digit display Meridian SL-1 set, the original dialed digits are displayed.) If the user does not answer the callback within the time limit, the call is removed from the queue and discarded. If the user accepts the callback within the time limit, the call is terminated. A CBQ call can be cancelled by the originating station via the existing procedures for Ring Again cancellation.

FEATURE INTERACTIONS

8.12 The CBQ feature interacts with the following Meridian SE-1 features.

- (a) Barge-h, Force Disconnect. Between the time a trunk is seized for a CBQ call and the user accepts the CBQ callback, the trunk can be stolen by the attendant or force disconnected through service change. If this occurs, there is no guarantee that the call can be terminated when the user accepts the CBQ callback. Under these circumstances, the call is treated like a new origination and **NARS/BARS** is used to reattempt termination. This can result in the call being blocked and being offered CBQ a second time.
- (b) Hunting, Call Forward, Multiple Appearance ON. CBQ callbacks to stations at a Meridian SL-1 **ESN** Node are offered only to the originating station, regardless of the hunting or call forwarding which may be in effect. Other appearances of a station's Directory Number (**DN**) are not offered the callback.
- (c) Attendant Functions. Because the Meridian SL-1 Ring Again feature is not supported at Meridian SL-1 attendant consoles, CBQ is not offered to an attendant regardless of the CBQ eligibility of the NCOS assigned to the attendant.

9. COORDINATED CALL-BACK QUEUING

9.01 The CCBQ feature enables stations at a Meridian **SL-1 ESN Main** to be offered CBQ when network calls are blocked at the serving Meridian SL-1 ESN Node. When facilities become available at the Meridian SL-1 ESN Node, the call originator at the Meridian SL-1 ESN Main is alerted by a callback from the node. (This feature requires that the **ESN Main** and associated **ESN Node** be equipped with the Network Signaling feature and CCBQ feature.) ←

ELIGIBILITY

9.02 When a station at an **ESN Main** originates a network call through an ESN Node, the NCOS of the call originator, call type, and whether or not the station is allowed access to the Ring Again feature is transmitted to the ESN Node. (If an authcode is entered at the ESN Main prior to dialing a network call, the NCOS associated with the authcode is transmitted to the **ESN Node**.) When received by the node, this NCOS is used to determine CCBQ eligibility and is used for the duration of the call, unless further modified by the Authcode Conditionally Last feature.

9.03 The CBQ eligibility tests are performed. In addition, a check is made that the incoming trunk group from the Meridian SL-1 ESN Main is defined (at the Meridian **SL-1 ESN Node**) to permit CBQ, and that the call type allows CBQ. CCBQ is offered to the user at the ESN Main if the eligibility tests are successful. If the tests are unsuccessful, standard call blocking is applied to the call.

9.04 As for stations at an ESN Node, the call originator at an ESN Main can invoke Ring Again upon receipt of ERWT if the originator's NCOS is defined, at the ESN Main, as **CBQ(a)** eligible.

OFFER

9.05 The CCBQ offer and acceptance sequences are identical to those for stations at the ESN Node (see Call-Back Queuing). The optional recorded announcement and overflow tone are provided by the ESN Node. In addition, after the recorded announcement is provided, the ESN Node transmits a signal to the Meridian SL-1 **ESN Main**. This signal indicates that the call is in a state which allows Ring Again.

9.06 When the call originator at the ESN Main activates Ring Again, the ESN Main assigns a unique 'queue identification' number to the call. This number is transmitted to the ESN Node to indicate CCBQ acceptance. At the Meridian SL-1 ESN Main, the call is placed in a holding queue. At the ESN Node, the call (together with the queue identification number) is placed into the trunk queue. The **ESN Main** to ESN Node tie trunk is released.

CALLBACK

9.07 When an outgoing trunk is seized by the Meridian SL-1 **ESN Node** for a CCBQ call, slow outpulsing is started to hold the trunk while a callback is made to the call originator at the Meridian SL-1 **ESN Main**. The **ESN Node** seizes an available tie trunk (Note) to the **ESN Main**, and transmits the 'queue identification' number of the call to the **ESN Main**. The **ESN Main** then initiates a callback to the call originator. Callback presentation to the call originator is as for standard Ring Again (see Call-Back Queuing).

Note: If no tie trunks to the Meridian SE-1 ESN Main are idle, the outgoing trunk is released and can be offered to another call. The CCBQ call retains its position in the queue but is not offered another trunk until a tie trunk to the ESN Main becomes available.

9.09 When the call originator at the ESN Main accepts the CCBQ callback, answer supervision is sent from the ESN Main to the ESN Node. The ESN Node then completes the call.

9.09 If the call originator is equipped with a 500/2500 set and is engaged in a call when the Meridian SL-1 ESN Node initiates a CCBQ callback, a signal is transmitted from the ESN Main to the Meridian SL-1 ESN Node. The Meridian SL-1 ESN Node releases the outgoing trunk and places the CCBQ call into a holding queue for 5 min. No attempt is made to seize another outgoing trunk for the call until the holding time expires. This process occurs only once. If the originating station is still busy, the CCBQ is cancelled automatically at the Meridian SL-1 ESN Node. No indication is given to the call originator of the CCBQ cancellation. To prevent the CCBQ call from remaining indefinitely in the holding queue at the Meridian SL-1 ESN Main, the ESN Main sets a time limit of one hour for CCBQ calls. When this time limit expires the CCBQ call is cancelled automatically. (CCBQ callback to a busy Meridian SL-1 station is as for normal Ring Again.)

9.10 The call originator at the Meridian SL-1 ESN Main can cancel the CCBQ call at any time. The Meridian SL-1 ESN Node is not aware of the cancellation until the CCBQ callback is attempted.

FEATURE INTERACTIONS

9.11 The CCBQ feature interacts with the following Meridian SL-1 features:

- (a) Meridian SL-1 Initialize. If the Meridian SL-1 Main initializes while calls are queued at the Meridian SL-1 Node, CCBQ callbacks from the Meridian SL-1 Node are not answered because the initialization has cleared the holding queue at the Meridian SL-1 Main. The Meridian SL-1 Node treats these calls as callback-no-answer calls and cancels the CCBQ automatically. If the Meridian SL-1 Node initializes, CCBQ calls in the trunk queue are lost. The Meridian SL-1 Main cannot detect this situation. To prevent calls from remaining indefinitely in the holding queue at the Meridian SL-1 Main, the Meridian SL-1 Main sets a time limit of one hour for CCBQ calls. If a callback from the Meridian SL-1 Node is not received within one hour, the Meridian SL-1 Main cancels the CCBQ call automatically.
- (b) Attendant Functions. Attendants at a Meridian SL-1 Main are not offered CCBQ. Attendant barge-in on trunks involved in CCBQ operations results in cancellation of the CCBQ call.
- (c) AIOD and ANI. Automatic Identification of Outward Dial (AIOD) and Automatic Number Identification (ANI) facilities can be used to complete CCBQ calls from a Meridian SL-1 ESN Node. The outgoing toll call is billed to the access tie trunk rather than the station at the Meridian SL-1 ESN Main.

COORDINATED
CALL-BACK QUEUING
AGAINST MAIN

9.12 The Coordinated Call-Back Queuing Against Main (CCBQAM) feature enables stations at Meridian SL-1 Nodes to be offered CBQ for network calls which are blocked at a Meridian SL-1 Main. When facilities become available at the Meridian SL-1 Main, the call originator at the Meridian SL-1 Node is alerted by a callback from the Meridian SL-1 **Main**. CCBQAM otherwise functions identically to CCBQ at the Meridian SL-1 node.

10. CALL-BACK QUEUING TO CONVENTIONAL MAINS

- 10.01 The CBQCM feature enables call originators at a Conventional Main to access the CBQ feature at a Meridian SL-1 ESN Node.
- ELIGIBILITY
- 10.02 When a station at a Conventional Main originates a network call through a Meridian SL-1 ESN Node, the NCOS assigned to the incoming trunk group is used to determine the CBQCM eligibility. This NCOS, as well as the incoming trunk group, must be defined as CBQ eligible.
- Note: If the call originates from an ETN switch, the TCM transmitted to the Meridian SL-1 Node by the ETN switch, must be compatible with the NCOS of tie at the serving Meridian SL-1 ESN Node.
- OFFER
- 10.03 The CBQCM offer to the call originator at a Conventional Main consists of an optional recorded announcement followed by special (interrupted) dial tone. (The announcement and tones are provided from the Meridian SL-1 ESN Node.) To accept the CBQCM offer, the call originator dials the extension number associated with the telephone being used for the call. When the last digit of the extension number is dialed, a confirmation tone (three, 256-ms bursts of dial tone) is sent from the Meridian SL-1 ESN Node to the call originator. The call is placed in the CBQ at the ESN Node when the call originator goes on hook.
- 10.04 The CBQCM offer can be refused by going on-hook any time before the last digit of the extension number is dialed, or by remaining off-hook for longer than 30 s after receipt of the confirmation tone. If the CBQCM is neither accepted nor rejected within 30 s, the caller is given overflow tone (from the Meridian SL-1 ESN Node) and the call is disconnected.
- CALLBACK
- 10.05 When an outgoing trunk becomes available at the Meridian SL-1 ESN Node, it is seized and slow outpulsing is started. The Meridian SL-1 ESN Node then seizes a tie trunk (Note) to the Conventional Main and outpulses the extension number of the call originator. The call originator must answer the callback before slow outpulsing is completed, otherwise, the callback is cancelled and the outgoing trunk is released.
- Note: If no tie trunks are currently available to the Conventional Main, the Meridian SL-1 Node releases the outgoing trunk. The CBQCM call retains its position in the queue but is not offered another outgoing trunk until a tie trunk to the Conventional Main becomes available.
- 10.06 When the call originator answers the CBQCM callback, answer supervision must be transmitted from the Conventional Main to the Meridian SL-1 ESN Node. Upon receipt of answer supervision from the Conventional Main, the Meridian SL-1 ESN Node transmits a tone (three, 256-ms bursts of dial tone) to notify the call originator that the call is a CBQCM callback, and completes the call.



10.07 If the call originator's station is busy when the callback is placed, the Meridian SL-1 ESN Node places the call in a suspended state for 5 min. After 5 min, another callback is attempted if the outgoing trunk is free. If the station which originated the call is still busy, the Meridian SL-1 ESN Node cancels the callback. If the call originator does not answer when the callback is placed, the CBQ is cancelled and a "no answer" condition is pegged on NTRF studies.



10.08 No provision is made for CBQCM cancellation by a call originator at a Conventional Main. Once the CBQCM offer is accepted, the call remains in the queue until the Meridian SL-1 ESN Node initiates a callback.

FEATURE
LIMITATIONS

10.09 Station users at Conventional Mains cannot activate Ring Again to refuse expensive routes after the ERWT is given.

10.10 The Meridian SL-1 ESN Node seizes the same tie trunk group that was used to initiate CBQCM for the CBQCM callback. Thus, these trunk groups must be two-way (incoming/outgoing).

10.11 Conventional Mains must provide answer supervision on tie trunks connected to the Meridian SL-1 ESN Node. These switches must also permit transmission or repetition of station dial pulses for CBQCM operation. This feature cannot be used with systems that operate in senderized mode. Operation may require adjustment of the interdigit timeout on systems that employ simulated cut-through operation.

10.12 Multiple callback queues are allowed per trunk group for the Conventional Main by dialing any digits (up to seven) based on the availability of Meridian SL-1 call registers.

10.13 Conventional Mains must not allow CBQCM callback calls to be modified by call transfer or call forward to an outside line. Such call modification can result in the tie trunk not being released at the end of the call.

11. NETWORK AUTHORIZATION CODES

11.01 The Authorization Code feature enables selected users to temporarily override the access restrictions assigned to a station or trunk. A user can enter an authorization code (authcode) to access more of the system facilities than would normally be allowed to the particular station or trunk because of the assigned Network Class of Service (NCOS), Class of Service (COS), and Trunk Group Access Restriction (TGAR) codes.

11.02 This feature is useful when a user initiates a call from someone else's telephone set and requires access to more system facilities (e.g., access to long distance calling) than are allowed to that set. Entering a valid authcode enables the user to access these additional features. Once a valid authcode is entered, the NCOS, COS, and TGAR associated with the authcode, replace the NCOS, COS, and TGAR associated with the telephone set, for the duration of the call.

11.03 The Network Authorization Code feature provides the customer with two package options:

- 1 Basic Authorization Codes (BAUT), for general applications
- 1 Network Authorization Codes (NAUT), for network applications.

BASIC
AUTHORIZATION
CODES

11.04 The BAUT package provides for up to 4096 authcodes of 1 to 14 digits. Users can enter an authcode after dialing the Special Service Prefix (SSP) and the digit '6', before dialing any call, including a NARS call. With the BAUT package, an authcode can be entered when:

- 1 originating a call from a local station or tie trunk
- 1 initiating a call transfer or conference from a local station
- 1 originating a call via the Direct Inward System Access (DISA) feature.

NETWORK
AUTHORIZATION
CODES

11.05 The Network Authorization Codes (NAUT) package provides for up to 20,000 authcodes of 1 to 7 digits. The NAUT package incorporates all the features of the BAUT package, adds a 'conditionally last' option for entering an authcode after dialing a NARS call, and allows the Meridian SL-1 attendant to enter an **authcode**.

AUTHCODE
CONDITIONALLY LAST

11.06 With the NAUT package, users can be prompted 'conditionally' for an **authcode** after dialing a **NARS/BARS** call. The prompt is by an 'authcode request' which consists of 10 bursts of dial tone, followed by steady dial tone. (The authcode request can, optionally, be preceded with an appropriate recorded announcement.) The user is prompted for an authcode entry only if:

- 1 an authcode was not previously entered
- the **FRL** associated with the user's NCOS is less than the service change assigned minimum FRL of the route list that **NARS/BARS** would use for the call.

11.07 Users at a Meridian SL-1 ESN Main or Conventional Main, connected via tie trunks to a Meridian SL-1 ESN Node, can (optionally) be prompted for an authcode entry after dialing a **NARS/BARS** call. The user is prompted for an authcode entry only if:

- no authcode was previously entered
- the FRL associated with the NCOS of the incoming (or two-way) tie trunk is less than the minimum FRL of the route list that **NARS/BARS** would use for the call
- the route is defined in the Route Data Block (**RDB**), overlay program 16, to prompt for an authcode entry on incoming NARS calls.

11.08 Users accessing a Meridian SL-1 ESN Node via the DISA feature to make a NARS call are prompted for an authcode entry if:

- no authcode was previously entered
- the FRL of the NCOS assigned to the DISA Terminal Number (TN) is less than the minimum FRL of the route list that **NARS/BARS** would use for the call.



ATTENDANT INPUT OF AUTHCODE

11.09 Normally, because a Meridian SL-1 attendant is not restricted from accessing any system resource, there is no need for the attendant to have an authcode. The Network Authorization Code package enables the Meridian SL-1 attendant to enter an authcode for other callers: i.e., the attendant can enter an authcode (after dialing the SSP and the digit '6') and complete a long distance call for a local station user whose COS is toll denied (TLD). If the CDR of authcodes is defined for the customer, the local station user's authcode digits appear in the CDR record for billing purposes.

11.10 Meridian SL-1 attendants are normally assigned an NCOS having a high FRL so that they can make any type of call, including **NARS** calls. An attendant can, however, be prompted for an authcode entry, if the FRL required to access a route list for a **NARS/BARS** call is greater than the FRL of the attendant's NCOS.

AUTHCODE VALIDATION

11.11 The Meridian SL-1 software validates an inputted authcode on the basis of the number of digits dialed and the dialed digits themselves. If the inputted authcode contains more/less digits than the defined authcode length (Authcode Data Block - **AUB**, overlay **88**), the authcode is deemed invalid. Similarly, if the dialed authcode digits are not defined in the Authcode Table (**AUT**, overlay **88**), the authcode is deemed invalid. When an invalid authcode is encountered, no response is given to the user until the Meridian SL-1 End-of-Dialing (**EOD**) timer expires. (This increases the security of authcodes by making it difficult for an unauthorized user to determine the length of a valid authcode.) When the EOD timer expires, overflow tone is given for 15 s and the call is force disconnected.

AUTHCODE
ADMINISTRATION

11.12 With the NAUT and BAUT packages, a **'classcode'** structure is used as part of authcode administration. A classcode is a definition of a combination of COS, TGAR, and NCOS codes. There can be up to 116 (O-115) classcodes defined through the Authcode Data Block (**AUB**, overlay **88**), each having a different combination of COS, TGAR, and NCOS codes. Each authcode is associated with a classcode. Authcodes which have the same combination of COS, TGAR, and NCOS codes are assigned the same classcode.

11.13 With the NAUT package, authcodes can be defined individually by the customer, or generated automatically by the Meridian SL-1. When defining or generating new authcodes, a classcode with which the new authcodes are to be associated is specified. The new **authcode** will then be automatically assigned the COS, TGAR, and NCOS codes associated with the specified classcode.

11.14 When an authcode is to be removed from use, a **facility** exists to prevent that authcode from being reused (i.e., the authcode will not be accepted as valid input when individually defining authcodes). This is accomplished through an **'exemptcode'**. When an authcode is removed from use, an **'exemptcode'** is assigned to the authcode in place of the classcode. The exemptcode is the month (e.g., JAN, FEB, etc.,) taken from the Meridian SL-1 system clock. If an exemptcode is not requested, the removed authcode is returned to the pool of unused authcodes, and can be reused at any time.

11.15 The Route List Block (**RLB**) program (overlay **86**) is used to define a minimum FRL for each route list. This minimum FRL (range O-7) is used to determine whether or not to prompt for an authcode entry after a call. If a minimum FRL is not specified, the actual minimum FRL in the initial route set is used. Similarly, the Route Data Block (**RDB**) program (overlay **16**) is used to define whether or not to prompt for an authcode entry on calls on incoming or two-way tie trunk groups.

FEATURE OPERATION

11.16 Authcode after SSP (**500/2500/Meridian** SL-1 Sets). To enter an authcode after the Special Service Prefix (**SSP**), the caller proceeds as follows:

- (1) If there is no call in progress, go off-hook or press a DN key. If there is a call in progress, switchhook flash (**500/2500** set) or press the call transfer or conference key (Meridian SL-1 set) to obtain special dial tone.
- (2) Dial the authcode access number (**SSP** and the digit **'6'**). Dial tone is removed after the SSP digit is dialed.
- (3) Dial the authcode digits. A second dial tone is heard if the authcode is valid. If the authcode is invalid, no response is given until the EOD timer expires, then overflow tone is given for 15 s and the call is force disconnected.
- (4) When the second dial tone is heard, dial the call in the normal manner. If call transfer/conference is in effect, complete the transfer/conference as normal.

11.17 Authcode After SSP (Meridian SL-1 Attendant). To enter an authcode after SSP, the attendant proceeds as follows:

- (19) If there is a call on the source loop, proceed to (29). If there is no call on the source loop, press an idle loop (**LPK**) key.
- (2) Dial the authcode access number (**SSP** and the digit '6'), followed by the authcode.
- (39) Dial as usual after receiving dial tone denoting a valid authorization code. (If the code is invalid, overflow tone is returned immediately.)

11.18 Authcode Conditionally Last. The following procedure is used to enter an authcode conditionally last from a **500/2500/Meridian SL-1** set or attendant console (**NAUT package only**):

- (19) Dial a call.
- (2) Receive an 'authcode request' (optional recorded announcement followed by 10 bursts of dial tone, followed by steady dial tone), indicating that an authcode entry is required.
- (39) Dial the authcode. Dial tone is removed after the first digit is dialed. If the authcode is valid, the call is processed as a normal call. If the authcode is invalid, overflow tone is returned when the EOD timer expires.

FEATURE
INTERACTIONS

11.19 Feature Key Operations.

- (a9) While a user is entering an authcode, the following feature keys operate as intended, and do not affect operation of the authorization code feature:
 - | Make Set Busy
 - . Buzz
 - Volume Control.
- (b) The operation of the following keys is ignored during authcode operation:
 - | Conference
 - | Override
 - | Call Forward and Call Transfer
 - Call Pickup
 - | Charge Account
 - Calling Party Number
 - | Privacy Release
 - Ring Again
 - | Barge-In and Busy Verify

- | speed call
 - | Recall
 - | Do Not Disturb
 - | Digit Display.
- (c) The following key operations about the authcode operation, and any digits entered for an authcode are ignored:
- | Directory Number
 - Paging
 - Voice Call
 - | Not Ready
 - | In-Calls
 - | Call Waiting
 - | Hold
 - | Release.
- (d) If the caller initiates a switchhook flash while entering an authcode, the results are unpredictable: the switchhook flash may be ignored, or interpreted as the digit '1'.
- (e) Authcodes after SSP can be stored as speed call or auto-dial entries. When this is done, the stored number (entry) must contain only the access code and authcode digits. All digits in the entry after the access code are interpreted as authcode digits. In the case of authcode conditionally last, authcodes can be stored as auto-dial entries, but not speed call entries. If necessary, the caller can continue to enter more authcode digits after operation of the auto-dial or speed call key. However, for security reasons, authcodes should not be stored as auto-dial or speed call entries.

11.20 Call Detail Recording. If the CDR recording of authcodes is specified, then each time an authcode is entered, a record is generated on **the** CDR device. The record is passed to CDR only if one of the following occurs:

- | the call becomes established (e.g., trunk is seized or local set answers)
- | the call cannot be completed (e.g., no trunks available)
- | the Ring Again feature is applied to the call.

11.21 Authcode Input Via Tie Trunks. **Authcodes** can be entered via access tie trunks. Incoming or two-way tie trunk groups at a switch equipped with the Network Authorization Code feature can be defined to prompt for an authcode entry.

↗

11.22 Direct Inward System Access. If a caller makes a NARS/BARS call in association with a valid DISA call, the NCOS associated with the DISA DN is used for NARS/BARS route selection. If the FRL of this NCOS is too low to access the route list that NARS/BARS has selected for the call, the caller will be prompted for an authcode entry, unless an authcode (i.e., Authcode after SSP) was entered previously.

↘

11.23 Barge-in/Busy Verify. If barge-in or busy verify is used by the Meridian SL-1 attendant to break into a connection where an authcode is being entered, the authcode entry will be affected. If the authcode entered is invalid as a result, the user will be given overflow tone when the EOD timer expires.

11.24 Centralized Attendant Service. The Central Attendant Service (CAS) feature enables several remote switches to share the attendant services at one central location (553-2681-100). A CAS attendant can enter an authcode, via a Release Link Trunk (RLT), before connecting or transferring calls to the connecting remote PBX. If the CAS attendant enters a NARS number via an RLT, the NCOS associated with the NCOS of the attendant at the remote PBX) is used in the NARS route selection process. If the FRL of this NCOS is inadequate, the CAS attendant may be prompted for an authcode entry.

11.25 Call Forwarding. The Meridian SL-1 call forwarding feature provides two customer options: call forwarding-originating party's COS (CFO) or call forwarding-forwarding party's COS (CFF). With the NAUT package and the CFO option, a caller may be prompted for an authcode entry after a call to a station which forwards the call to a NARS number. With the CFF option, the user will not be prompted by the local switch for an authcode entry after such a call.

11.26 Network Class of Service. An NCOS is assigned to each station and each incoming tie trunk at a Meridian SL-1 ESN Node. An authcode entry modifies the user's NCOS for the duration of the call. The FRL associated with the user's assigned NCOS is used to determine if it is necessary to prompt for an authcode entry. After an authcode is collected and validated, the NCOS associated with the authcode is used for the duration of the call.

→

11.27 Network Alternate Route Selection. During NARS/BARS route selection, the FRL associated with the call originator's NCOS is compared with the FRL of the selected route list. If the originator's FRL is lower and no authcode was entered previously, the system will prompt for an authcode entry. A valid authcode modifies the originator's NCOS and hence, FRL. This new FRL is then used for route selection.

11.28 Network Queuing. When an authcode is entered, the NCOS associated with the authcode is used to determine Network Queuing capabilities.

11.29 Coordinated Dialing Plan. Authcode after SSP can be used before dialing a CDP call. If the NAUT package is equipped, the 'conditionally last' request for an authcode entry applies.

LIMITATIONS

11.30 Users on **PBX** or **Centrex** systems connected via tie trunks to a Meridian SL-1 ESN Node, can use the authcode conditionally last feature, provided that these systems transmit or repeat all digits dialed by the users in response to the authcode request. This feature cannot be used by certain systems that operate in senderized mode. Correct operation may require adjustment of EOD timeout on systems that employ simulated cut-through operation. End-to-end signaling software can be used to guarantee operation. ←

11.31 In an ESN network consisting of multiple Meridian SL-1 switches equipped with the Network Authcode package, authcodes should be requested only once on a given call. This requires careful engineering of:

- the tie trunk group option for authcode prompting
 - ┆ the FRL values assigned to route lists.

11.32 Users at a Meridian SL-1 ESN Main or Conventional Main, arranged for the UDP via a dedicated trunk group to a Meridian SL-1 ESN Node, can use the authcode conditionally last feature at the ESN Node in the same manner as those stations located directly at the ESN Node. However, these users cannot access the authcode after SSP feature via the same trunk group.

12. COORDINATED DIALING PLAN

12.01 The CDP feature enables a customer having a number of local switches, to coordinate the dialing plan of the stations at these switches. The CDP feature enables a station user to call any other station within the CDP group of switches by dialing a unique 3- to 7-digit number assigned to the station. A CDP can be arranged to provide a centralized public exchange network capability which channels access to and from the public network through a single Meridian SL-1 switch within the CDP group.

12.02 The CDP software provides the translation and digit manipulation capability that is necessary to implement the coordinated dialing plan. Calls dialed within the CDP format can be terminated locally after digit translation and digit deletion or, alternatively, calls can be routed to a remote switch in the CDP group following digit translation, route selection, and digit deletion and/or insertion. Fig. 12-1 illustrates how a coordinated dialing plan might be implemented at two customer locations.

STEERING CODES

12.03 Referring to Fig. 12-1, users at Location D can call stations at Location E by dialing 43XXX or 52XXX. Similarly, users at Location E can call stations at Location D by dialing 2XXXX or 3XXXX. If a user at Location D dials 43XXX or 52XXX to reach a station at Location E, Location D uses the digits '43' or '52' as a DSC (Distant Steering Code) to select the trunk group to Location E. Similarly, if a user at Location E dials 2XXXX or 3XXXX to reach a station at Location D, Location E uses the digit '2' or '3' as a DSC.

12.04 The same format is used for calling local stations; e.g., users at Location E dial 43XXX or 52XXX to reach local stations at Location E. In this case, the Meridian SL-1 interprets the digits '43' or '52' as a LSC (Local Steering Code) and deletes them from the dialed number in order to terminate the call locally.

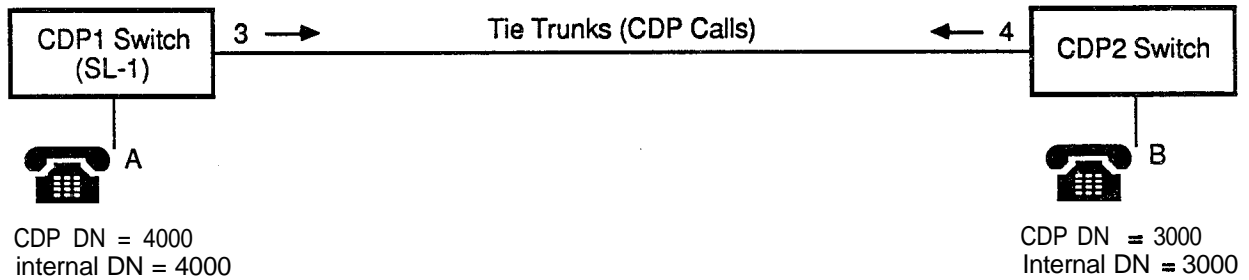
12.05 If the Meridian SL-1 switch at Location E is arranged to provide centralized access to the public exchange network, the digit '9' at Location E is a TSC (Trunk Steering Code) for public exchange access. At Location D, the digit '9' is a 'trunk steering code' which invokes digit manipulation to insert the required digits to route the call, through Location E, to the public exchange network. Similarly, users at Location D can call the attendant at Location E by simply dialing '0', if Location D does not assign the digit '0' as the local attendant access code.

12.06 Steering codes can be composed of one, two, three or four digits. At each switch in the CDP group, the steering codes must be distinct (i.e., left-wise unique) from any assigned access codes. As Fig. 12-1 shows, '0' is reserved as the attendant access code; '1' is reserved as the Special Service Prefix (SPRE); '7' is reserved as a Meridian SL-1 trunk access code; '8' is reserved as a NARS access code; and '9' is reserved as the public exchange network access code. This means there are five digits remaining that can be used as the leading digits of steering codes (i.e., '2', '3', '4', '5', and '6'). The CDP feature supports up to 5000 steering codes.

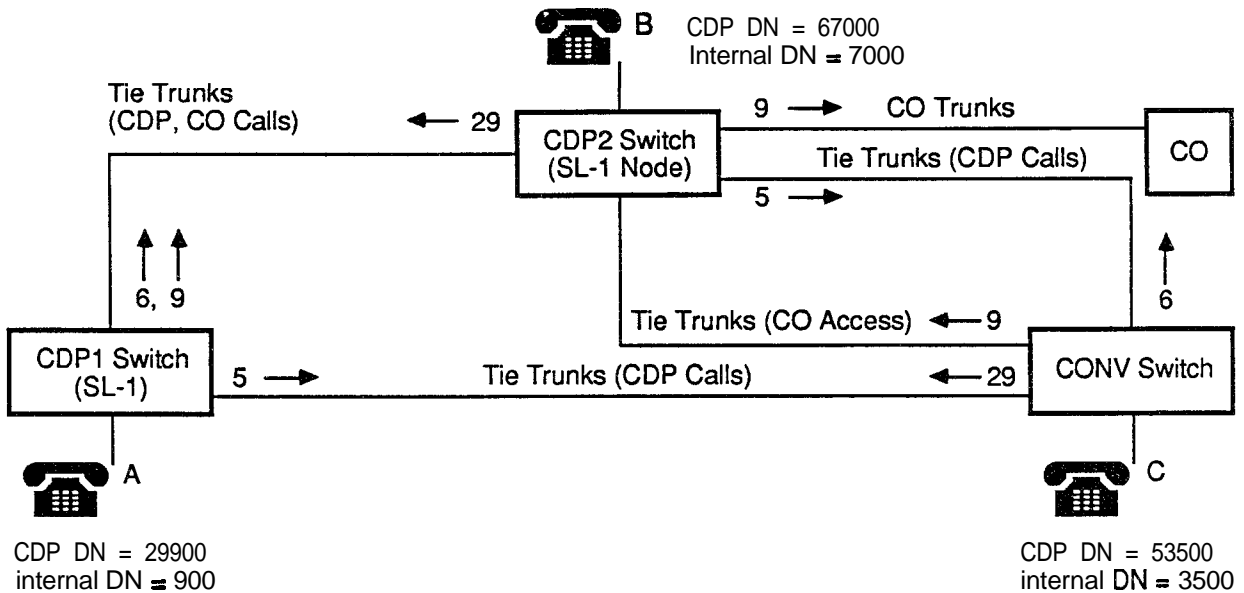
12.11 As shown in Fig. 12-2(b), switch **CDP2** is arranged to provide centralized access to the public exchange network. For users at the **CONV** switch to access this capability, a separate tie trunk route must be provided to switch **CDP2**. This is because switch **CDP2** is arranged to insert the digit '6' on the incoming tie trunk route from the **CONV** switch used for **CDP** calls. For public exchange network calls, the digit '9' must be inserted on the incoming tie trunk route from the **CONV** switch. Similarly, if users at the **CONV** switch are to be allowed access to the **ESN** capabilities (e.g., **NARS**) at switch **CDP2**, another tie trunk route must be provided for this purpose.

CDP ROUTING

12.12 Up to 32 route lists can be defined at a switch equipped with the **CDP** feature software. (If **CDP** is equipped at a Meridian SL-1 **ESN** Node, 256 route lists can be defined.) A route list is used to define the alternate route choices for **CDP** calls to a particular destination. Route choices in a route list are called route list entries. There can be up to three (0-2) route list entries associated with each route list. (If a switch is equipped with the **NARS** software in addition to the **CDP** feature software, **NARS** route lists, maximum 256, can be shared by both **NARS** and **CDP** calls.)



a) CDP DN (without steering code)



b) CDP DN (with steering code)

DN = Directory Number

Fig. 12-2
Typical CDP Configurations

12.13 Route lists are associated with each 'distant steering code' and 'trunk steering code' that can be dialed at a CDP switch. Each code is defined to the CDP software, together with the route list number that must be accessed for call completion to the destination indicated by the steering code. The entries in the specified route list are then searched sequentially for an available and eligible trunk route. Local steering codes are not associated with route lists.

CDP DIGIT MANIPULATION

12.14 Route list entries can be associated with digit manipulation tables. There can be 32 (O-31) digit manipulation tables defined at a CDP switch. If CDP is equipped at Meridian SL-1 ESN Node, 256 (O-255) tables can be defined. (Digit manipulation table 0 is used as an indication that no digit manipulation is required.) Each digit manipulation table (except 0) can be defined to delete a number (0 to 15) of leading digits of a dialed CDP number, and to insert up to 24 different leading digits, including the asterisk (*) to indicate a dialing pause, where required.

CDP TIME-OF-DAY SCHEDULES

12.15 Two (O-1) TOD schedules can be defined at a CDP switch. (If CDP is equipped at a Meridian SL-1 ESN Node, eight (O-7) TOD schedules can be defined.) Each route list entry is associated with a TOD schedule. When a route list entry is selected for a CDP call, the CDP software compares the current time-of-day with the TOD schedule assigned to the route list entry. If the current time-of-day is within the interval defined by the TOD schedule, the route list entry is used for the call; if the current time-of-day is not within the interval defined by the TOD schedule or, if the TOD schedule is 'turned OFF' (in software), the route list entry is not eligible for the call. TOD schedules can be selectively 'turned ON' or 'turned OFF' by the customer (through service change).

QUEUING

12.46 Queuing against local stations is provided by the standard Meridian SL-1 Ring Again (RGA) feature. For calls directed to a remote CDP switch, Ring Again can be applied if all local outgoing trunk routes to the remote CDP switch are busy or blocked. Ring Again cannot be applied against busy or blocked stations/trunks at the remote CDP switch. Blocking tone is not provided until the full CDP number (or trunk steering code) is dialed.

FEATURE INTERACTIONS

12.17 **NARS/BARS.** The CDP feature can be implemented at a switch equipped with the NARS/BARS software feature. If such is the case, the following considerations apply:

- steering codes for CDP calls must be distinct from the assigned **NARS/BARS** access codes
- CDP numbers can be integrated with the ESN UDP; e.g., a four-digit CDP number can be the same as the last four digits of a seven-digit UDP number
- **NARS/BARS** route lists, digit manipulation tables, and TOD schedules can be shared by CDP calls (lower CDP route lists must be numbered 0-31)
- users eligible for the OHQ and CBQ features, can use them when placing CDP calls

1 FCAS does not apply to CDP calls.

12.18 **AIOD** and **ANI**. Calls made to the public exchange network when the AIOD or ANI feature is equipped will have either the internal DN recorded if the call originates at the CDP switch interfacing to the public network, or the trunk access code if the call originates at another CDP switch.

12.19 Meridian SL-1 Attendant Features. If a user at a local CDP switch calls the local attendant, the local user's internal DN (not the full CDP DN) is displayed. If a user at a CDP switch calls a Meridian SL-1 attendant at another CDP switch, the trunk access code and member number of the incoming trunk are displayed. The following attendant features are supported at the local CDP switch, but are not supported between CDP switches:

- | automatic timed recall
- | barge-in, busy verify
- camp-on
- | interposition calling.

12.20 **COS/TGAR** Treatment. For CDP calls, all Class of Service (**COS**) treatment remains the same as standard Meridian SL-1 treatment with the exception of conditionally toll-denied (**CTD**) and conditionally unrestricted (**CUN**) COS, which are treated as unrestricted (**UNR**). Users with an FR2 class of service can make local CDP calls, but cannot make CDP calls to distant switches. Trunk Group Access Restrictions (**TGAR**) are ignored for the purpose of routing CDP calls.

12.21 Code Restriction. Meridian SL-1 code restriction is applied to calls made only from stations with a toll-denied (**TLD**) class of service. Standard or flexible code restriction can be applied, on a trunk route basis, to public exchange network trunk calls.

12.22 Call Detail Recording. The local internal DN (not the complete CDP DN) is recorded in the normal CDR manner. The maximum internal DN length remains at four digits. The full CDP DN is shown in the dialed number field.

12.23 Common Control Switching Arrangement. A CDP number can be part of a CCSA dialing plan. Digit absorption and manipulation for **CCSA** calls is handled as usual by the Meridian SL-1 switch. A **CCSA** call can terminate at a switch in a CDP group other than the switch which hosts the **CCSA** network. This operation is transparent to the originator of the **CCSA** call.

12.24 Direct Inward Dialing. Because a CDP DN can be up to seven digits long, the capability of inserting up to six leading digits on DID trunks is provided.

12.25 End-to-End Signaling. End-to-End signaling is allowed for CDP calls.

12.26 Call Modification. Call modification (e.g., call transfer, call forward, conference) is allowed for CDP calls. When using these features, the user dials within the CDP format.

12.27 Hunting. Hunting across different switches in a CDP group is not supported. Standard Meridian SL-1 hunting can be applied to local CDP calls.

12.28 Message Center. The message center capability is not supported across CDP switches, however, it operates as normal locally.

12.29 Meridian SL-1 Digit Display.

- | Outgoing CDP Call. The complete dialed CDP DN is displayed at the originating set.
- | Incoming CDP Call. The trunk access code and member number of the incoming trunk route is displayed.
- | Internal CDP Call. At the originating set, the complete dialed CDP DN is displayed. (If the call hunts or is picked up by another station, the internal DN of the answering station is displayed.) At the terminating set, the internal DN of the originating set is displayed.

13. NETWORK TRAFFIC MEASUREMENTS

13.01 The NTRF feature provides traffic measurement data related to network performance and network traffic. The NTRF feature can be equipped at Meridian SL-1 ESN Nodes and Meridian SL-1 ESN Mains. Effective use of this data enables the network communications manager to assess the effectiveness of the network, and to identify specific areas of network operation where improvements are warranted.

13.02 The network traffic measurements accumulated at a Meridian SE-1 switch equipped with the NARS (Meridian SL-1 ESN Node), BARS (Meridian SL-1 ESN Main), or CDP feature encompass the following areas of operation (in addition to regular Meridian SL-1 traffic measurements):

- Network Class of Service (NCOS)
- Routing
- Off-Hook Queuing (OHQ)
- Call-Back Queuing (CBQ)
- Coordinated Call-Back Queuing (CCBQ)
- Call-Back Queuing to Conventional Mains (CBQCM)
- Incoming Trunk Groups.

ROUTING TRAFFIC MEASUREMENTS

13.03 A route list is a list of outgoing alternate trunk routes to a specific location from a Meridian SL-1 switch. Trunk routes in a route list are termed route list entries. The number of route lists/entries that can be defined at a Meridian SL-1 switch depends on the facilities+ equipped at that switch. Table 13-A lists the parameters for these different features and feature combinations.

Table 13-A
SUMMARY OF MERIDIAN SL-1 NETWORKING FEATURE PARAMETERS

PARAMETER	FEATURES EQUIPPED AT SWITCH				
	BARS	NARS	CDP	CDP WITH BARS	CDP WITH NARS
→ NCOS Groups Facility Restriction	0-7	0-15	0-3	0-7	0-15
Levels Digit Manipulation	0-7	0-7	0-7	0-7	0-7
Tables	1-255	1-255	1-31	1-255	1-255
Route Lists	0-127	0-255	0-7	0-127	0-255
Route List Entries	0-7	0-7	0-2	0-7	0-7
FCAS Tables	1-127	1-255	-	1-127	1-255
→ SDR Tables	0-255	0-512	-	0-255	0-512
Steering Codes			1-5000	1-5000	1-5000

Note 4: NCOS = Network Class of Service
FCAS = Free Calling Area Screening
SDR = Supplemental Digit Restriction.

Note 2: If the NARS and BARS features are equipped in the same switch but for different customers, the highest parameter values apply to that switch; e.g., if one customer has NARS and another customer has BARS, the NARS parameters apply to the BARS customer.

Note 3: If the New Flexible Code Restriction (NFCR) feature (553-2001-105) is equipped in conjunction with CDP, the number of available NCOS groups is 8 (0-7).

13.04 The routing traffic measurements (TFN001) provide data related to route list utilization. The measurements show how often a route list was accessed, which entries in the list were used, and whether the call was successful in completing a selection or connection. Routing traffic measurements are available at Meridian SL-1 ESN Node and Meridian SL-1 ESN Main switches.

13.05 The routing traffic measurements contain the following statistics for each defined route list.

- (a) Route List Requests. This measurement identifies the total number of call attempts for which the called destination translations identified this route list to attempt call completion.
- (b) Route List Requests served Without Delay. This measurement reflects the total number of network calls which were routed without encountering blocking or queuing.
- (c) Expensive Route Acceptances. This measurement identifies the number of calls which were allowed to complete over an expensive trunk route after the Expensive Route Warning Tone (ERWT) was given.

- (d) Route List Requests Standard Blocking. This measurement identifies the number of call attempts which could not be served because a route or queuing process was not available to a user. The blocked call may have been routed to overflow tone, a recorded announcement, or the attendant.
- (e) Route List Entry Usage Count. This measurement identifies the number of calls which were routed successfully over a particular route (entry) in a route list. A count is maintained for each route list entry.

OHQ MEASUREMENTS

13.06 Traffic measurements for Off-Hook Queuing (OHQ) are associated with each route list and identify the utilization of the OHQ feature. The OHQ measurements are included with the routing traffic measurements (TFN001), and contain the following statistics (for each route list).

- (a) Quantity of Calls Placed in OHQ. This measurement identifies the number of calls which attempted to use a route in the route list but, because facilities were not immediately available, the call was permitted to remain off-hook to wait for facilities.
- (b) Average Time in OHQ. This measurement identifies the average duration that calls remained in the OHQ until a route became available. The value (expressed in units of 0.1 s) represents the average time in the queue. Calls which timed out in the queue before a route was selected are also included in the average.
- (c) Quantity of Calls Abandoned from OHQ. This measurement identifies the number of calls which were placed in the OHQ but were abandoned: i.e., the caller went on-hook before a route became available or the time limit was reached.

CBQ MEASUREMENTS

13.07 Traffic measurements for CBQ are associated with each route list and identify the utilization of the feature. The CBQ measurements are included with the routing traffic measurements (TFN001), and contain the following statistics (for each route list).

- (a) Quantity of CBQ Calls. This measurement identifies the number of calls which were offered CBQ and accepted the offer.
- (b) Average Time in CBQ. This measurement identifies the average duration (in units of 0.1 s) calls remained in the CBQ. Calls which were cancelled and calls which were served are included in this measurement.
- (c) Quantity of CBQ Offerings. This measurement is a count of the number of calls which were offered CBQ, regardless if the offer was accepted or not.
- (d) Quantity of CBQ User Cancellations. This measurement identifies the number of calls which were removed from the CBQ on the call originator's request (i.e., cancellation of the Ring Again feature).

ROUTING TRAFFIC
REPORT OUTPUT
FORMAT

13.08 The routing traffic measurements are outputted for each route list as shown in Table 13-B.

Table 13-B
TFNOOI ROUTING FORMAT

(System ID) (Customer Number)	TFN001						
RLST xxx	aaaaa	bbbbbb	ccccc	dddddd	eeeee	fffff	
	RT	ggggg	ggggg	ggggg	ggggg	ggggg	ggggg
		ppppp	ppppp	ppppp	ppppp	ppppp	ppppp
	OHQ	hhhhh	iiii	jjjj			
	CBQ	kkkkk	llll	mmmmm	nnnnn		

where:

- a = route list requests
 - b = route list requests served without delay
 - c -- expensive route acceptances
 - d = route list requests standard blocking
 - e = not defined (all zeros)
 - f = not defined (all zeros)
 - g = route list entry usage count for each entry in the route list
 - h = quantity of calls placed in OHQ
 - i = average time in OHQ (in units of 0.1 s)
 - j = quantity of calls abandoned from OHQ
 - k = quantity of CBQ calls
 - l = average time in CBQ (in units of 0.1 s)
 - m = quantity of CBQ offerings
 - n = quantity of CBQ user cancellations
 - p = quantity of calls involving **SL1TD** on which alternate routing was performed during the call
 - x = route list number
-

NCOS
MEASUREMENTS

13.09 Traffic measurements are collected for each defined NCOS group (0-15) to indicate the grade of service, in terms of blocking and queuing delay, being provided by the system. If a grade of service is determined by the communications manager to be inappropriate for users in a particular NCOS group, then the communications manager can either reassign the users to another NCOS group, redefine the characteristics of the existing NCOS group, or change the routing parameters. NCOS measurements are available at Meridian SL-1 ESN Node and Meridian SL-1 ESN Main switches.

13.10 The NCOS traffic measurements (TFN002) contain the following statistics for each defined NCOS group.

- (a) Quantity of Calls Attempted. This measurement identifies the total number of call attempts generated by users in an NCOS group.
- (b) Routing Requests Served Without Delay. This measurement identifies the number of call attempts which were routed without encountering blocking or queuing.
- (c) Expensive Route Acceptance. This is a count of the number of callers who accepted an expensive route to complete a call.
- (d) Network Call Standard Blocking. This measurement identifies the number of call attempts which could not be completed because a route or queuing process was not available to a user. The blocked call may have been routed to overflow tone, a recorded announcement, or the attendant.
- (e) Quantity of Calls Refusing Expensive Routes. This measurement identifies the number of calls which were given ERWT and elected not to use the expensive route.
- (f) Quantity of Calls Placed in OHQ. This measurement identifies the number of calls that were placed in the OHQ.
- (g) Average Time in OHQ. This measurement identifies the average duration that calls remained in the OHQ until a route became available. The value (in units of 0.1 s) represents the average time that calls were in the queue. Calls which timed out in the queue before a route was selected are also included in the average.
- (h) Quantity of CBQ Calls. This measurement identifies the number of calls which were offered CBQ and accepted the offer.
- (i) Average Time in CBQ. This measurement identifies the average time that calls waited in the CBQ for a route to become available. It includes calls which requested a cancellation, calls which were served, and direct Ring Again against trunks. The average time is expressed in units of 0.1 s.

NCOS
MEASUREMENTS
OUTPUT FORMAT

13.11 The network class of service measurements are outputted for each defined NCOS group as shown in Table 13-C.

Table 13-C
TFN002 NCOS - FORMAT

(System ID) (Customer Number)		TFN002					
NCOS	xxx OH- Q CBQ	aaaaa ggggg iiii	bbbbbb h h h h h	ccccc h h h	ddddd	eeeee	fffff

where:

- a = quantity of calls attempted
- b = routing requests served without delay
- c = expensive route acceptances
- d = network call standard blocking
- e = not defined (all zeros)
- f = quantity of calls refusing expensive routes
- g = quantity of calls placed in OHQ
- h = average time in OHQ (in units of 0.1 s)
- i = quantity of CBQ calls
- j = average time in CBQ (in units of 0.1 s)
- x = network class of service group.

INCOMING TRUNK GROUP MEASUREMENTS

13.12 The Incoming Trunk Group (TFN003) measurements provide an indication of the incremental traffic that was imposed on incoming trunk groups by the network queuing features. Data are provided for each incoming or two-way trunk group that is offered OHQ, CCBQ, or CBQCM. These measurements are available at Meridian SL-1 ESN Nodes.

13.13 For each incoming (or two-way) trunk group, the following measurements are accumulated:

- (a) Quantity of Calls Placed in **OHQ**. This measurement identifies the number of incoming trunk calls that were placed in the OHQ for possible connection to another trunk group.

- (b) Average Time in **OHQ**. This measurement reflects the average time (in units of 0.1 s) that calls waited in the OHQ for a trunk to become available. The average time includes those calls that were removed from the OHQ by caller abandonment, or were removed from the queue after expiration of the OHQ time limit.
- (c) Quantity of Incoming Calls Offered **CCBQ** or **CBQCM**. This measurement identifies the number of incoming trunk calls that were blocked at the Meridian SL-1 ESN Node, and the user was given the option of accepting a Meridian SE-1 ESN Node initiated callback when facilities become available. The measurement relates to use of the CBQ feature by users at a Meridian SL-1 ESN Main (Coordinated Call-Back Queuing) or Conventional Main (Call-Back Queuing to Conventional Mains).
- (d) Quantity of Calls Accepting **CCBQ** or **CBQCM**. This measurement identifies the number of incoming trunk calls that were blocked at the Meridian SL-1 ESN Node, were offered CBQ, and accepted the offer. The count relates to CBQ acceptances by users at a Meridian SL-1 ESN Main or Conventional Main.
- (e) Average Time in **CBQ**. This measurement (expressed in units of 0.1 s) reflects the average time that users at a Meridian SL-1 ESN Main or Conventional Main remained in the CBQ (at the Meridian SL-1 ESN Node) for a facility to become available.

Note 1: When a CCBQ callback is offered to a busy station at a Meridian SL-1 ESN Main, the call is removed from the queue for 5 min. then reinserted in the queue. This process occurs only once. The additional queuing time is added to the average time. The 5 min suspension time is not included in the average time, nor is it reinsertion into the queue pegged as another CBQ call.

Note 2: When a CBQCM callback is offered to a station at a Conventional Main that is busy or fails to answer the callback, the call is removed from the queue and reinserted into the queue as specified in Note 1.

- (f) Quantity of Calls Blocked in Callback. This measurement identifies the number of CBQ callbacks (**CCBQ** or **CBQCM**) initiated by the Meridian SL-1 ESN Node that could not be completed because an outgoing trunk group (to the Meridian SL-1 ESN Main or Conventional Main) was not available.
- (g) Callback Attempts No Answer and Cancellation. This measurement identifies the number of callback attempts that were not successful because the caller failed to answer the callback. CBQ callbacks to a station at a Meridian SL-1 ESN Main that has previously cancelled CBQ are treated as callback attempts no answer.

INCOMING TRUNK
GROUP
MEASUREMENT
FORMAT

13.14 The incoming trunk group measurements (**TFN003**) are outputted as shown in Table 13-D.

Table 13-D
TFN003 INCOMING TRUNK GROUP - FORMAT

	(System ID) (Customer Number)	TFN003				
TRKG	xxx OH- Q CBQ	aaaaa	bbbbbb	eeeeee	fffff	ggggg

where:

- a -- quantity of calls placed in OHQ
- b = average time in OHQ (in units of 0.1 s)
- c - - quantity of incoming calls offered CBQ (CCBQ or CBQCM)
- d = quantity of calls accepting CBQ offer (CCBQ or CBQCM)
- e -- average time in CBQ (in units of 0.1 s)
- f = quantity of blocked CBQ callbacks (CCBQ or CBQCM)
- g = quantity of callback attempts not answered or cancelled
- x = incoming trunk group number (O-127)

OHQ THRESHOLD VIOLATION MEASUREMENT

13.15 The OHQ overflow threshold measurement (TFN101) provides an indication that more than the expected number of users are timing out in the OHQ. This means that OHQ is offered and accepted, but a trunk does not become available before the service-changeable OHQ time limit expires. This could result from trunks being out of service, an incorrectly defined OHQ time limit, or temporary traffic overload.

13.16 The output format for this threshold measurement is shown in Table 13-E.

Table 13-E
OHQ THRESHOLD VIOLATION MEASUREMENT

(System ID) (Customer Number)	TFN101	
OHQT	aaaaa	bbbbb
where:		
aaaaa =	the number of OHQ calls which timed out (overflowed) in the OHQ before an available trunk was found.	
bbbbb =	the threshold. This value (expressed in units of 0.1 percent) represents the total number of OHQ overflows, divided by the total number of OHQ offers plus the OHQ overflows.	

TRUNK GROUP
OVERFLOW
THRESHOLD

13.17 The existing Meridian SL-1 Percent All Trunk Busy Threshold (TFC104) is changed to Percent Trunk Group Overflow Threshold. This percentage value represents the number of overflows, divided by the number of successful calls plus overflows.

13.18 The output format for this threshold is as follows.

(System ID) (Customer Number) (Trunk Group)	TFC104
---	--------

aaaaa bbbbb

where:

aaaaa = is the all trunks busy (overflow) peg count

bbbbb = the threshold. This value (expressed in units of 0.1 percent) represents the all trunks busy peg count, divided by the total number of successful calls plus all trunks busy overflows.

TRAFFIC
MEASUREMENT
OPTIONS

13.19 New traffic measurement options are introduced with the NTRF feature. These options are set and/or queried through use of the Traffic Control program (overlay 02) in the normal manner (553-2001-450). The options are:

a 1 - to generate Routing Measurements (TFN001)

• 2 - to generate NCOS Measurements (TFN002)

- 3- to generate Incoming Trunk Group Measurements (TFN003).

SWITCHED SERVICES NETWORK

ELECTRONIC SWITCHED NETWORK

SIGNALING GUIDELINES

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1. GENERAL

1.01 The **SL-1*** Electronic Switched Network (ESN) is a private communications network intended for use by large business customers **with distributed operating locations.** (ESN is described in **309-3001-100.**)

* SL-1 is a **trademark** of Northern **Telecom** Limited

1.02 This practice describes the signaling **guide-**lines and considerations that are applicable to SL-1 ESN networks. Information presented here can be used to determine **the** compatibility of a switch that **is to** be incorporated as part of an **SL-1 ESN network.** A companion practice, **309-3001-181,** should be used in conjunction **with** this practice to determine transmission requirements.

2. BACKGROUND INFORMATION

PBX Networks

2.01 PBX networks have three basic **elements**: telephones, PBX switch, and transmission facilities. Each telephone is connected to a PEX **switch** which establishes connections between **the telephones**. Connections between telephones on different switches in a PBX network are established over a **transmission** facility (**i.e.**, trunk) between **the** pair of switches, or over a tandem connection of transmission facilities and intermediate switches,

2.02 A user establishes a call to another telephone by dialing a string of digits which direct the connection to that telephone. In **Tandem Tie Trunk Networks (TTN)**, the dialed digits steer the connection through the network. A string of successive access codes cause facilities to connect in tandem until the switch which has the destination telephone is reached. Each time a switch connects a **new** facility in tandem, it passively passes digits to the **connected** facility,

2.03 In an **SL-1 ESN network**, the **SL-1 ESN node** switches collect all digits of the called number, and pass the full called number between switches. The called number is **dialed** in a Uniform Dialing Plan (UDP) format. Users at switches connected to **SL-1 ESN nodes** may have to dial in a **TTN format** to reach the node before dialing the called number in UDP format. Similarly, the **SL-1 ESN node** may control routing to connected switches by **transmitting** a **sequence** of access **codes, followed** by the called number.

2.04 Switches are also provided with trunks to the public **network**. Calls to telephones in the public network (and to other switches) can be placed via these trunks. However, trunks between **switches** (tie trunks) are normally provided because the calling can be done at lower cost.

PBX Designations

2.05 PBX switches in a network configuration have designations that depend on the network function. **Designations** used include node, main, satellite, and **CCSA** tandem.

2.06 Node Switch. Node is the designation in an ESN network for an **SL-1** switch with the ESN Network Alternate Route Selection (**NARS**) software package. The **SL-1 ESN node** has the full capabilities for **selecting trunk** routes, while other

non-node switches have only limited **route** selection capabilities,

2.07 Main Switch. A Main Switch is a switch **which** has an outgoing trunk route to only one node. This route may be both incoming and outgoing, but routes to other nodes must be incoming only to the **Main** PBX, since the Main PBX cannot select between nodes for the routing of network calls. A Plain **PBX** may have 'CO, **PX**, and WATS trunks, as well as tie trunks to other switches.

2.08 A **PBX** which has outgoing trunk routes to more than one node, but does not meet the requirements to be classed as a node, is also classed as a main. The outgoing routes may be assigned different access codes, leaving route selection to the **user, or** may be selected by an automatic route selection capability.

2.09 Tributary Switch. A tributary switch is a PBX which has a trunk route to a **Main** PBX, but not to a node. A tributary PBX may have CO, PX, and WATS trunks, as **well** as tie trunks to other switches.

2.10 Satellite Switch. The only difference **between** a **satellite** and tributary PBX switch is that the satellite **switch** has neither incoming Central Office (C.O.) trunks nor attendant service. **For** the purpose of this **practice** satellite and tributary **PBX** switches are treated the same.

2.11 CCSA Tandem Switch. A Common Control **Switching** Arrangement (CCSA) tandem switch is the major switching vehicle in a CCSA network. The tandem switch performs trunk-to-trunk **switching** only. Its function is similar to that of an **SL-1** ESN node, the difference being that an SL-1 ESN node can support telephones in addition to trunk-to-trunk switching. **Switches** with tie trunks to the tandem switch are called Main **switches**. The CCSA network supports a dialing plan similar to the ESN Uniform Dialing Plan.

2.12 In the context of ESN, the CCSA tandem switch is the interface point between the ESN network and a CCSA network. The two networks can be arranged to **function** as a single integrated network.

Supervision

2.13 Supervision is a binary signal associated with each direction of transmission on a **trunk** facility, **Its** states are called on-hook and off-hook, analogous to conditions of a telephone set on-hook (hung up) **or** off-hook (in use). Each of the switches connected to a trunk sends a supervision signal to, and receives a supervision signal from the connected switch. Thus, the trunk has four supervision states. The trunk **is** idle when both directions are on-hook. Off-hook is sent when a call is initiated on an idle trunk. This action is called **"seizing"** the trunk. The distant **switch** receives the off-hook signal, and prepares to receive digits. Transient off-hook conditions returned from the destination switch **may** occur during call set **up**, but a steady off-hook is not transmitted until the called station answers. This off-hook signal is called **answer** supervision. The supervision changes to on-hook when the called station hangs up or is disconnected.

2.14 When a switch serves as an intermediate (**tandem**) connection **between** trunks, it normally sends the supervision signal it receives from each incoming trunk to the connected outgoing **trunk**. It also monitors for a disconnect signal so as to idle the trunks and be ready for **new** calls,

2.15 In some cases, no provision is made to return an answer supervision signal from the destination when the called station answers. Under this condition, **an off-hook** signal may be returned **from** an intermediate switch. This off-hook signal is called substitute answer **supervision**. It is provided to remove transmission impairments associated with the on-hook condition on some trunk facilities, and to distinguish a **call** which has been blocked from one which may have reached a destination party.

2.16 A typical case where a called station answer indication **is not returned**, is a connection via a **public network** trunk. Even if provision can be made to provide an indication of answer, it normally **isn't**. The **PBX switch** which connects **the call** to the **public network** trunk should be arranged to provide an off-hook signal to the incoming trunk, on completion of digit sending to the public network trunk.

Digit Transmission 2.17 In an ESN network, digits of the called number are transmitted between pairs of switches in one of two modes: Dial Pulse (DP) or Dual Tone Multifrequency (DTMF).

2.18 Dial pulse is the older technology. Each digit is represented by a **string** of pulses. The digit zero is represented by ten pulses. Each other digit is represented by the corresponding number of pulses. The pulses are transmitted as interruptions of direct current.

2.19 DTMF is a more modern technology, Digits are transmitted over the speech path via a tone code, Transmission of digits takes place at a higher rate than dial pulse (typically two to ten times faster).

2.20 Modern PBX switches are compatible with either mode. Older equipment is only compatible with dial pulse. ESN uses DTMF wherever practical, to take advantage of its higher transmission speed. However, dial pulse is sometimes required. Furthermore, there are some transmission impairments associated with DTMF. These impairments are normally only important when equipment for receiving DTMF digits is still connected after conversation begins to take place. The impairments are removed when the receiving equipment is disconnected.

Nodes of Operation 2.21 A PBX switch can operate in one of two modes when routing a call over a tie trunk: cut-through mode or senderized mode,

2.22 Cut-Through Operation. With cut-through operation, a trunk is accessed immediately following an access code, Subsequent digits are forwarded to the trunk as dialed. The station user monitors call progress tones from connected switches. The user may be required to pause in dialing to monitor for dial tone, or may be required to abandon prior to completing dialing due to blocking tone.

2.23 Pure cut-through operation provides the greatest flexibility for providing compatible operation for calls originated at main and tributary switches. The number of digits transmitted to the node can be flexible. (The node can prompt for additional digits when required for the authorization code feature.)

2.24 Senderized Operation. With senderized operation, **all digits of** the called **number** are **collected** before an outgoing trunk is accessed. The trunk is accessed, and digits are transmitted to set up the call. The transmitted digits need not be the same as those dialed (**i.e., as a result of NARS digit manipulation**). The user does **not** receive call progress tones until all digits have been transmitted. No tones are provided during dialing other than a locally generated **dial tone following** the trunk access code.

2.25 Senderized operation limits flexibility. The main PBX must be programmed to determine how **many** digits to collect **before** forwarding those digits to the node. Usually, no more than 12 digits may be collected and forwarded,

2.25 DTMF to DP Conversion A variation on cut-through operation is to provide **DTMF** to dial pulse conversion. The user dials **DTMF** digits, but dial pulses are transmitted to the trunk. The converter may block transmission in the **caller** to called party direction, or may introduce loss in the reverse direction, while waiting for digits. The converter **must** detach to remove these transmission impairments. Since the last digit can not be easily detected, timing is often used to determine end of dialing. **When** the switch has completed outpulsing, it waits a specified time for additional digits. **If** no digits are received in that interval, conversion is disabled. The timing may not properly distinguish a pause in dialing from a last digit, prematurely cancelling the **forwarding** of digits. Extending the timing **causes** the transmission impairments to exist during conversation if the called **party** answers quickly.

2.27 The SL-1 ESN nodes combine cut-through and senderized modes of operation. The switch collects the access code and enough digits to select a trunk. During this interval, the operation is very close to a register sender mode. The trunk is accessed and a string of digits outpulsed (not necessarily the same as those dialed). Subsequent digits are forwarded to the trunk as dialed, in a receive and resend mode, which is closer to cut-through operation.

Outpulsing Control 2.28 After a tie trunk is **seized**, digits of the called number will normally be transmitted. (The only exception is a manual trunk, which rings a designated station when seized.) **However**, most terminating equipment requires a variable time interval to prepare for reception of **digits**. This time interval often depends on the **switch's** call-processing load. Therefore, a fixed delay before **sending** digits would be unreliable. There are four commonly used ways of handling start dial control:

- **Immediate start**, Applies in those cases where a short fixed delay is required for the **switch** to prepare to receive digits.

- **Delay for dial tone**. A dial tone is **provided when** the switch is ready to receive digits.

- **Wink start**. A momentary off-hook signal is sent **when** the terminating equipment is ready to receive digits.

- **Delay dial**. An off-hook signal is sent to signify that the switch is **not** ready to receive digits. An **on-hook** signal is sent when the switch is **ready** to receive digits.

2.29 The delay for dial tone is used when a user controls digit sending, as in a **TTN**. **Wink start** and delay dial are used in registerized digit **sending**. Normally, only one start dial control is used. However, dial tone may be combined **with** any of the other three on some switches.

2.30 The SL-1 switches in an ESN network are able to work **with** any of the start dial signals,

3. CALL SETUP SEQUENCES IN ESN

3.01 Following is a description of the various call setup sequences that can occur between conventional PBX **equipment and** an SL-1 ESN node, with emphasis **on** signaling compatibility. Calls are described (1) to the node, (2) between **nodes and** (3) from the node. Several cases are included, reflecting variations such as the type of machine involved (cut-through or **senderized**) and whether or **not** tandeming is involved.

3.02 The SE-1 PBX without ESN main or ESN node software is treated as a cut-through PBX in the **following** discussion. The **SL-1** PBX with ESN main software is not addressed.

Calls to the Node

3.33 Case 1. This case pertains to a call to the **SL-1** ESN node from a station at a cut-through main switch.

- (1) The user at the main PBX goes off-hook, **receives** dial tone from the main **PBX**, then dials the network access code.
- (2) The **main** PBX seizes a tie trunk to **the** SE-1 ESN node, and provides audio **transmission** to the caller. This permits subsequent call-progress tones from the node to be **heard** by the caller.
- (3) **The** node returns dial tone to the caller **when** it is ready to receive digits.
- (4) The user **dials** the desired **number and**, if required, the authorization code. The digits **are** transmitted to the node as dialed. (The node provides an Authorization Code Request Tone **only** if the authorization **code** is required. If not, the routing takes **place** immediately following the last digit of the called number.)
- (5) The node proceeds to set up the call.

3.04 Case 2. This describes a call from a cut-through non-senderized tributary PBX to the **SL-1** ESN node via a cut-through non-senderized main PBX. (Wherever practical, a direct trunk group to the SL-1 ESN node should be provided to avoid using the main **PBX** as a tandem switch.)

- (1) The user at the tributary PBX goes off-hook, receives dial tone from the tributary PBX, and diafs a tie trunk access code to access the main PBX.
- (2) The tributary PBX seizes a tie trunk to the main PBX, and provides audio transmission so that tones from the main PBX can be heard by the user, and dialed digits from the user can pass to the main PBX. The main PBX is normally arranged to provide dial tone to incoming tie trunks.
- (3) Upon receiving dial tone from the main PBX, the user then dials the ESN network access code and desired number. The call setup proceeds as in Case 1 except that the user's dialed digits and the tones from the node pass through both cut-through switches.

3.05 Case 3. This describes calls from stations at a senderized main PBX to an SL-1 ESN node,

- (1) The user goes off-hook, receives dial tone, dials the ESN access code, receives second dial tone from the main PBX, and then dials the desired number.
- (2) The senderized main PBX does not seize a tie trunk to the node after receiving the access code. Instead it collects the digits of the called number, and then seizes a tie trunk to the node. The node does not provide dial tone.
- (3) The senderized main PBX and the node are mutually arranged to utilize either a wink-start or delay-dial signal as a start-dial signal to initiate outpulsing.
- (4) When the main PBX receives the appropriate start-dial signal, it outpulses the called number to the node, and then connects the user so that subsequent ringing signals, etc., can be heard.

3.06 The SL-1 authorization code feature is not supported in this case. The limitations which prohibit the authorization code feature are:

- The senderized main PBX has no provision to provide the authcode request tone for authorization code digits

- T&e senderized main PBX does not have the capability to register enough digits for the authorization code and called number

Forwarding of additional digits after the senderized main PBX has outpulsed the called number to the node is not practical,

3.07 Case 4. This applies to calls from a cut-through tributary PBX to the SL-1 ESN node via a senderized main **PBX**. This case is similar to calls from a senderized main **PBX**. The user at a cut-through tributary PBX dials a tie trunk access code to reach the main PBX and receives dial tone from the main PBX. **From** this point on, the call is handled as though it originated at the senderized main **PBX**.

3.08 Case 5. This applies to calls from a **senderized** tributary PBX to the **SL-1** ESN node via a cut-through main **PBX**. This is not permitted because of signaling compatibility problems. The tributary cannot provide the proper outpulsing control for routing the call through the main to the node. Direct trunks must be provided from the tributary PBX to the node in this situation. (The tributary then becomes a main PBX, by definition.)

3.09 Case 6. In this case, both the main PBX and tributary PBX are senderized. This is also not permitted due to compatibility problems similar to Case 5. Direct trunks **must** be provided to the node **from** the tributary PBX. The tributary PBX thus becomes a **main PBX**.

Call Coapletions

3.10 Call completions to stations on the **node**, stations at other nodes, public network trunks from the node and **from** connected nodes are handled in the normal manner.

3.11 The node completes calls (via tie trunks to the main) to the following destinations:

- stations at the main PBX
- stations at a tributary PEX connected to the main PBX
- off-network stations via public network trunks terminating on the main
- off-network stations via public **network** trunks terminating on the tributary

3.12 In all cases, call routing to the main is **initiated** by an off-hook signal sent to the tie trunk. The basic sequences for call completion are:

- (a) To reach a station at the main PBX, **outputse** the station Directory Number **(DN)**.
- (b) To reach a station at the tributary PBX, **out-**pulse an access code for a tie trunk to the tributary, pause if necessary, and **outputse** the station **DN**.
- (c) To reach a public network station via a CO trunk terminating on the main, **outputse** the access code for the CO trunk, **followed** by the public network number.
- (d) To reach a public network station via the tributary **PBX**, the node **outputses** an access code for the main to tributary **tie** trunk, followed by the access code to the CG trunk, followed by the **public** network number.

3.13 Case 1. This case applies to a call from an **SL-1 ESN node** to a station at either a cut-through or a senderized main **PBX**.

- (1) The node seizes a tie trunk to the main and delays for the main to become ready to receive digits.
- (2) The node **then** **outputses** the station DN digits to the main **PBX**.

3.14 Case 2. This case involves a **call** from an **SL-1 ESN node** to a tributary PBX via a cut-through main **PBX**. This is similar to Case 1 except the initial digit(s) **outputsed** **by** the node **is** an access code for a tie trunk connecting the main PBX to the **tributary** PBX. The procedure then is as follows:

- (1) The node inserts a fixed pause (for a delay for dial tone) after the access code, unless both main and tributary are step-by-step (SXS) switches.
- (2) The node resumes **outputsing** **when** the fixed pause interval elapses or dial tone is detected, as appropriate.

- (3) The resumed **outpulsing** will be the DN of the station **at** the tributary **PBX**.

3.15 Case 3. This case involves a **call** from an **SL-1 ESN node** to an **off-net** station via a cut-through main **PBX**.

- (1) The access code **outpulsed** by the **SL-1 ESN node** is for a **CO trunk**, instead of the **trunk** as in **Case 2**.
- (2) A fixed pause or a delay for dial tone after the access code is required, even when both the main **PBX** and the **CO** are **SXS** machines,
- (3) The resumed outpulsing consists of the public network number rather than a **PBX** station number,

3.16 Case 4. This case applies **when** the main **PBX** is **senderized** and tandems a call from the **SL-1 ESN node** to a tributary **PBX** or to a **CO**.

3.17 The **beginning** of the call Setup **sequence** when the main **PBX** is **senderized** is the same as the previous cases. However, **once** the **node** has begun **outpulsing**, it outpulses all of the digits without pausing. Outpulsing should be **DTMF** digits, if the main **PBX** can **receive DTMF digits**, regardless of the capability of the tributary **PBX** or **CO**. The **senderized main** **PBX** collects the digits, translates and prefixes if applicable, **and completes** the call to the next **switch**.

Calls **To** Other
Private Networks

3.18 Tandem Tie Trunk Network. Calls to a **TTTN** are engineered similarly to calls to a **main** **PBX** with tributaries. The maximum **number** of switches **connected** in tandem is five **in** a **TTTN**. Thus, up to three access codes may have to be outpulsed, with pauses. To avoid the requirement for a large **number** of digits to be outpulsed, trunks to several switches **in** the **TTTN** should be arranged so that no more than two trunks **in tandem** are required to reach any **station** set in the **TTTN**.

3.19 Common Control Switching Arrangement. A **n** **SL-1 ESN node** **with tie** trunks to a **CCSA** switch is arranged to **output** 7 digits to the **CCSA** switch to complete network calls to stations in that part of the network. Optionally, the **SL-9 ESN** nodes may **output** 10 digits to complete **off-network** calls via the **CCSA** switch as well.

Calls From **Other
Private** Networks

3.20 An **SL-1** ESN node is able to function as a TTTN switch for calls from a TTTN. The user *in* a TTTN sequentially dials access codes to add trunks in tandem.

3.21 Calls from a TTTN do not require different engineering **than** calls from a main PBX, Other than an **SL-1** route data block option to arrange the incoming trunk for TTTN operation. **TTTN** operation is currently supported by SL-1 and is not changed for **ESN**.

3.22 Calls from a CCSA switch can be arranged to terminate on an SL-1 ESN node or any other **switch** which is part of the **SL-1** Coordinated Dialing Plan (**CDP**). Current SL-1 CCSA trunk options can accommodate this, Tie trunks from a CCSA **switch** must be provided to all switches in the **ESN network** (including those that are part of the GDP) to be accessible from the CCSA switch. An **unambiguous** numbering **plan** encompassing both the **ESN network** and **CCSA** must be arranged, The **SL-1 ESN** node routes calls in the **CCSA numbering plan** to the CCSA switch.

Requirements for
Dialing Pauses

3.23 When **outpulsing** to main PBX and to a TTTN, the **SL-1** ESN nodes are occasionally required to **pause** at various points in digit strings to allow for trunk access and register **attachment**. **Failure to pause causes digits to** be missed, calls to be **connected to wrong** numbers or lost altogether.

3.24 The SL-1 ESN software can **provide** for pauses following trunk access codes in the **NARS** translation tables. The general rule is that each trunk access code outpulsed **must be followed** by a pause. However, there are a number of situations **where** the pause is not required.

3.25 In determining whether the pause is required, one must consider:

- **what** type of PBX has **been** reached in the dialing
- **what** piece of **equipment** is being accessed.

3.26 The situations where pauses are not required are:

- the access code is for connecting a step-by-step PBX to a step-by-step PBX

- the access code is for connecting an SL-1 PBX to any other PBX, providing that subsequent pauses are not required
- the access code is 9 for a CO trunk via a **centrex** PBX (but not to other CO trunks)
- the access code is for an automatic route selection feature 05 any **PEX**.

3.27 A potential problem area is where a trunk access code requiring a pause is made after the call has been routed through one or more SL-1 switches. While the SL-1 ESN node need not pause between access codes for routing through the SL-1 switches, if it does not, a problem occurs at the point where the pause is required. The SL-1 switches in the connection will insert the proper delay after the access code before resending digits, However, the time spacing **between** digits is not maintained. The trailing digits "catch **up**" with the leading digits. The time delay required after an access code is eliminated. To avoid this problem, the **SL-1** ESN node should insert pauses after each access code for this call routing.

SWITCHED SERVICES NETWORK

ELECTRONIC SWITCHED NETWORK

TRANSMISSION GUIDELINES

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1. GENERAL

1.01 The **SL-1*** Electronic switched Network (ESN) is a private communications network intended for use by large business customers with distributed operating locations. (ESN is described in 309-3001-100.)

1.02 This practice describes the major transmission considerations that are taken into account when planning an ESN network. The ESN transmission objective is to provide transmission quality comparable to that for public network calling. While this objective can easily be met for on-network calls, it is unreasonable to meet this objective for tandem combinations of public and private network trunks, since this produces a connection with inherently poorer performance. However, the sacrifice in performance should be small enough to be tolerable or unnoticeable. Noticeably poorer performance leads to an excessive number of calls placed by manually initiated bypass routing through the Direct Distance Dial (DDD) network.

*SL-1 is a trademark of Northern Telecom Limited

2. TRANSMISSION CONSIDERATIONS

Echo

2.01 All voice connections between station sets require two **directions** of transmission for conversation to take place. When the **signal** transmitted in one direction is reflected over the other directional path, the caller hears his/her own voice **with** a slight delay. The effect is perceived differently, depending on the delay, as sidetone, rain barrel effect, or echo. **Two-wire** facilities require care in matching **impedances** to prevent reflections. Four-wire facilities do not generate reflections themselves, but do not eliminate the reflection problem in built-up connections, since there **will** be, in most cases, **2-wire** facilities to the station sets.

2.02 The objection to echo increases **with** the echo delay. The Via Net Loss (VNL) plan provides an increasing loss depending on delay. However, the loss also reduces the received volume. Limits are placed on the amount of loss used to suppress echo, **when** these limits are exceeded, echo suppressor devices are provided instead of loss.

Loss

2.03 The provision of good transmission requires the following compromises:

- The need for sufficiently **low** (one-way) loss in each direction to provide satisfactorily high received **volumes**.
- **Minimum** contrast in received volumes on different calls,
- The need for sufficiently high **round-trip** losses to ensure adequate performance from the standpoint of suppressing talker echo, *noise*, and near-singing,

2.04 The following loss plan has been developed for ESN. The network is partitioned into node-to-node connections, node-to-main connections and main-to-satellite or tributary connections. The plan requires that:

- node-to-node trunks have a maximum loss of **3.5 dB**
- node-to-node tandem connections have a **maximum** loss of **4.1 dB**

- node-to-main trunks have a maximum loss of 2.5 dB.

2.05 These loss objectives are met by installing echo suppressors and reducing the loss to 0 dB on trunks **when** the objective loss is exceeded with VNL alone,

Tandem Switching

2.06 Switching Network Tandem A public network tandem switch is usually collocated with the carrier facilities that serve the switch. Tandem connections can be made between trunks terminating at this switch without significant degradation of transmission performance.

2.07 PBX Network Tandem Switching. A PBX network tandem switch is usually located on customer premises remote from the carrier facilities that serve the PBX switch. These carrier facilities are generally located at a telco switching center and connected to the PBX switch by local loop plant (cable). If the telco switching center does not have long-haul carrier facilities, short-haul carrier facilities are used to connect to another telco switching center which has these facilities. Thus, a tandem connection made at the PBX switch, can introduce the distortion of two loop plant connections and two carrier facility connections, both avoided in the public network tandem connections.

Trunk Routing Rules

2.08 Trunk routing rules define the allowed connections between node, main, tributary, and satellite PBX switches. These routing rules are summarized in Table 2-A.

Table 2-A
TRUNK ROUTING RULES

From	To	NODE	MAIN	TRIBUTARY	SATELLITE
NODE		Yes (Up to 4 links)	Yes (Note 4)	No (Note 3)	No (Note 3)
MAIN		Yes (One Node Only)	(No:: 1)	Yes (One Main Only)	Yes (One Main Only)
TRIBUTARY		No (Note 3)	Yes (One Main Only)	No (Note 2)	No (Note 2)
SATELLITE		No (Note 3)	Yes (One Main Only)	No (Note 2)	No (Note 2)

Notes:

1. Permitted for non-tandem trunks if the connected main PBX switches are part of a coordinated dialing plan. Also, routes of this type already in place when ESN is installed may be allowed to remain. **However**, these routes should be **eliminated** as part of **network** evolution, to support the dialing plan.
2. Routes of this type already in place when **ESN** is installed may be allowed to remain. These routes should be **eliminated as** the network evolves.
3. This route is permitted by upgrading the tributary or satellite switch to main switch capabilities,
3. One-way routes from nodes to main PBX switches are not restricted,

3. TRANSMISSION PLANNING

3.61 The planning of electronic switched networks can be partitioned into four major tasks:

- (a) Assess the current equipment to identify locations of nodes, and designate existing equipment to **remain** in place as main, satellite and tributary **PBX**.
- (b) Plan the tie trunk routes,
- (c) Plan the network routing.
- (d) Plan off-network call routing.

3.02 Network planning is both interactive and continuing. As each task is completed, previous tasks should be repeated to optimize the network design.

Placement of Nodes

3.03 The first decision is the choice of locations for node switches. (In most cases the network planner is given little choice in this matter. One or more obsolete switches may already be earmarked for replacement. Other **switches** which might be prime candidates for node capabilities are prohibited from replacement.)

3.04 **Wherever** the network planner is given a choice, the **following** transmission factors should be considered:

- (a) Best transmission is obtained if the node can be located near a **telco toll** switching center, to minimize the loop plant and short-haul facilities **between** the node and long-haul facilities,
- (b) The node should be near the middle of its cluster of **main PBX** switches, **so** most main-to-node facilities fall **in** the short-haul category,
- (c) If earth satellite facilities are used extensively, nodes should be located near the earth station facilities.

3.05 Once the nodes are designated, the **remaining** PBX switches are designated as main, **satellite** or tributary switches. All switches which have direct trunk groups to a node are main PBX switches. Those switches which access a node via a **main PBX**

are satellite or tributary **switches**, depending on **whether** or not they have incoming **C.C.** trunks.

Tie Trunk Routes

3.06 Each pair of nodes in the network represents a potential tie trunk route. Traffic considerations and tariffs determine how **many** routes are **equipped** and how many trunks are required for each route. In general, the objectives of least cost and best transmission both dictate that the number of tandem trunks required to establish any connection be **kept** to a minimum. Thus, direct trunk routes should be established wherever practical.

3.07 Direct trunk routes represent a radical departure from Tandem Tie Trunk Networks (TTN). For TTN, routing is usually organized into major trunk route **"highways"**, with **"feeder"** routes. This structure is efficient when alternate routing is not permitted, as for TTN, but is not efficient when alternate routing, as for **ESN**, is supported. Thus, converting a large TTN to an ESN can have a large impact on permitted routings.

Network Call Routing

3.08 The direct and alternate routes to be used for routing calls **must** be planned so that transmission can be evaluated on each route and **appropriate** controls established on individual trunks to meet objectives. Table 3-A gives the maximum number of node-to-node routes which may occur in networks of various sizes. The table establishes the upper limits **on** the number of routes, to ensure that no potential route is overlooked.

Table 3-A
 ROUTE COUNT TABLE
 (MAXIMUM NUMBER OF NODE-TO-NODE ROUTES)

NODES;	DIRECT ROUTES	2 TRUNKS IN TANDEM	3 TRUNKS IN TANDEM	4 TRUNKS IN TANDEM
1	0	0	0	0
2	1	0	0	0
3	3	3	0	0
4	6	12	12	0
5	10	30	60	60
6	15	60	180	360
7	21	105	420	1260
8	28	168	840	3360
9	36	252	1512	7560
10	45	360	2520	15120
11	55	495	3963	27720
12	66	660	5940	47520

Assumptions:

- 1, Each node has trunks to every other node,
- 2, **All** routes that do not include the same node more than once are valid, [In practice, the number of valid routes is considerably less, since many of the routes **will** not make sense in a real work **environment.**]

Transmission Controls

3.09 Transmission controls are first established for each direct trunk route; then two-trunk tandem routes; then **three-trunk** tandem routes, and so on. For each route, an Echo Suppressor (ES) control or VNL value is specified, (VNL applies to land circuits less than 1800 miles [**2880 km**]; longer land circuits and all satellite circuits require echo suppressor control,)

3.10 The required VNL value is determined from the round-trip delay, which depends primarily on the **type** of facility and distance. Table 3-B gives approximate VNL values for varying distances (air-line miles) based on the type of facilities typically provided **by a telco.** (The actual distance could be considerably greater since the actual signal path is less direct. **However,** margin has been added to accommodate average deviation from direct routing,)

Table 3-B
LOSS TABLE FOR LAND CIRCUITS

DISTANCE Miles (km)	VNL (dB)
0 - 100 (0 - 160)	
100 - 400 (160 - 640)	1.0
400 - 700 (640 - 1123)	1.5
700 - 1000 (1120 - 1600)	2.0
1000 - 1300 (1600 - 2080)	2.5
1300 - 1600 (2080 - 2560)	3.0
1600 - 1800 (2560 - 2880)	3.5

Note: Land circuits longer than 1800 miles (2880 km) and all satellite circuits require echo suppressor control.

3.11 Tuo-Trunk Tandem Routes, Wherever **two VNL** trunks are in tandem, their VNL losses are summed. If the loss exceeds 4.1 dB, at least one of the trunks should be equipped with echo suppressors if the route is to be permitted. Normally, the higher loss trunk is selected for echo suppression but another consideration is the minimization of the number of trunks which **must be** changed to echo suppressor control. Thus, trunks which appear **most** frequently in high-loss connections should also be considered prime candidates. Table 3-C **summarizes** this requirement,

Table 3-C
TRANSMISSION CONTROL REQUIREMENTS FOR
TYO-TRUNK TANDEM CONNECTIONS

TRUNK 1	TRUNK 2	REQUIREMENT
E.S.	E.S.	No Action
E.S.	VNL	No Action
VNL	E.S.	No Action
VNL	VNL	If total loss is less than 4.1 dB, no action, otherwise, change 1 trunk to E.S. or prohibit t&is connection.

3.12 Three-Trunk Tandem Routes, For smaller **networks**, all three-trunk tandem routes should be sorted and tested for proper controls. For larger networks, the number of three-trunk tandem routes **actually permitted** is small in comparison to the number of possible routes. Therefore, the more efficient approach is to sort out the permitted routes and to consider only them.

3.13 For three-trunk tandem routes, echo suppressor combinations must be considered as **well** as total loss. Whenever a **VNL** tie trunk is connected between two trunks **with** echo suppressors, that trunk must be equipped with echo suppressors, to ensure that intermediate echo suppressors are disabled.

3.14 Table 3-D **summarizes** the transmission control requirements for three-trunk tandem connections.

Table 3-D
**TRANSMISSION CONTROL REQUIREMENTS
 FOR THREE-TRUNK TANDEM CONNECTIONS**

TRUNK 1	TRUNK 2	TRUNK 3	IF LESS THAN 4.1 dB	TOTAL LOSS IF GREATER THAN 4.1 dB
E.S.	E.S.	E.S.	No action required	No action required
VNL	E.S.	E.S.	No action required	No action required
E.S.	VNL	E.S.	Prohibit this connection (preferred) or change trunk 2 to B.S.	
E.S.	E.S.	VNL	No action required	No action required
VNL	E.S.	VNL	No action required	Change trunk 1 or 3 to E.S. or prohibit this connection
E.S.	VNL	VNL	No action required	Trunk 2 must be B.S. or prohibit this connection
VNL	VNL	E.S.	No action required	Trunk 2 must be E.S. or prohibit this connection
VNL	VNL	VNL	No action required	One or more must be E.S. or prohibit this connection

Note: **When** a trunk is changed to echo suppressor (E.S.) connection, all combinations in which it appears must be checked for routing violations.

Off-Network Call Routing 3.15 The **transmission** properties of a call depend to a large extent on the type of facilities (Table 3-F) over **which** the call is transmitted. Because of these properties, certain destinations are dis-allowed for some types of off-network call routing. These are summarized in Table 3-E.

Table 3-E
PERMITTED OFF-NETWORK CALL ROUTINGS

ROUTE OFF-NETWORK CALL			DESTINATIONS PERMITTED				
FROM	VIA	VIA	STATION	C.O. TRUNK	FEX TRUNK	WATS TRUNK	OTHER COMMON CARRIER
Node			Yes	Yes	Yes	Yes	Yes
Tie Trunk	Node		Yes	Yes	Yes	Yes	Yes
Node	Main (Note)		Yes	Yes	Yes	No	No
Node	Main	Tributary OR Satellite	Yes	No	No	No	No
Main	Tributary OR Satellite		Yes	Yes	Yes	No	No

Note: Off-network calls to **C.O.** or **PEX** trunks are allowed only within local calling **areas.**

Network Facilities 3.16 Private Network Facilities. Private **network** facilities can be ordered from the local telephone company and other common carriers. (Although facilities are ordered on a point-to-point basis, the **telco** must be informed of the **overall** planned **network** and will generally assist in the design and selection of facilities.) The facilities which may be ordered from the telephone **company** are:

(a) 2-Wire Trunks.

- TL 11M/E (E&M Type I Signaling)
- TL 12M/E (E&M Type II Signaling).

(b) 4-Wire Trunks.

- TL 31M/E (Type I Signaling)
- TL 32M/E (Type PI Signaling).

3.17 Several qualities of conditioning beyond the "basic" line quality are available, including: C1, C2, C3, C4, and D1. Conditioning is the tolerance on frequency response and delay distortion. Generally, the "basic" line quality is adequate for voice applications. The conditioning is usually required for voiceband data applications.

3.18 Table 3-F summarizes the requirements for ordering facilities,

Table 3-F
PRIVATE NETWORK FACILITY REQUIREMENTS

FROM	TO	FACILITY REQUIRED	SL-1 CARDS
Node	Node	TL 31M/E Interface (4-Wire E&M Type I)	QPC237
Node	Main	TL 31M/E Interface (4-Wire E&M Type I)	QPC237 QPC71
Main	Satellite	TL 11M/E or TL 12M/E (2-Wire E&M Type I)	QPC71

3.19 **When** telco facilities are ordered, the FCC registration number **must be** provided. Different interfaces **must** be ordered for non-registered **equipment. It** is acceptable to have registered and non-registered equipment attached to the same facility, as **well** as to have a tie trunk between telco and customer-provided PBX equipment, **When** ordered from a telco, the facilities are furnished with **VNL** loss, or 0 **dB** loss if echo suppression is provided. **When** customer-owned facilities are used, it is the customer's responsibility to insert the **VNL** loss into the facility.

3.20 Public Network Facilities. Public **network** facilities **must be** ordered from the telephone **com-**
pany. (A summary of these facilities is **given** in Table 3-G.) The facilities **must be** identified as connecting to customer-provided equipment and the FCC registration number **must be** given. The facilities ordered can include:

- PBX central office trunks
- PBX foreign exchange trunks, to specific foreign exchanges
- **PBX inward** WATS trunks
- PBX outward PATS trunks
- Off-premise stations,

3.21 In addition, direct trunks to **SPRINT** and other common carrier systems **may** be ordered from these service **suppliers.** These services provide indirect access to the public network.

Table 3-G
 PUBLIC NETWORK FACILITY REQUIREMENTS

SERVICE REQUIRED	FACILITY REQUIRED
Local calling area	PBX -- C.O. trunk
Calling area local to a distant exchange	PBX PEX trunk
Wide calling area within same state	Intra-state WATS outgoing PBX trunk
Calling area, all bordering states	Interstate WATS outgoing PBX trunk, Band 1
Calling area within USA, Canada, bordering states and beyond	Interstate WATS outgoing PBX trunk, Band 2 and higher
Calling to major cities in United States	Non-Bell services such as SPRINT or NCI

4. TRANSMISSION PERFORMANCE

Voice Quality Performance

4.01 The quality of a voice connection made over tandemed trunks is a function of the composite characteristics of the trunks. Each trunk added to the connection degrades the overall transmission performance. Thus, some limits must be placed on the number of trunks permitted in tandem, as well as which trunks may be connected.

4.02 To maintain adequate voice quality while at the same time keeping the routing restrictions from becoming unduly complex, ESN is partitioned into two basic connection categories, each with its own set of requirements. The connection categories are:

- node-to-node
- node-to-main, satellite or tributary.

4.03 Node-to-Node Connections. The restrictions on node-to-node connections are:

- (a) No trunk has a loss exceeding 3.5 dB.
- (b) No combination of trunks used for a valid connection has a loss exceeding 4.1 dB.
- (c) Split echo suppressors are provided at each end of each trunk equipped with echo suppressors. Each echo suppressor is enabled or disabled by the SL-1 switch at its end.
- (d) Tandem connections of echo-suppressor controlled trunks are permitted, provided the intermediate echo suppressors are disabled. The SL-1 switch disables the echo suppressors it controls when a direct connection is made between two echo-suppressor controlled trunks, thus meeting this requirement. Routes with one (or more) intermediate non-echo-suppressor controlled trunks between echo-suppressor controlled trunks are not allowed because the intermediate echo suppressor will not be disabled.
- (e) Generally, no more than three tie trunks should be connected in tandem. (An absolute limit of four is imposed between echo-suppressor controlled trunks.) SL-1 software is arranged to disable echo suppres-

sors when it tandems a call from an echo-suppressor controlled trunk to another such trunk. It does not disable echo suppressors on other connections.

4.04 Node-to-Main, Satellite or Tributary Connections. Restrictions on these tie trunks are:

- (a) The node-to-main trunk is normally a land circuit not exceeding 250 miles (400 km). If this objective cannot be met, the main PBX is treated as a node for transmission planning, and the transmission planning of the node-to-main tie trunk is considered part of node-to-node transmission planning.
- (b) The node-to-main tie trunk, if less than 250 miles, must have a loss not exceeding 2.5 dB.
- (c) Main-to-satellite and main-to-tributary trunks must have a loss not exceeding 2 dB.

Note: In some cases, the telco may only be able to provide non-VNL trunks which have a loss exceeding these objectives. If the loss is significantly higher than VNL, the SL-1 switchable pad must be in the 'pad-out' mode for connection to these trunks.

4.05 ESN can route calls over private network facilities to public network trunks, in order to minimize toll charges. The public network has a designed loss which does not take into account the added loss of extending the call over private network facilities.

4.06 ESN, as any other private network, provides inherently poorer performance on these connections than if the call were routed directly to the public network. The amount of degradation must be kept small enough that the connection is acceptable to most users. This loss is restricted as follows:

- (a) Off-network long distance connections are only to be established from trunks terminating on nodes. Thus, the private network loss added to the public network loss is limited to:
 - 4.1 dB for calls originating at node stations

6.6 dB for calls originating at main stations.

- (b) Off-network local connections may exit at node and main PBX. For nodes, the loss is as in (a). For mains, the loss is limited to:

6.6 dB for calls originating at node stations

9.1 dB for calls originating at main stations.

Voiceband Data Performance

4.07 Voiceband data modems are used to transmit data between private network switches. General guidelines on the expected performance of various modem types for different ESN connections are provided here.

4.08 The guidelines are based on documented transmission performance of Bell System private lines and actual measurements of SL-1 and several metropolitan area private lines. The expected performance can be stated in statistical terms only to reflect the wide performance range of actual circuits.

4.09 The probability of success in completing a data call is based on an overall average of all Bell System private lines. A particular line or group of lines may be much worse or much better than the average. Therefore, the expected performance stated here should be viewed as a general indication only.

4.10 Tables 4-A and 4-B specify the expected performance for ESN connections with 1, 2 and 3 trunks in tandem. The percentage of successful calls is given for a given modem bit-rate. Table 4-A applies to modems with Automatic Adaptive Equalizers, and no significant improvement is expected with C2 or C1 conditioned lines relative to the basic quality private lines. Using D1 conditioned lines (guaranteed lower noise) can improve performance as shown in the table. For modems without automatic (i.e., fixed) equalizers, significant improvement can result from using C1 or C2 conditioning over basic line performance (see Table 4-B).

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Table 4-A
 PERFORMANCE OF MODEMS WITH
 AUTOMATIC ADAPTIVE EQUALIZERS

NUMBER OF TRUNKS CONNECTED IN TANDEM	MODEM BIT RATE (Probability of Success)	
	BASIC C1, C2 CONDITIONING	D1 CCNDITIONING (4-Wire Facility)
1	2403 b/s (80%)	4803 b/s (<80%)
2	2400 b/s (50%)	4800 b/s (50%)
3	Not Recommended	2400 b/s (50%)

Table 4-B
 PERFORMANCE OF MODEMS WITH FIXED EQUALIZERS

NUMBER OF TRUNKS CONNECTED IN TANDEM	MODEM BIT RATE (Probability of Success)		
	BASIC CCNDITIONING	c2 CONDITIONING	c3 CONDITIONING
1	2400 b/s (75%)	2400 b/s (80%)	2400 b/s (90%)
2	2400 b/s (50%)	2400 b/s (60%)	2400 b/s (75%);
3	Not recommended	2400 b/s (50%)	2400 b/s (65%)

5. MAINTAINING TRANSMISSION PERFORMANCE

5.01 **Maintaining** high transmission performance in private networks requires constant vigilance on the part of the **network** administrators. The experience in the Northern Telecom network demonstrates that transmission performance degrades over time. The degradation can be traced to inadequate maintenance of the facility by the supplier and differences in performance when the supplier makes administrative changes in the assignment of equipment to provide tie trunk service,

5.02 **Installation.** After a tie trunk facility is installed, a transmission test is performed to verify that the facility meets tariffed requirements. Only after the performance has been verified should the facility be turned up for service.

5.03 **Scheduled Maintenance.** Scheduled maintenance is performed at regular intervals whether or *not* trunk faults are known to exist. Testing should be carried out once a week until confidence is gained that less frequent testing is required,

5.04 **Corrective Maintenance.** Follow-up to trouble reports generated **by users** or software diagnostics identifies various trunk problems in need of correction.

Transmission Test-
ing of Tie Trunks

5.05 ESN switches have been provided with the **capability** of accessing a quiet **termination** or **1020-Hz** test tone at a remote ESN switch. These capabilities permit testing of tie trunk transmission performance. (Bead 553-2751-104 for further information on **SL-1** trunk transmission testing capabilities,)

5.06 **Testing Loss.** The following test is **performed at each** end of each tie trunk, so that both directions *of* transmission are checked:

- (1) From the maintenance terminal, load the trunk test overlay program and access the remote test tone for the trunk to be tested,
- (2) Connect a transmission level meter to the **"facility in"** access jacks of the trunk under test. **Measure** the level of the **1020-Hz** test tone. The permitted level requirements, based on a switched-in **pad** mode at the far-end **SL-1 switch**, are given in **Table 5-A**.

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Table 5-A
TRANSMISSION LEVEL REQUIREMENTS

TRUNK LOSS	DESIGN (dB)	MINIMUM (dBm)	MAXIMUM (dBm)
	0	-17.5	-11.5
	0.5	-18	-12
	1.0	-18.5	-12.5
	1	-19	-13
	2.0	-19.5	-13.5
	2.5	-20	-14
	3.0	-20.5	-14.5
	3.5	-21	-15

5.07 If the requirement in Table 5-A is not *met*, the fault must be isolated to the trunk facility or SL-1 equipment. The **following** test aids in this isolation. Perform the test at each end of the facility.

- (1) From the maintenance terminal, load the trunk test overlay program and access the local test tone,
- (2) Connect a transmission level meter to the "equipment **out**" jack of the suspect trunk. The level requirement is:
 - For QPC71 circuit packs: **-15 dBm** minimum, **-13 dBm** maximum
 - For QPC237 circuit packs: **-15.5 dBm** minimum, **-13.5 dBm** maximum.

5.08 If the requirements in (2) are not met at either or both facility ends, corrective **maintenance** must be performed, and the test repeated. If the requirement is met at both facility ends, the facility supplier should be **requested** to perform corrective maintenance.

5.03 **Testing Noise.** The **following** test is to be performed at each end of each tie trunk, so that **both** directions of transmission are checked,

- (1) From the **maintenance** terminal, load the trunk test overlay program and access the remote quiet termination for the trunk to be tested.
- (2) Connect a noise meter to the "facility **in**" access jacks of the trunk under test. The noise requirements are given **in** Table 5-B.

Table 5-B
TRANSMISSION NOISE REQUIREMENTS

DISTANCE MILES (Km)	MAINTENANCE (dBrnC)	IMMEDIATE ACTION (dBrnC)
0-15 (0-24)	28	36
51-100 (82-160)	28	36
101-200 (162-320)	29	36
201-400 (322-640)	31	36
401-1000 (642-1600)	33	40
1001-1500 (1602-2400)	35	40
1501-2500 (2402-4000)	36	40
2501-4000 (4002-6400)	39	44
	41	46

Note: Trunks with a noise measurement within the **MAINTENANCE** category **may** be **left** in service. Trunks with a noise measurement within the **IMMEDIATE ACTION** category should be immediately removed from service. In either case, **maintenance** action should be promptly initiated.

5.10 If the requirements of Table 5-B are not met, the fault must be isolated to the trunk facility or SL-1 equipment. The following test will accomplish this isolation. Perform the test at each end of the facility,

- (1) From the maintenance **terminal**, **load** the trunk test overlay program and access the local quiet termination.
- (2) Connect a **noise** meter to the 'facility **out**' jack of the suspect trunk. The requirement is that the noise not exceed 23 **dBrnC**.

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5.11 If the requirement is not met at either or both facility ends, corrective maintenance should be performed on the appropriate SL-1 and the test repeated. If the requirement is met at both facility ends, the facility supplier should be requested to perform corrective maintenance,

6. TRANSMISSION CONSIDERATIONS FOR REMOTE NETWORK ACCESS

6.01 One application of private networks is to reduce toll charges on public network to public network calls. The user in the public network makes a local or toll-free (**INWATS**) call to one of the private network switches, then calls to an off-network destination via either the Direct **Inward** System Access (**DISA**) feature or attendant assistance.

Gain

6.02 The **public** network transmission plan does not support tandeming of two (**or more**) connections. Such tandeming inherently results from using the private network to make a public network to public network call. The **loss** can be partially offset by **gain**.

6.03 Public network to public network calling via the private network has relatively small usage and savings, so it does not justify change to the private network transmission plan. Instead, gain is applied on the access trunks, which are used exclusively for incoming calls. The gain is applied independently of the connection established through the private network. Thus, the amount of gain provided is a compromise **which** optimizes grade of service on the more important connections, but possibly degrades service on some others,

Gain Devices

6.04 The gain required is bi-directional between **2-wire** interfaces and is provided by devices called repeaters. There are two types of suitable bi-directional gain **devices**: fixed and switched gain. Fixed-gain devices are extremely sensitive to impedance mismatches at the **2-wire** interfaces, such mismatches can **cause** oscillation. Switched gain overcomes the oscillation problem, but introduces speech impairments due to the switching action,

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Fixed Gain

6.05 Bi-directional fixed gain can be implemented either by an amplifier on the **2-wire** path or two unidirectional amplifiers in a **4-wire arrangement**, interfacing the 2-wire path through hybrids. Both schemes are sensitive to the impedances at the 2-wire interfaces. Adjustable matching networks are required to allow the device to interface a variety of facilities. The more gain required, the closer the impedances must be matched to the interfaces in order to prevent oscillation or poor transmission associated with near oscillation. The impedance at the interfaces is partly determined by the impedances of facilities **switched** into a connection. Ultimately, those impedances limit the practical gain,

Voice-Switched
Gain

6.06 The voice-switched gain amplifier avoids the stability problems by dynamically applying gain. Only one **direction** of transmission can have gain at a time. A loss equal to the gain is provided **in** the opposite transmission direction,

6.07 Gain is applied by monitoring for speech and applying gain in the talker-to-listener direction when speech is detected. In the idle mode, a small loss is inserted in both directions. Compensation for impedances is required in order for the direction sensing circuitry to function properly. Table 6-A compares fixed gain and voice-switched gain.

Gain with Com-
pression

6.08 On some connections, there is the possibility that signal levels will already be high and gain **may** increase the levels above FCC specified limits. Some manufacturers **provide** compression options to guard against exceeding FCC limits. *Speech* compression ensures that signal levels will not exceed a specified maximum: **-9 dBm** generally. When speech levels, if amplified without compression, exceed that level, the gain is dynamically reduced so the **output** level does not exceed **-3 dBm**. The **-9 dBm** limit is specified in FCC requirements.

6.09 Gain **with** compression is possible on both fixed gain and voice-switched gain repeaters and does not change the adjustment procedure.

Table 6-A
COMPARISON OF FIXED GAIN AND VOICE-SWITCHED GAIN

FIXED GAIN	VOICE-SWITCHED GAIN
Does not degrade speech by gain switching action.	Gain switching action may degrade speed.
Must permanently attach to a trunk and be adjusted for its impedance,	May be switched into connections as required, but is affected somewhat by facility impedance,
Not sensitive to voice levels,	May require voice activation adjustment to provide required sensitivity.
May oscillate under certain circumstances.	"Unconditionally" stable.
"Transparent" to voiceband data and DTMF signaling,	Must be switched out for voice-band data. May impair DTMF signaling.

- Recommendation 6.10 **When** the **required** test equipment is available and the type of facility provided by the **telco** is known, fixed-gain units should be used because the performance is superior. **However,** switched-gain units should be used **where** the required transmission performance is not so critical, or the more complex alignment procedure for fixed-gain units cannot be performed.
- Adjustment 6.11 The gain should initially be set to **6 dB** in each transmission direction. This value of gain is a compromise between optimized grade of service and practical considerations to avoid oscillation and other transmission impairments. Based on experience, the gain may then be adjusted to greater or less than **6 dB**.

Application to
Trunks

5.12 To minimize the effect on the overall **trans-**mission plan, the gain units should be installed only on trunks **which** are used to access the **net-**work primarily for calling via **tie** trunk facilities or to off-network destinations. Calls originated on-network are to be blocked from accessing these trunks.

6.13 Arrange one-way incoming DISA **C.O.** trunk **groups,** using ground start trunks. The trunks are used for calling over tie trunks and off-network. Calls terminating on the same PBX should be placed via the attendant, OK via DID trunks, if provided.

6.14 **When** the **DISA** feature is equipped on DID trunks, do not provide gain units. DID trunks carry traffic both to station sets and the DISA feature, and gain is not desirable for the traffic to station sets. Instead, a **new** ground-start **C.O.** trunk group should be added and the DISA feature moved **to** that group. Gain units can then be provided on that group,

7. DEFINITIONS

7.01 Following are definitions of the **terms** used within this practice.

Public Network
Facilities

7.02 PBX Central Office Trunks **PBX** Central Office (CO) trunks connect the PBX switch to the central office which serves the PBX location. The trunks appear as station lines at the central office **equipment**.

7.03 Foreign Exchange Trunk. A Foreign **Exchange** (FEX) trunk provides a direct connection between a PBX switch and a remote central office other than the central office which serves the location of the **PBX**.

7.04 Wide Area Telecommunications Service, **Wide Area Telecommunications Service (WATS)** provides a bulk-rate service for incoming or outgoing toll calls **within** selected geographical regions (bands). A **WATS** trunk must **terminate** on a central office equipped to provide EATS, which may be the office that would normally serve that customer, or it may be a **remote** central office.

(a) Outgoing WATS. An outgoing **WATS (OUTWATS)** trunk is **used** exclusively for outgoing bulk-rate calls from a customer's **PEX** to a **defined** geographical area via the toll network.

(b) Incoming WATS. An incoming **WATS (INWATS)** trunk is used exclusively for incoming calls **from** a defined geographical area to a customer's **PBX**.

2-Hire Facility

7.05 A 2-wire facility is characterized by supporting transmission in two directions simultaneously, where the only method of separating the **two** signals is by the propagation directions. Impedance **mismatches** cause signal energy passing in each direction to mix with the signal passing in the opposite direction.

4-Wire Facility

7.05 A **4-wire** facility supports transmission in two directions, but isolates the signals by frequency division, time division, space division, or other techniques that enable reflections to occur without causing the signals to mix together. A facility is also called **4-wire** if its interfaces to other equipment meet this 4-wire criteria (even if **2-wire** facilities are used internally) as long as crosstalk between the **two** transmission directions is negligible as measured at the interface.

PBX Types

7.07 **Main PBX.** A **main PBX** is one which has a **Directory Number (DN)** and can connect PBX stations to the public network for both incoming and outgoing calls. A main PBX can have an associated **satellite PBX**. A main PBX can be part of a **Tandem Tie Trunk Network (TTTN)**. If the main PBX provides tandem switching for tie trunks, it is called a **tandem PBX**. In the context of ESN, a main **PBX** has tie trunks to only one node.

7.08 **Satellite PBX.** A satellite PBX has no direct **incoming** connection from the public network. All **incoming** calls are routed **from** an associated main PBX over tie trunks. This definition places no restrictions on the handling of outgoing calls from the satellite **PBX**. A satellite PBX can have one-way **outgoing** trunks to the central office, in addition to outgoing service on trunks to the main **PBX**. In the context of ESN, a satellite PBX has no direct trunks to a node, however, calls to the node can be made through the main PBX.

7.09 **Tributary PBX.** The only difference between a **satellite and tributary PBX** is that the tributary **PBX** has direct incoming connection from the public network.

7.10 **Tandem PBX.** A **tandem** PBX is used as an intermediate switching point in a TTTN to connect tie trunks together, in addition to the usual PBX functions,

7.11 **ESN Node PBX.** An ESN node PBX is an SL-1 switch equipped with the node software package of ESN. It performs tandem switching with software-controlled alternate route selection to bypass busy trunk groups.

7.12 ESN Main PBX. An ESN main PBX is an SL-1 switch with the ESN main software package. It performs tandem switching between a node and public **network trunks**, and between a node and the **main's** satellite and tributary switches. An ESN main PBX has outgoing tie trunks to only one node.

PBX Networks

7.13 Tandem Tie Trunk Network. A Tandem Tie Trunk Network (**TTN**) is a **switched** customer network which uses tie trunks to interconnect PBX switches in different locations. Calls are routed between the switches by progressive dialing of access codes.

7.14 Common Control Switching Arrangements. A **Common Control** Switching Arrangement (CCSA) is a Bell System offering which divides equipment between PBX **switches** and **tandem** switches. (The tandem switches are located on telco premises, and perform only a tandem switching function.) Network calls by a user at a PBX switch are made by dialing an access code (typically the digit '8') which connects the **user** to the telco **tandem** switch. The user then dials a '1'-digit number to reach the desired station at another connected switch. Routing between tandem switches is performed under **common** control.

7.15 Electronic Switched Network. An Electronic Switched Network (ESN) is a Northern **Telecom** private network offering which has a **dialing** plan similar to that used in a CCSA. However, tandem **switching** functions are performed by PBX **switches** located on the customer's premises. Fewer restrictions on tie trunk routes are required by ESN than by the CCSA. CCSA requires that at least **two** tie trunks be connected in tandem to route a call between a pair of PBX switches. ESN uses one tie trunk to perform the same routing wherever there is sufficient traffic to justify the route.

Transmission Level Point

7.16 As an analog signal passes over a **transmission** facility, it encounters gains and losses which are part of the facility design and net out to zero. For example, a facility which uses a **cable** may have loss in the cable and may compensate for this loss by **gain** in one or more amplifiers. On the other hand, some losses are introduced to improve grade of service and are not recovered. The term Transmission **Level** Point (TLP) is used to discriminate a gain or loss which is recovered from one which is not.

7.17 TLP requires definition of a point in a **transmission system** as a reference level. Once **this** point is defined, the reference levels at other points can be derived. For example, the input to a transmission system might be defined as a **-2 TLP**. Assume that the signal passes through a **10 dB** amplifier which compensates for loss elsewhere. The output of the amplifier is at a **+8 TLP** (**[-2] + [+10]**). A **-10 dBm** signal at the input **will** be **0 dBm** at the output. The same signal is **-8 dBm0** (**8 dB below** the reference level) at the **system input**. It is **also -8 dBm0** at the amplifier output, *The* signal has the same level, referenced to the TLP, because the gain in this case is offset by loss elsewhere, and does not show up as gain or loss at the signal destination. A second example is a transmission system with a **2 dB** deliberate **loss**. The input and output **TLP's** are **-2**. A signal in at **-10 dBm** is **-8 dBm0**. The signal leaves the system with a level of **-12 dBm**, or **-10 dBm0**. The difference in level, even though the TLP is the same, is the loss introduced deliberately in the facility.

7.18 **2-wire** and **4-wire** trunks in PBX networks are usually **-2 TLP**. However, some 4-wire trunks have a **-16 TLP** at the facility input and a **+7 TLP** at the facility output. If the trunk has **0 dB** loss, an input signal of **-26 dBm** (**which** is **-10 dBm0**), is received at a level of **-3 dBm** (**23 dB** gain) **which** is also **-10 dBm0**. This means that loss elsewhere in the connection cancels out the **23 dB** in the facility.

INTERGRATED SERVICES NETWORK ←

MERIDIAN SL-1* ←

REMOTE PERIPHERAL EQUIPMENT
DESCRIPTION, INSTALLATION, CONNECTIONS AND TESTING

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Reason for Reissue: To add information on the disabling of QBL14 trip circuitry and **include** numerous miscellaneous changes. Arrows in the margins indicate changes.

1. DESCRIPTION

1.01 Remote Peripheral Equipment (**RPE**) is **used** to increase the 50 ft (**15 m**) range of the loop between the Common Equipment (**CE**) and Peripheral Equipment (**PE**) shelves in an SL-1 system.

1.02 The increased **range** enables the PE to be placed closer to the stations it serves which in turn increases the serving range of the SL-1 system.

1.03 The RPE employs a carrier **link** between the local and remote end. The carrier link may consist of one of the following:

- a wire pair with no repeaters or other interface (“In-House” **RPE**)
 - | a digital carrier link meeting **T1 specifications** (such as Northern Telecom’s **LD-1**)
 - | a microwave radio link (which meets **T1** interfacing specifications)
 - | a fiber-optic link (which meets **T1** interfacing specifications).

2. EQUIPMENT CONFIGURATION

- Carrier System 2.01 A block diagram of the basic RPE system is shown in Fig. 2-1. A 1.544 MB/s multiplexed digital carrier system (such as LD-1) or microwave radio link is required for each RPE system. A maximum of two network loops may be connected through two RPE carrier shelves (one shelf at the local equipment location and one at the remote location). A complete RPE shelf (two network loops) requires four digital carrier lines.
- Carrier Shelf 2.02 The same type carrier shelf for the RPE equipment is used at the local and remote locations. The QSD 6 (left-hand mounted) and QSD11 (right-hand mounted) shelves may be mounted in any SL-1 PE cabinet. The power supply connector is a 2-pin type. Each shelf has a power converter pack to derive its required voltages from a -48 V supply provided by a QBL14 power distribution box. All cables from carrier shelves are connectorized.
- 2.03 Each loop services a maximum of four PE shelves. RPE network loops are fully assigned to RPE use, and no other PE shelves can be served by these loops.
- 2.04 Each loop requires four cable pairs (two carrier lines) between the carrier shelf and the carrier system for transmission and signaling. A maximum of two cable pairs are required for maintenance purposes. These are the Order Wire (OW) and Fault Locating (FLP) and are optional depending on the distance between the carrier shelves and on the location of the Office Repeater Bay (ORB) in the system (see Fig. 6-4).
- 2.05 Each RPE system requires at least one ORB (Fig. 2-2) and line repeaters unless the remote equipment is within about 2500 ft (726 m) of the SL-1 equipment. The location of an ORB at both local and remote ends of the carrier line is strongly recommended. This effectively allows isolation of the carrier span from the SL-1.
- 2.06 An ORB provides the following:
- ▮ span line powering
 - error monitoring
 - fault-locate system access
 - order-wire termination with DDD access
 - line looping.
- 2.07 A typical RPE configuration is shown in Fig. 2-3. Each RPE system requires a carrier shelf at the local and remote locations. They are cabled to the SL-1 cross-connect terminal via two NE-A25B cables.
- 2.08 Carrier shelves at the local equipment location and at the remote location are connected to QPC50 network packs and PE shelves respectively via NE-A18QA connector cables (see Fig. 4-1 and 4-2).

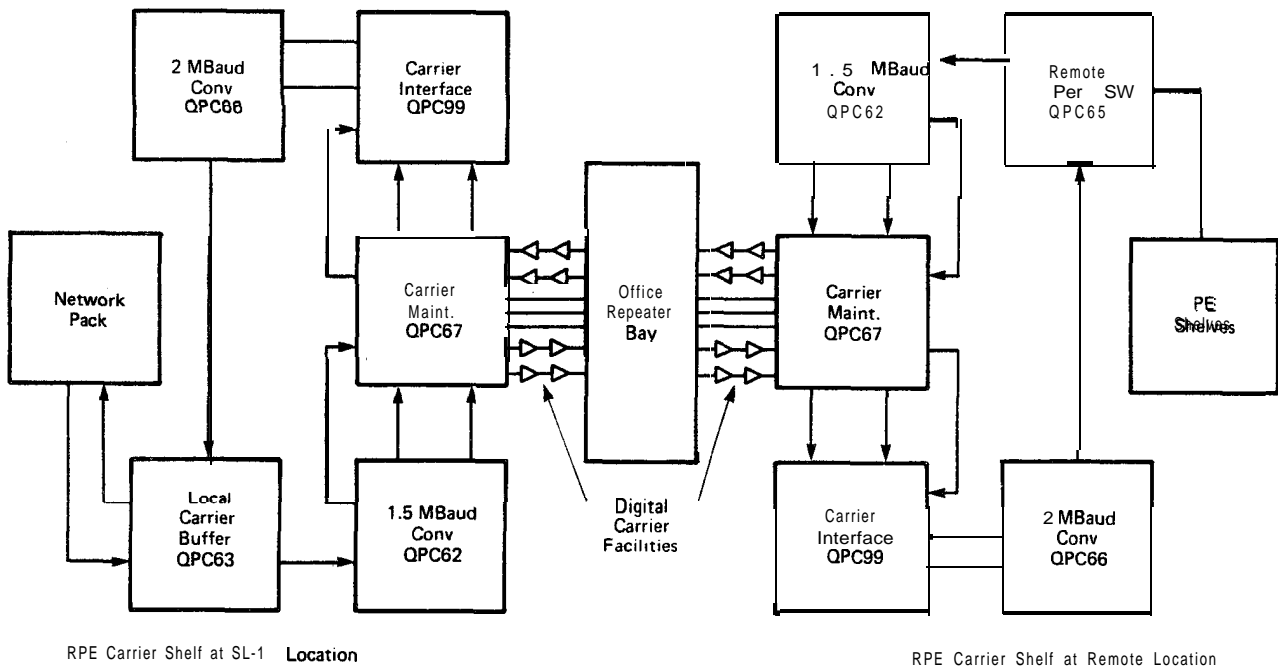


Fig. 2-1
RPE Block Diagram

2.09 Peripheral Equipment shelves at the remote location are connected to the cross-connect terminal via four NE-A25B connector cables in the same manner as regular SL-1 PE shelves. See 553-2YY1-210, -215 for cabling and terminating instructions.

2.10 Two network' loops connect to the carrier shelf and they can each serve a maximum of four PE shelves. Circuit pack positions 1 to 4 and 5 to 8 serve network loops X and Y respectively.

→ Circuit Packs

2.11 The same circuit packs are used in the local and remote carrier shelves (see Fig 2-3) except for the QPC63 (local) and the QPC65 (remote) packs. All circuit packs have designated slot positions' to function properly. The QPC62 and QPC99 packs have option switches on their circuit boards. (See Fig 4-4 and 4-6). Circuit pack handling procedures are described in 553-2001-205.

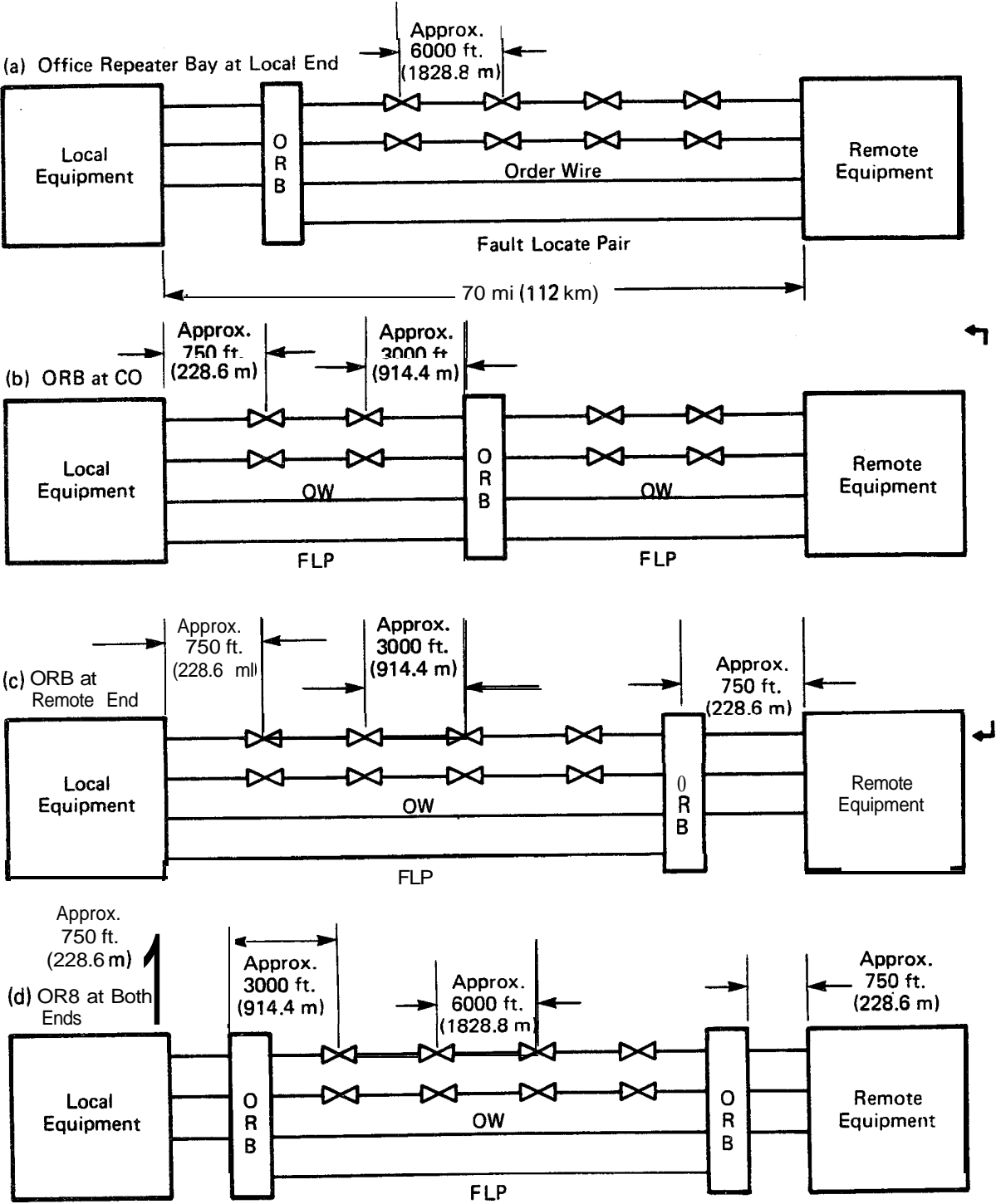


Fig. 2-2
Possible OR6 Locations in RPE System

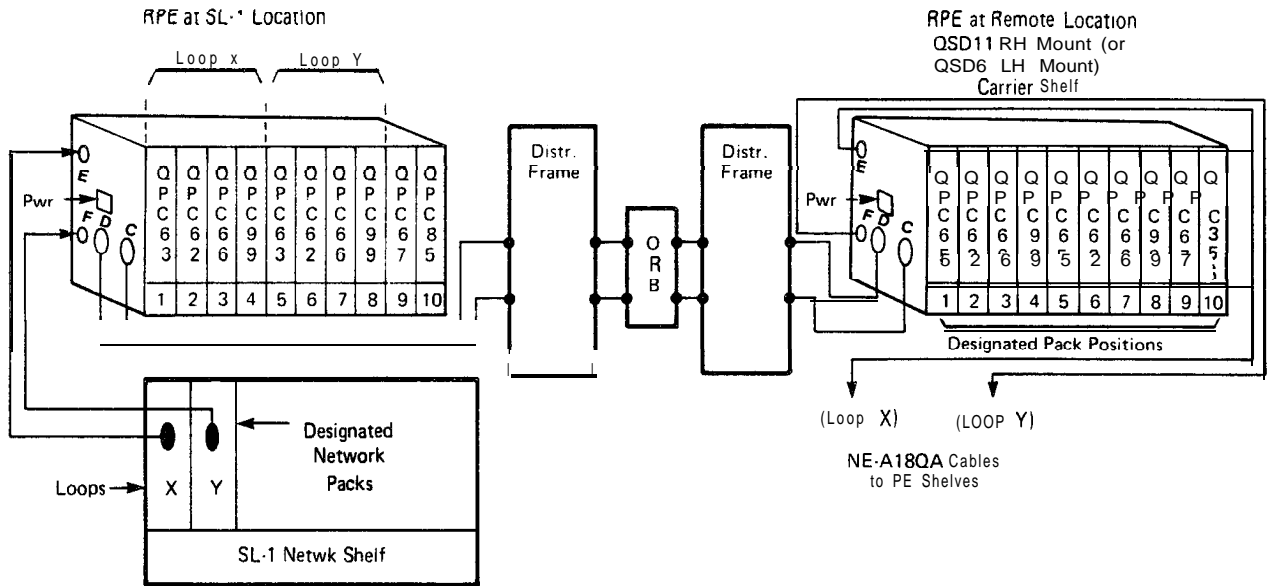


Fig. 2-3
Typical RPE Equipment Configuration

→ Emergency Transfer

2.12 Designated **500-type** sets are cross-connected through emergency transfer units to outgoing trunks at the remote location. The sets are connected to these trunks when the normally operated relays of the emergency transfer units release as a result of any of the following

- loss of -48 V carrier shelf power
- loss of -48 or ± 10 V (under control of the QPC84 Power Monitor circuit pack)
- Carrier failure (network loops are controlled independently)
- manual operation of emergency transfer switch on consoles (Note)
- 1 manual operation of emergency transfer switch on the QPC84 circuit pack (Note).

Note: This will not affect a remote location.

3. EQUIPMENT DESCRIPTION

- CABINETS**
- 3.01 Standard **SL-1** cabinets are used to house RPE equipment. A **QCA8**, **QCA37** or **QCA74** cabinet is required at the remote location to provide power for the **QBL14** power distribution box that powers the carrier shelves. A **QCA6**, **QCA8**, **QCA23**, **QCA37** or **QCA74**, cabinet may be used at the local location to house the **QBL14** power distribution box.
- 3.02 RPE carrier shelves may be installed in any **SL-1** cabinet (except **QCA60**). They must be within 12.5 cable feet (380 mm) of the **QBL-14** power distribution box.
- CARRIER SHELVES**
- 3.03 The following carrier shelves are available:
- ▮ **QSD6** left-hand mount shelf
 - **QSD11** right-hand mount shelf.
- 3.04 Purpose. The carrier shelf accommodates the circuit packs listed in Table 3-A. These packs can only function in the designated pack positions shown in Fig 2-3.

Table 3-A
CARRIER SHELF CIRCUIT PACKS

LOCATION	CODE	DESCRIPTION	QUANTITY
Local and Remote	QPC85/QPC190/QPC355	5/12-Converter	1 per shelf (Note)
	QPC67	Carrier Maintenance	1 per shelf
	QPC62	1.5 M Baud Converter	2 per network loop
	QPC66	2.0 M Baud Converter	2 per network loop
	QPC99	Carrier Interface	2 per network loop
Local only	QPC63	Local Carrier Buffer	1 per network loop
Remote only	QPC65	Remote Peripheral Switch	1 per network loop

Note: **QPC190/QPC355** circuit packs can only be used in **QSD11B** series B and **QSD6B** series B shelves or **QSD11C** and **QSD6C** shelves. **QSD11B** series A and **QSD6B** series A shelves only work with a **QPC85** circuit pack. Using the **QPC190** in the earlier vintage shelves will damage both the **QPC190** itself and the **QPC99** (Carrier Interface).

3.05 Quantity. One shelf is required at the local and remote locations for each two network loops.

3.06 Location. Any PE shelf position in a cabinet must be within 12.5 cable ft (3800 mm) of the **QBL14** power distribution box that powers the shelf.

3.07 Features. Carrier shelf features are as follows:

- | steel and aluminum construction
- | printed circuit backpanel
- fully connectorized power and signaling connections
- | international rack mounting standards (19 in [483 mm])
- | approximate weight: 35 lbs (15.9 kg) fully equipped.

PE SHELVES

3.08 Left- or right-hand mount shelves are used. See 553-2001-150 for their description.

LOCAL CARRIER
→ BUFFER - **QPC63**

3.09 Purpose. Generates from the 2.048 MHz clock a 1.544 MHz clock. Decodes and provides enables for outgoing and incoming data. Delays the data incoming from the carrier so that its frame relative to the outgoing data frame is equivalent to that returning from a peripheral buffer. Relays line status information to the processor. Decodes line control information from the processor.

3.10 Quantity. One for each network loop connected to the carrier shelf at the local equipment location.

3.11 Location. Position 1 and 5 of the carrier shelf at the local equipment location. Position 1 for the first network loop connected to the shelf; position 5 for the second network loop connected.

1.5 M BAUD
CONVERTER - **QPC62**

3.12 Purpose. Converts an SL-1 loop into two carrier loops. Contains switch-selectable line equalizers.

Note: **QPC62C** and converters of later vintage must be used when the 12 V option setting is required.

3.13 Quantity. Two for each network loop, one in the local shelf, one in the remote shelf.

3.14 Location. Position 2 for the first network loop; position 6 for the second network loop in each carrier shelf.

2 M BAUD CONVERTER
- **QPC66**

3.15 Purpose. Converts two carrier loops into an SL-1 loop.

3.16 Quantity. Two for each network loop, (one in the local and remote carrier shelves).

3.17 Location. Position 3 for the first network loop, and position 7 for the second loop in each carrier shelf.

CARRIER INTERFACE -
QPC99

3.18 Purpose.

- (a) Contains two carrier line receivers with 7.5 dB pads built in. Converts the bipolar line signals into TTL level signals. Provides facilities for LD-1 carrier looping, monitors system, and invokes emergency transfer if carrier fails.

- (b) Contains an option switch on its circuit board. In circuit packs of vintages F and subsequent, the -7.5 dB pads are switch selectable. The settings on switches 1 to 4 and 7 to 12 determine the location of the pack and should be set as shown in Fig. 4-5. Switches 5 and 6 determine **loopback** conditions. With 5 closed, the loop carrier is looped for an additional 8 s. With 6 closed, **loopback** occurs when dc is on the Fault-Locate Pair and bipolar violations occur on carrier. With 6 open, **loopback** occurs when:
 - | DC is on the Fault-Locate Pair and TRIOS present
 - | DC is on the Fault-Locate Pair and excessive Bipolar Violations (BPV)
 - TRIOS present.
- (c) The later vintages also have a ROUT jack for each channel to allow a test signal to input into the system. A Manual Loop Back (MLB) ← switch is also added to allow looping of the **system** for fault clearing. All other features of the earlier vintages are retained.

3.19 Quantity. Two for each network loop (one in each carrier shelf).

3.20 Location. Position 4 for the first network loop and position 8 for the second loop in each carrier shelf.

CARRIER MAINTENANCE - QPC67

3.21 Purpose. Contains an M-type (3017 Hz) fault-locate filter for fault-locate testing in the LD-1 system, dc detection circuitry for the fault-locate pair, and carrier **loopback** relays to facilitate software maintenance testing. Terminates and gives access to the order wire pair via a jack and binding posts on the faceplate.

3.22 Quantity. One in each carrier shelf.

3.23 Location. Position 9 in the carrier shelf.

5/12-V CONVERTER - QPC190/QPC355/QPC85

3.24 Purpose. Converts -48 V dc to i- 12 V and + 5 V dc for the carrier shelves.

Caution: **QPC190/QPC355** circuit packs can only be used in **QSD11B** series B and **QSD6B** series B shelves or **QSD11C** and **QSD6C** shelves. **QSD11B** series A and **QSD6B** series A shelves only work with a QPC85 circuit pack. Using the **QPC190** in the earlier vintage shelves will damage both the **QPC190** itself and the **QPC99** (Carrier Interface).

3.25 Quantity. One for each carrier shelf.

3.26 Location. Position 10 in each carrier shelf.

REMOTE PERIPHERAL SWITCH PACK - QPC85

3.27 Purpose. Each SL-1 loop at a remote site has a Remote Peripheral Switch associated with it. The card provides:

- Shelf, card and line enables plus the bypass bit to the shelves it serves at the remote site.

- Cyclic scanning of the terminals it serves for incoming signaling messages
- | Monitoring of time slot 0 for outgoing (from Peripheral Signaling Card to BPS or terminal) messages
- | Assembling ingoing (**RPS** to **PS**) messages

3.28 Quantity. **One** for each network.

3.29 Location. Position 1 and 5 of the remote carrier shelf for the first and second network loops respectively.

CABLES

3.30 The following cables are used in RPE installations:

- | NE-A18Q to interconnect the local carrier shelf to a network pack and to interconnect the PE shelves to each other and to the remote carrier shelf.
- | NE-A25B cables are used to connect jacks C and D of the local and remote carrier shelves to the cross-connect terminal and to connect jacks A, B, C, and D of each PE shelf to the cross-connect terminal. (See Fig 4-1 and 4-2.)

QBL14 POWER
DISTRIBUTION BOX

3.31 Purpose. Distributes -48 V to a maximum of 4 carrier shelves. Equipped with **circuits** to provide a low voltage (-42 V) disconnect.

3.32 Quantity. One for every 4 carrier shelves.

3.33 Location. Above the QUX3 power distribution unit or above the **QBL5** power distribution box in a **QCA8** cabinet. May also be installed in a QCA6 cabinet above the **QUX1** or **QBL3** units. In QCA28 and **QCA37** cabinets the unit can be mounted in any unequipped shelf **location**.

4. ASSEMBLY AND INSTALLATION

4.01 RPE may be shipped fully assembled in PE cabinets, or shelves and circuit packs may be shipped individually packaged for installation in existing PE cabinets.

4.02 Refer to 553-2001-205 for the unpacking and handling of equipment before starting installation procedures. When new cabinets are to be installed, refer to **553-2YY1-210** for grounding and power requirements and wiring diagrams, cabinet installation and inspection procedures.

4.03 Charts 4-1 and 4-2 give the installation and cabling procedures for RPE shelves at the local and remote equipment locations respectively. Chart 4-3 provides the remote alarm installation. Fig. 4-1 and 4-2 illustrate RPE shelf cabling.

4.04 Sets, consoles, and add-on-modules are installed and connected as described in **553-2YY1-211** and -215.

Chart 4-1 LOCAL RPE INSTALLATION

STEP PROCEDURE

- 1 Install **QSD6** or **QSD11** (left-hand or right-hand mounted) carrier shelf with no circuit packs inserted. (See **553-2YY1-210** for shelf installation procedures.)
- 2 Install an NE-A18Q cable from carrier shelf connector jacks E and F to each network pack (Fig. 4-1).
- 3 Install two NE-A25B cables from shelf connector jacks C and D to cross-connect terminal (Fig. 4-2).
- 4 Insert **QPC85**, **QPC190** or **QPC355 5/12-Converter** pack (depending on shelf vintage) in position 10 of the shelf.

Caution: The **QPC190** and **QPC99** (Carrier Interface) will be damaged if the **QPC190** is used on earlier vintage shelves. Refer to the 5112 V converter description herein for associated shelves.

- 5 Perform cabinet inspection procedures (see **553-2YY1-210**) if a new cabinet was installed.
- 6 If there is no existing RPE equipment, then it is necessary to install a QBL14 power distribution box (see Fig. 4-3). Refer to note at end of Chart 4-2 for trip circuitry information.
- 7 Install the power cable (supplied with carrier shelf); connecting it from the QBL14 unit to the carrier shelf power jack (see Fig. 4-3).
- 8 Terminate cables as described in Part 6.
- 9 Connect power to RPE carrier shelf.

Chart **4-1** Continued
LOCAL RPE INSTALLATION

STEP PROCEDURE

- 10 Check option switch settings and header pin strapping on **QPC62**, **QPC66** and **QPC99** packs (See 553-2201-211).
 - 11 Insert all other circuit packs in their designated positions. (See Fig. 2-3.)
-

Chart 4-2
REMOTE RPE INSTALLATION

STEP PROCEDURE

- 1 Install **QSD6** or **QSD11** (left-hand or right-hand mounted) carrier shelf.
- 2 Install (left-hand or right-hand mounted) PE shelves.
- 3 Install two NE-A25B cables from carrier shelf connector jacks C and D to the cross-connect terminal (Fig. 4-2). Terminate cables as described in Part 6.
- 4 Install NE-A18QA cables from carrier shelf jacks E and F to PE shelves for network loops X and Y (Fig. 4-2).
- 5 Install NE-A18QA cables between PE shelves (Fig. 4-2).
- 6 Install four NE-A25B type cables from jacks A, B, C, and D of each PE shelf to cross-connect terminal. See **553-2YY1-215** for designations and terminating procedures.
- 7 Perform cabinet inspection procedures. See **553-2YY1-210** if a new cabinet was installed.
- 8 If there is no existing RPE equipment, then it is necessary to install a QBL14 power distribution box (see Fig. 4-3).
- 9 Install the power cable (supplied with carrier shelf); connecting it from the QBL14 unit to the carrier shelf power jack (see Fig. 4-3). See note for trip circuitry information.
- 10 Insert **QPC85**, **QPC190** or **QPC355** pack (depending on shelf vintage) in shelf position 10.
 - Caution: The **QPC190** and **QPC99** (Carrier Interface) will be damaged if the **QPC190** is used on earlier vintage shelves. Refer to the 5112 V converter description herein for **associatied** shelves.
- 11 Connect power to RPE carrier shelf.
- 12 Check option switch settings and header pin strapping on **QPC62**, **QPC66** and **QPC99** packs (See 553-2201-211).

Chart 4-2 Continued
REMOTE RPE INSTALLATION

STEP PROCEDURE

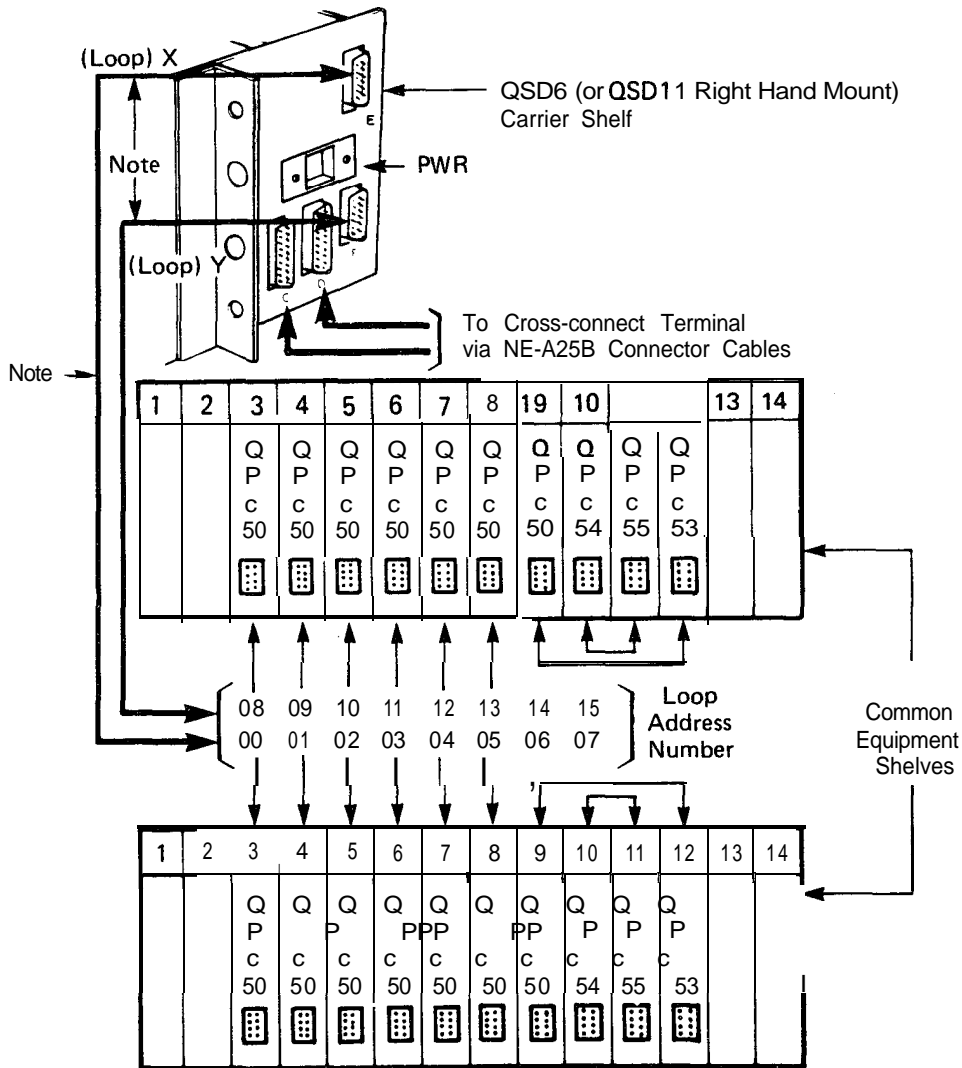
- 13 Insert all other circuit packs in their designated positions. (See Fig. 2-3.)

Note: QBL14 Power Distribution Box located in the Remote Peripheral **Equipment** (RPE) contains undervoltage detection circuitry that trips the input circuit breaker when the dc input voltage drops below -42 volts. This circuit prevents the RPE from dropping the battery voltage below -42 volts which would cause permanent battery damage.

Where battery backup is not provided a short interruption of the main ac input power will cause the input breaker to trip. This interruption keeps the power to the remote RPE equipment off when the ac power source is restored.

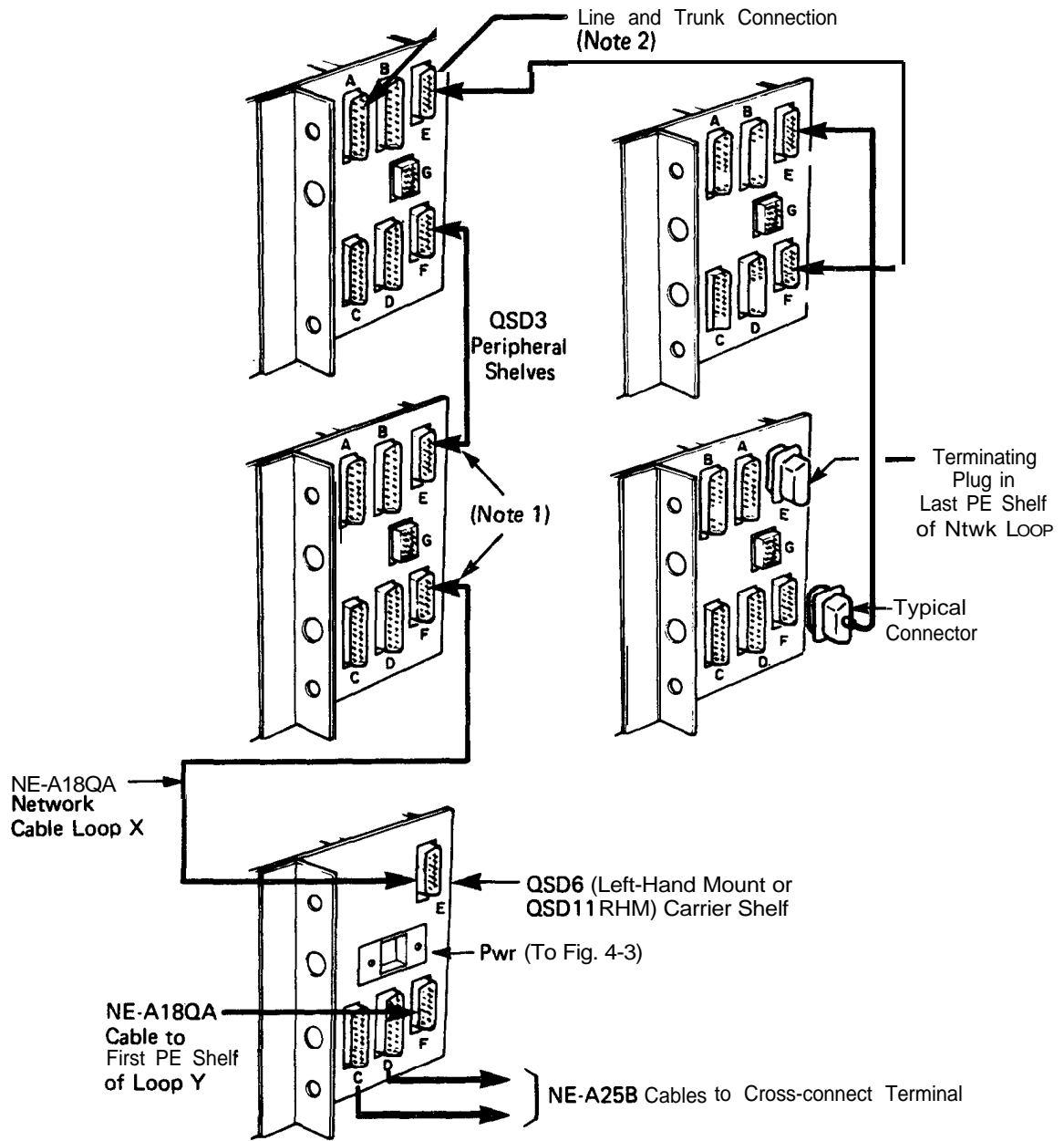
In this situation (no battery backup) the trip circuitry can be disabled by removing the single wire from the D terminal on the input breaker of the QBL14 (See Fig. 4-8). The disconnected end should be wrapped with insulating tape and the wire dressed back.

The foregoing also applies to the QBL14 used with the local carrier shelves if reserve battery is not installed at the main location.



→ Note: NE-A18QA cables are available in the following lengths:
 1, 2, 4, 10, 15, 20, 25, 30, 35, 40, 45 feet
 (.3, .6, 1.2, 3.0, 4.6, 6.0, 7.6, 9.1, 10.7, 12.1, 13.7, m)
 Total length of cable from network pack to RPE carrier shelf must not exceed 45 ft. (13.7 m)

Fig. 4-1
 Local End Network Loop and Carrier Shelf Cabling



Note 1: NE-A18QA cables are available in the following lengths:
 1, 2, 4, 10, 15, 20, 25, 30, 35, 40, 45 feet.

(.3, .6, 1.2, 3.0, 4.6, 6.0, 7.6, 9.1, 10.9, 12.1, 13.7 m)

Total Length of cable from PE Shelf to RPE Carrier Shelf
 must not exceed 45 ft. (13.7 m)

Note 2: 1 of 4 connectors (A,B,C,D) cabled to cross-connect
 terminal via NE-A258 cables.

Note 3: Connector designations are similar on other type
 Peshelves

Fig. 4-2
 Remote End PE Shelf and Carrier Shelf Cabling

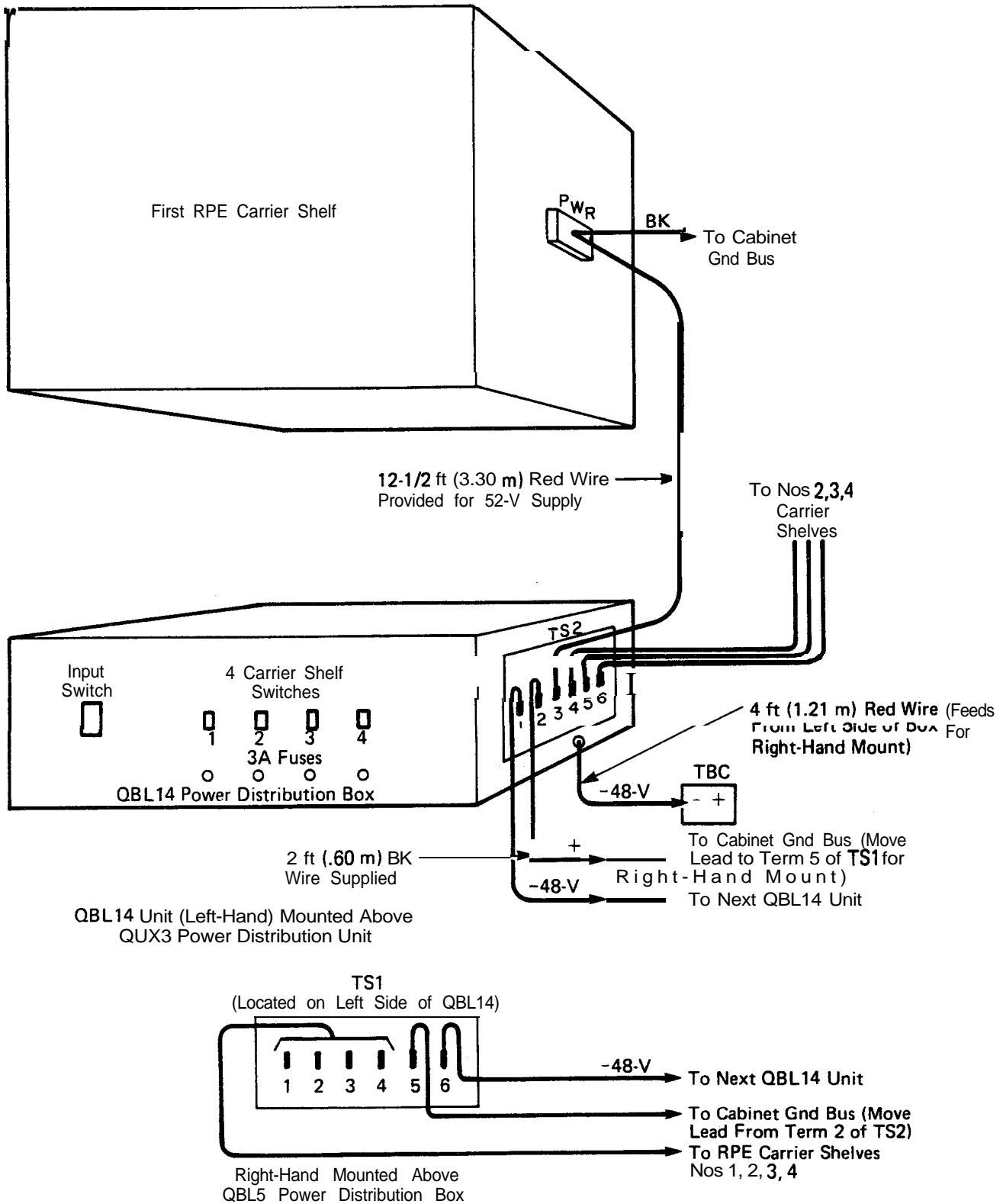
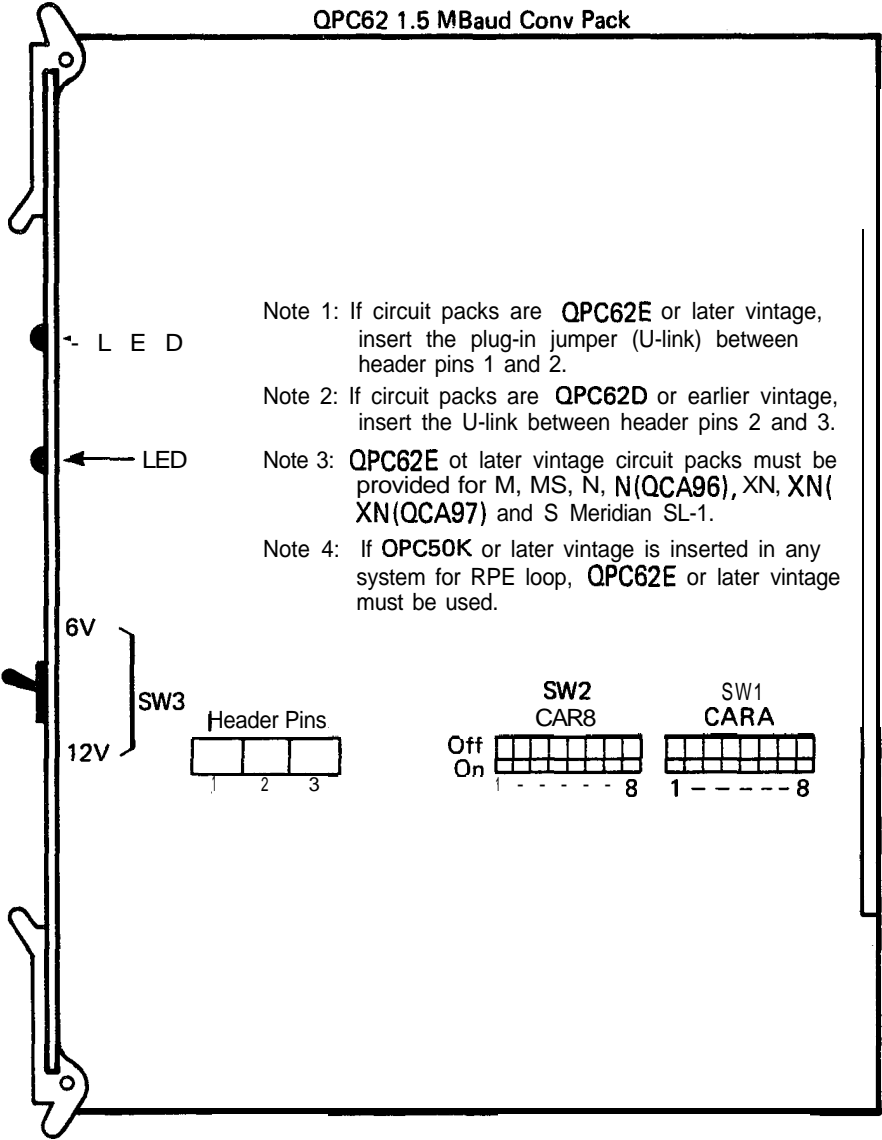


Fig. 4-3
QBL14 Power Distribution Box Wiring and Connections

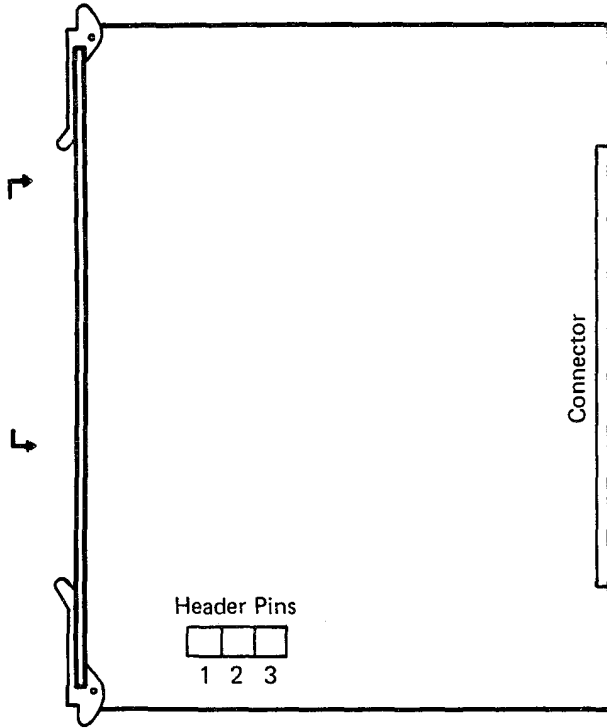


Note: Refer to 553-2201-211 for switch settings.

(III, 835)

Fig. 4-4
QPC62 1.5 M Baud Converter Pack Option Switch and Header Pin Locations and Settings

PRACTICE 553-2601-200



Note 1: If circuit packs are **QPC66D** or later vintage, insert the plug-in jumper (U-link) between header pins 1 and 2.

Note 2: If circuit packs are **QPC66C** or earlier vintage, insert the U-link between header pins 2 and 3.

Note 3: **QPC66D** or later vintage circuit packs must be provided for **SL-1M**, **MS**, **N**, **N** (**QCA96**), **XN**, **XN** (**QCA97**), or **S Meridian SL-1**.

Note 4: If **QPC50K** or later vintage is inserted in any system for RPE loop, **QCD66D** or later vintage must be used.

(Ill. 1098)

Fig. 4-5
CIPC66 2.0 M Baud Converter Pack Header Pin Strapping

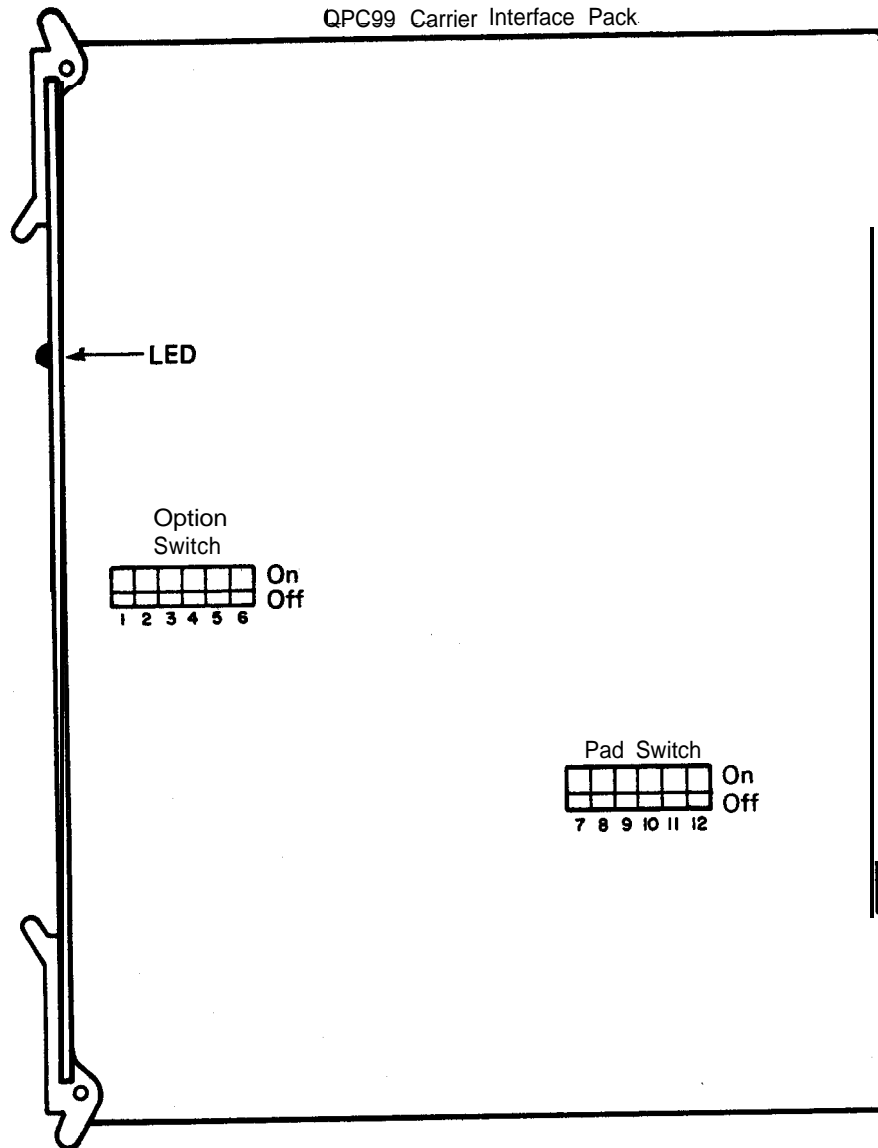
Chart 4-3
REMOTE ALARM INSTALLATION

In an RPE installation, a power failure at the remote end cannot be detected at the local end. To overcome this, a wire pair can be connected from the remote QPC84 power monitor to the alarm input of the local QPC84. This can be used to generate TTY and/or visual alarm indication of power failure at the remote end.

The procedures in this chart refer to Fig. 4-7.

STEP PROCEDURE

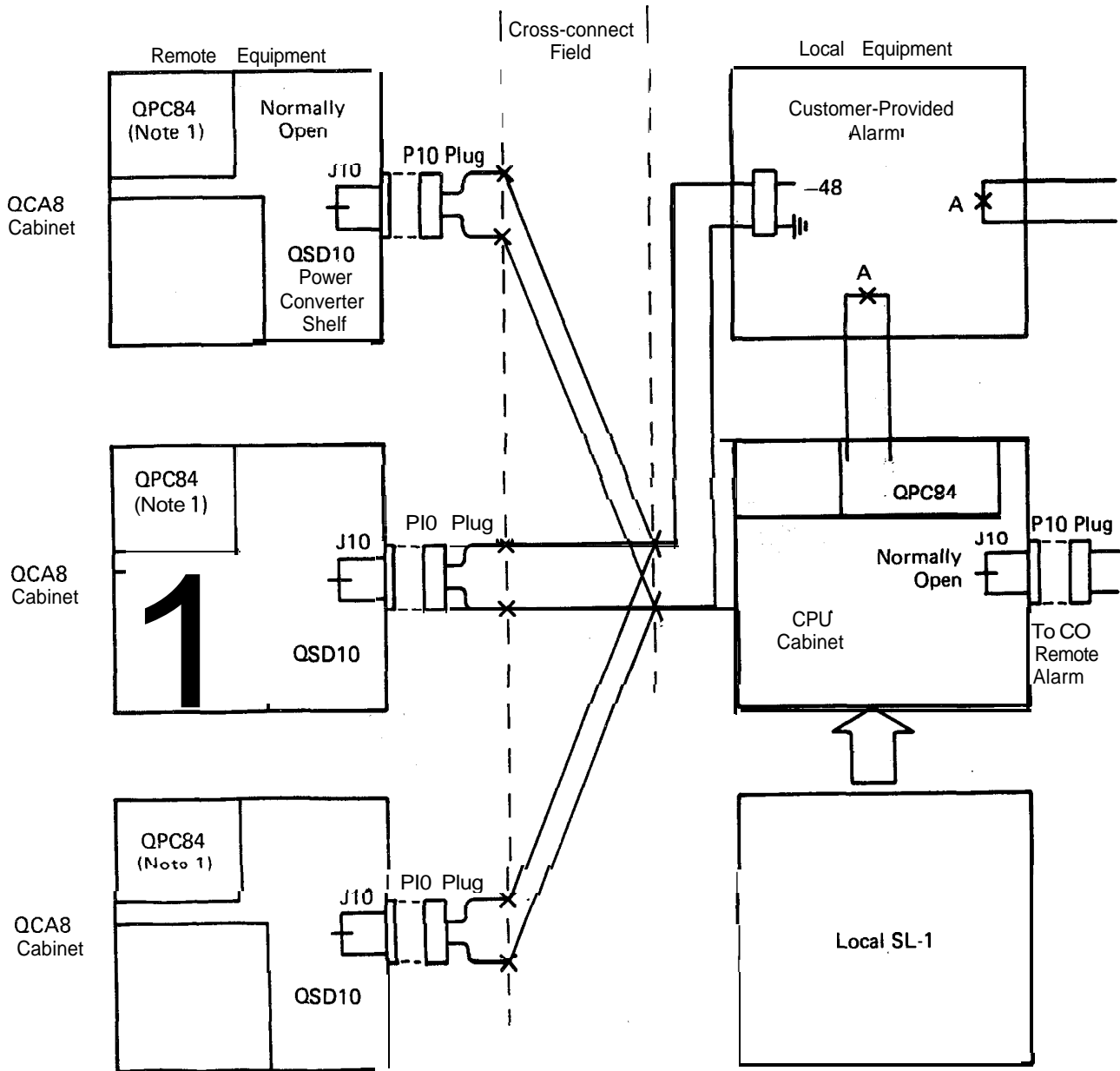
- 1 Using procedures given in **553-2YY1-210**, install connector cables (**NEA25B** or equivalent) from the **P10** plug of each remote **QCA8** to the cross-connect field.
- 2 Install and designate a **NEA25B** (or equivalent) connector cable from the cross-connect terminal to the main location.
- 3 At the main location, connect the two alarm leads (via the cross-connect terminal) to a customer provided alarm with provision for audio indication of remote power failure.
- 4 Connect 2 leads from the main alarm to **TS3** of the **QCA6** (**L** system), **QCA10** (**VL** system), **QCA23** (**LE** system), **QCA24** (**VLE** and **XL** systems), **QCA28** (**A** system), or **QCA37** (**SL-1M** system).



Note: Refer to 553-2201-221 for switch settings.

(Ill. 836)

Fig. 4-6
QPC99 Carrier Interface Pack Option Switch Settings



Notes:

1. The **QPC84** circuit pack, in the power converter shelf, provides LED indication of out-of-tolerance voltages and controls the system alarm indication.
2. The rectifier is present in L, LE, M and A systems only. When the QSD4 is not present, TS3 is located on OUT-type power units (for fans).
3. See 553-2YY1-210 and -215 for connecting information.

Fig. 4-7
Remote Alarm Connection

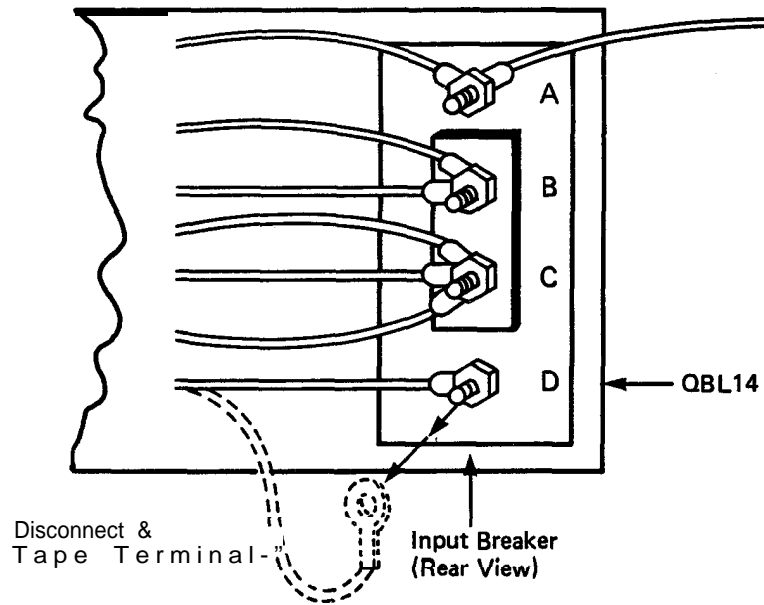


Fig. 4-8
Input Breaker of **OBL14** Power Distribution Box

5. CARRIER INTERFACE

LAND-BASED CARRIER

5.01 The SL-1 RPE hardware has been designed to interface with Northern Telecom's LD-1 carrier apparatus which conforms to the T-1 industry standard. Therefore any carrier system conforming to T-1 signaling standards should be able to interface with SL-1 RPE: Minor differences in carrier maintenance can generally be accommodated by option switches in the carrier interface pack.

GENERAL ENGINEERING CONSIDERATIONS

5.02 In addition to the **T1** carrier rules, the following rules also apply:

- The distance from local to remote equipment cannot exceed 70 miles (**112 km**).
- | Line repeaters are powered from the Office Repeater Bay (**ORB**) since no power is available from the SL-1 interface.
- | The SL-1 interface with the LD-1 carrier line contains an M-type fault location filter (**3017 Hz**) without level of polarity options. The fault location filter is powered by the SL-1 RPE equipment. No other M-type filters can be used in the same span.

5.03 If an ORB is being used, the cable **between** the ORB and the MDF should have the following characteristics

- | The impedance presented to the SL-1 equipment should be 100 ohms at **722 kHz**.
- | The total distance between the SL-1 and the ORB or first line repeater should not exceed 750 feet (**250 m**).
- Cable, designed for PCM or digital signals, should consist of **individually** shielded twisted pairs (e.g., NE-750A to **NE-759A-type cables**).

CARRIER SPECIFICATIONS

5.04 The SL-1 RPE is compatible with carrier facilities having the characteristics shown in Table 5-A and Fig. 5-1.

Table 5-A
CARRIER CHARACTERISTICS

Line Rate	1.544 Mb/s (± 200 b/s)
Signal	bipolar, 50% duty cycle
Output Level	<p>Option 1</p> <p>If the equipment interfaces directly with the carrier line, or with an office repeater bay which is less than 150 feet away from the common equipment, the positive and negative output pulse heights are 3 V $\pm 10\%$ and the imbalance between the positive and negative pulses is less than 5%.</p> <p>Option 2</p> <p>When interfacing with an Office Repeater Ray which is between 150 and 750 feet away from the common equipment, the positive and negative output pulse heights are 6 V $\pm 10\%$ and the imbalance between positive and negative pulses is less than 5%. With this configuration one of two possible equalizers is inserted in the carrier line, in order to make the total cable and equalizer loss about 6 dB. This will provide the required 3 V pulses at the ORB.</p>
Output Pulse Width	The output pulse width at half pulse height is 324 \pm 30 ns. Unbalance between positive and negative pulse width at half pulse height is less than ± 15 ns .
Output Rise and Fall Time	The output rise and fall time is less than 90 ns .
Overshoot	The overshoot at the trailing edge of the output pulse is between 20 and 40% of pulse height with decay to 10% or less of base line to peak overshoot within 400 ns.
Impedance	The nominal impedance at the line interface is 100 Ω .
Output Jitter	The maximum jitter on the digital output signal can be 30 ns RMS.
Input Level	The positive and negative input pulse height will be in the range of 0.07 to 3 V with a possible imbalance between positive pulses of 5%.
Input Jitter	The system can accommodate a low frequency input jitter of up to 20 ns RMS. Jitter with frequencies above 2 kHz should not be more than ± 50 ns peak.
Interpulse Time	The time between pulses in adjacent time slots will be 648 \pm 20 ns.

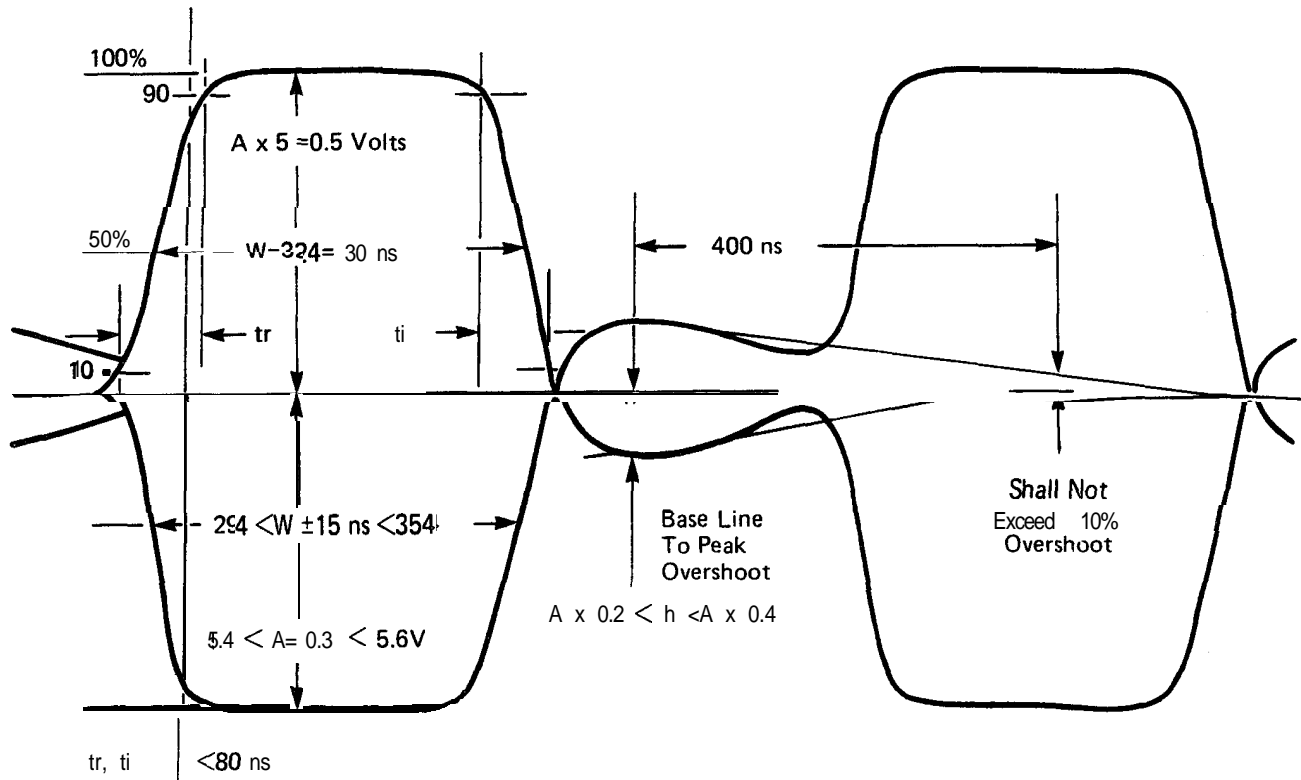


Fig. 5-1
Output Pulse Characteristics (6 V option selected and no output equalizer terminating into 100 ohm load)

IN-HOUSE RPE (IRPE)

5.05 Since there is a carrier repeater located in the **QPC99**, it is not necessary to use ORB's and extra repeaters between the local and remote ends of the RPE if the distance between them is typically less than 2500 ft (750 m). In this case the standard T-1 procedures for cable installation should be followed. Other considerations are as follows:

- The impedance presented to the SL-1 system should be 100 Ω at 772 kHz.
- Unless the cable effectively separates transmit and receive pairs, separate cables for transmit and receive should be used to avoid crosstalk.
- Bridges, taps and loading coils, and building-out capacitors are to be avoided.
- Cable environment can affect transmission line capacitance and resistance, causing impedance mismatches and signal reflections. Cable should therefore be clean and dry.

5.06 IRPE Maximum Distances. Permissible IRPE operating distances depend on total losses and noise between the local and remote locations.

- (a) If separate cables are used for transmit and receive directions and if no noise or crosstalk from other pairs or external sources reaches the signal pairs, the maximum allowable loss at 772 kHz is 26 dB. In this case the following approximate values apply:

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- 3200 ft (975 m) of 26 AWG wire
- | 3800 ft (1158 m) of 24 AWG wire
- 5100 ft (1554 m) of 22 AWG wire.

(b) Table 5-B shows the operating limits when using several carrier systems without any additional equipment in the same group.

Table 5-B
NUMBER OF 1.544 MB/S SYSTEMS IN ONE BINDER GROUP

WIRE GAUGE	NO OF RPE SYSTEMS PER BINDER GRP MAX. CABLE LENGTH (Ft [m])		
	4	6	12
26			
24	2500 (762)	2100 (640)	2100 (670)
22	3600 (1097)	3000 (914)	2700 (213)

(c) If only one twin carrier system and SL-1 equipment (e.g., SL-1 sets, 500 sets) are used in the same binder group, the following cable length limits apply:

- 2500 ft (762 m) of 26 AWG wire
- | 2800 ft (853 m) of 24 AWG wire
- 3600 ft (1097 m) of 22 AWG wire.

(d) If other high transient switching pairs are used in the same binder group, the maximum cable length is limited by LD-1 engineering rules for high noise environments and the following worst case limits apply:

- | 1900 ft (579 m) of 26 AWG wire
- 2100 ft (640 m) of 24 AWG wire
- | 2400 ft (732 m) of 22 AWG wire.

MICROWAVE RADIO

5.07 Cabling between the SL-1 and the carrier facility should meet the same criteria outlined for land-line carrier systems. In addition, the complete microwave system must also meet the overall limits for land-based carrier systems and conform to T1 interfacing specifications.

5.06 Distance Limits. As in land-line carrier systems, the distance at which the SL-1 can operate is governed by a time out in the peripheral signaling card of 1.5 ms when interrogating the peripheral equipment. This 1.5 ms value is the maximum total round-trip time allowable, including propagation time and any delay introduced by signal processing at microwave stations. The maximum distance allowed is given by the following formula:

$$\text{Maximum Allowable Distance (miles)} = \frac{(M \times V - D)}{2}$$

M = maximum allowable delay (**ms**)
V = propagation velocity (**miles/ms**)
D = any other processing delays.

5.09 Example: In the case of a land-line system, assume that the maximum allowable delay (**M**) is 1.25 ms to give a safety margin of 0.25 ms. The propagation velocity of signals through wire is 115 **miles/ms**. Assume there are no delays in the system. Thus the formula:

$$\frac{(MV - D)}{2} = \frac{(1.25 \times 115) - 0}{2}$$

= 71.875 or about 70 miles.

6. CONNECTIONS

6.01 All cables except connector C and D cables of carrier shelves are terminated and designated as described in **553-2YY1-215**. Connector C and D cables are terminated as shown in Tables 6-A **and 6-B**.

6.02 Cross-connect wiring to QUA1 emergency transfer unit as shown in Fig. **6-1** and Table 6-R

6.03 **Sets** and consoles are cross-connected to PE shelves as described in **553-2YY1-215**.

6.04 Connections for **500-type** sets and trunks that connect through emergency transfer units are given in Fig. 6-1. A ring ground start button must be provided on these sets, if the trunks are ring ground start,

6.05 RPE local and remote carrier shelves are cross-connected to the **digital** carrier cable pairs as shown in Fig. 6-2 and 6-3.

6.06 RPE maintenance pairs are cross-connected as shown in Fig. 6-4.

Table 6-A
 CABLE C DESIGNATIONS
 (Local and Remote Locations)

NETWORK LOOP	CARD POS	CONN C PIN NO.	LEAD CLR	LEAD DESIGNATION
X	9	1	BL-W	C1
X	9	26	W-BL	C2
X	9	2	o-w	C3
X	9	27	W-O	c4
Y	9	3	G-W	c9
Y	9	28	W-G	C10
Y	9	4	BR-W	C11
Y	9	29	W-BR	C12
	9	5	S-W	OWT
	9	30	w-s	OWR
	9	6	BL-R	FLT
	9	31	R-BL	FLR
	4	8	O-R	FLWA
	4	33	R-O	FLWB
	4		G-R	WFIT
	4		R-G	WFIR
		9 to 25, 34 to 50 (see Notes)		GND
				GND

Note: 1 These leads are grounded in the carrier shelf when cable connectors C and D are installed in jacks C and D. They are cut off at the MDF when cables C and D go directly to an MDF. Carrier problems are less likely to occur when the cables terminate directly on the MDF.

Note 2: When cables C and D go to an intermediate cross-connect terminal, i.e., an SL-1 cross-connect terminal, they are terminated at that terminal and extended to the MDF. These leads are then cut off at the MDF.

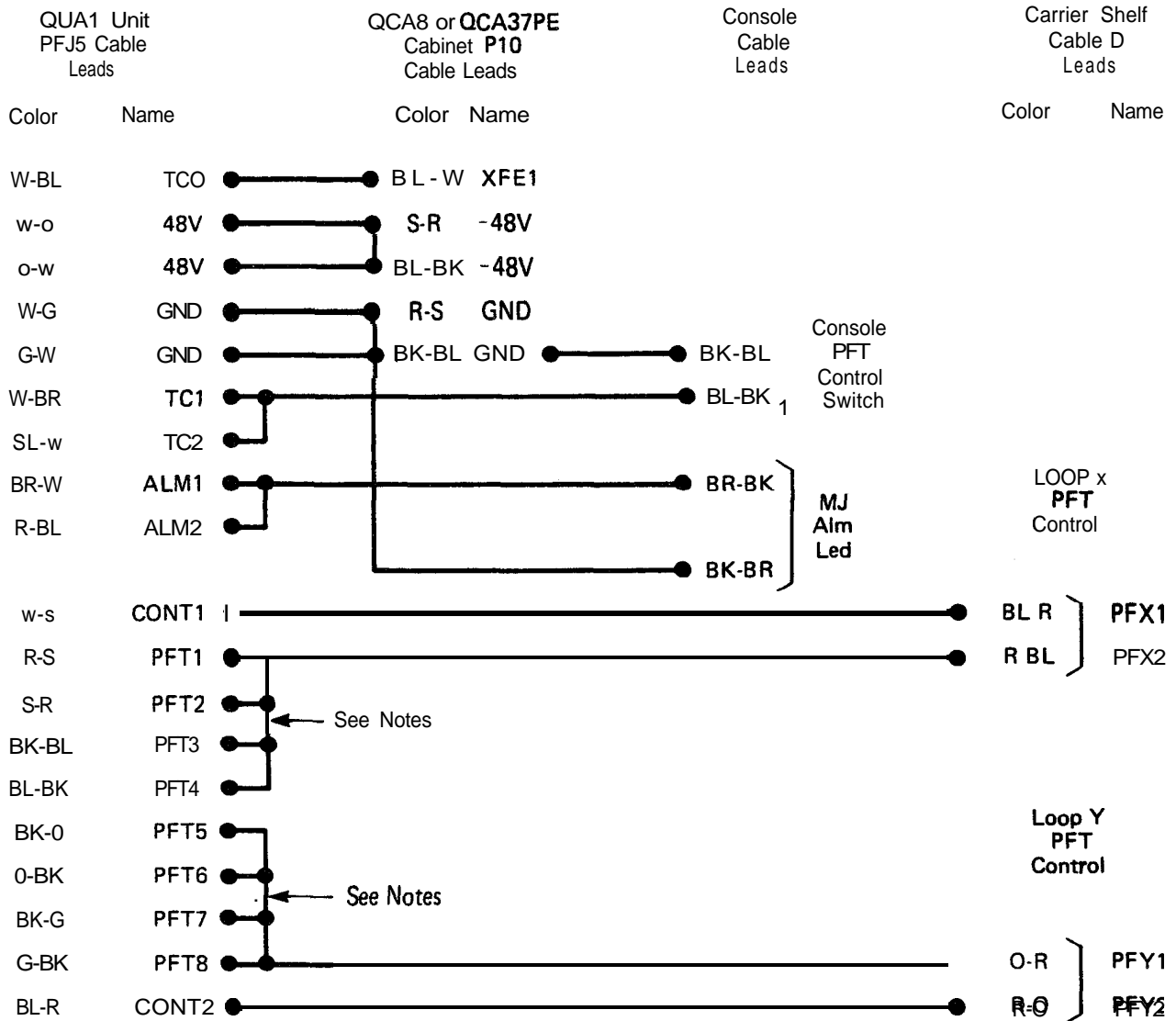
Table 6-B
 CABLE **D** DESIGNATIONS
 (Local and Remote Locations)

NETWORK LOOP	CARD POS	CONN C PIN NO.	LEAD CLR	LEAD DESIGNATION
X	9	1	BL-W	C5
X	9	26	W-BL	C6
X	9	2	O-W	C7
X	9	27	W-O	C8
Y	9	3	G-W	C13
Y	9	28	W-G	C14
Y	9	4	BR-W	C15
Y	9	29	W-BR	C16
	5	5	S-W	GND
	4	30	W-S	GND
	4	6	BI-R	PFX1
	8	31	R-BL	PFX2
	8	7	O-R	PFY1
	9	32	R-O	PFY2
	9	8	G-R	DCST
	9	33	R - G	DSCR
	9	9	BR-R	DETT
	9	34	R-BR	DETR
		10-25		
		35-50	(see Notes)	

Note: 1 These leads are grounded in the carrier shelf when cable connectors C and D are installed in jacks C and D. They are cut off at the MDF when cables C and D go directly to an MFD.

Note 2: When cables C and D go to an intermediate cross-connect terminal, i.e., an SL-1 cross-connect terminal, they are terminated at that terminal and extended to the MDF. These leads are then cut off at the MDF.

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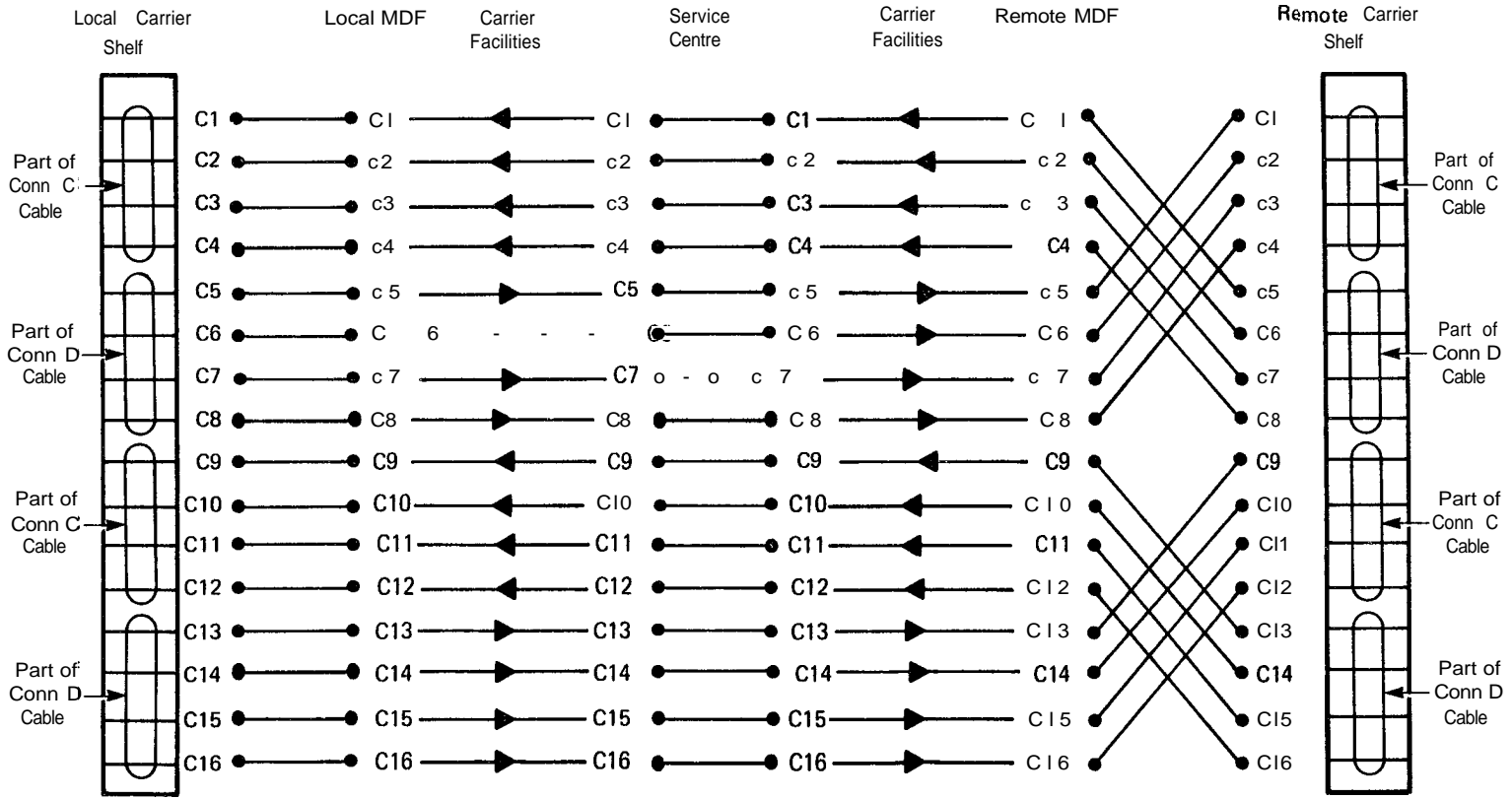
Notes: 1 Typical wiring of 24 PFT stations (12 served and controlled by Loop X, 12 by Loop Y). Each PFT1 through PFT8 lead controls 3 stations, the bridging of the PFT1 to PFT8 leads adds or deletes stations to Loops X or Y. See 553-2YY 1-215 for connections of PFT stations and CO trunks to the PFT control leads.

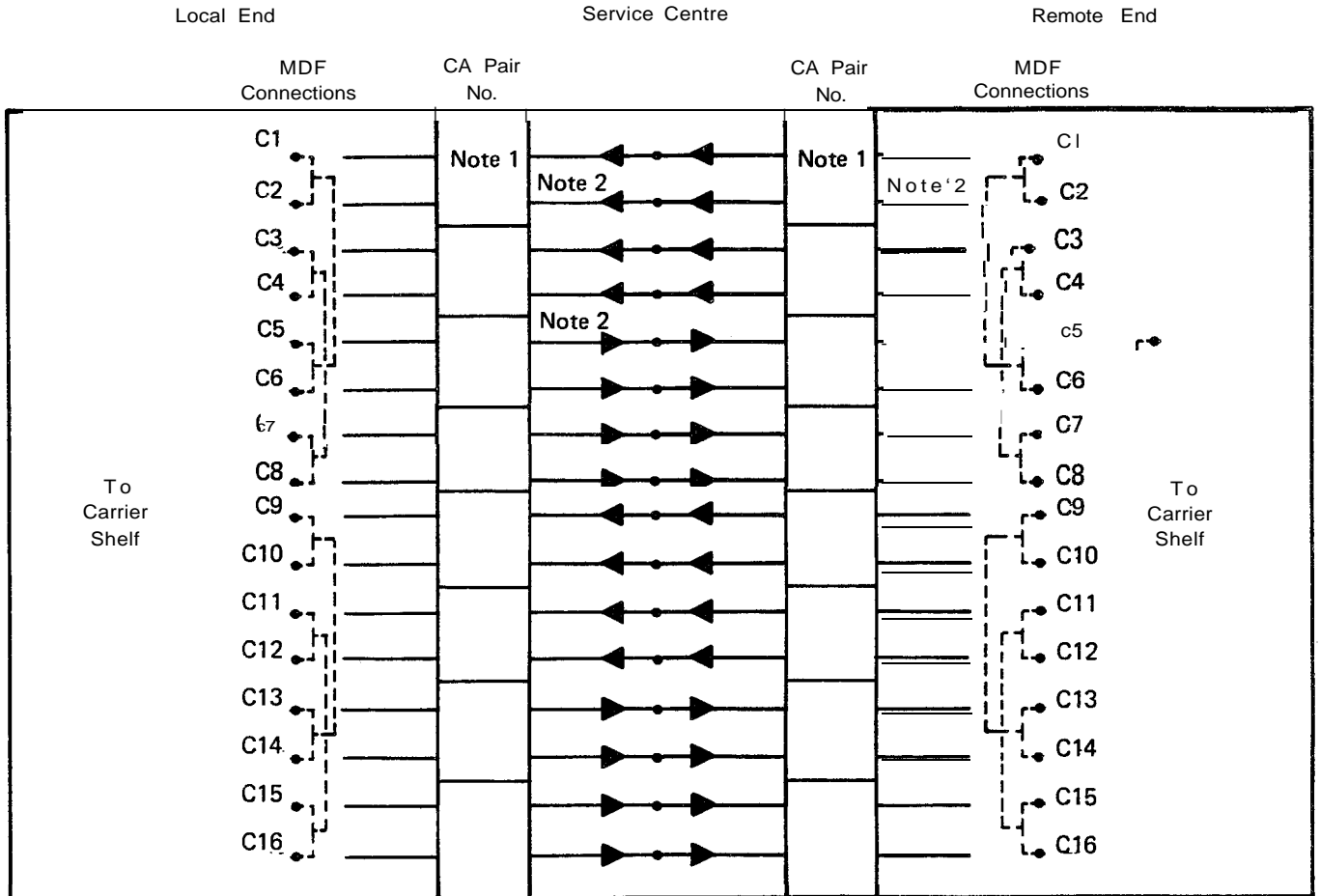
2. PFT stations served by PE shelves of Loop X must not be connected to the same PFT circuits that provide PFT for stations of Loop Y.

● — ● Indicates installer provided wiring

Fig. 6-1
PFT intercabinet and Console Cross-Connections (QUA1 shown)

Fig. 6-2
PFT RPE Local-To-Remote RPE Cross-Connections





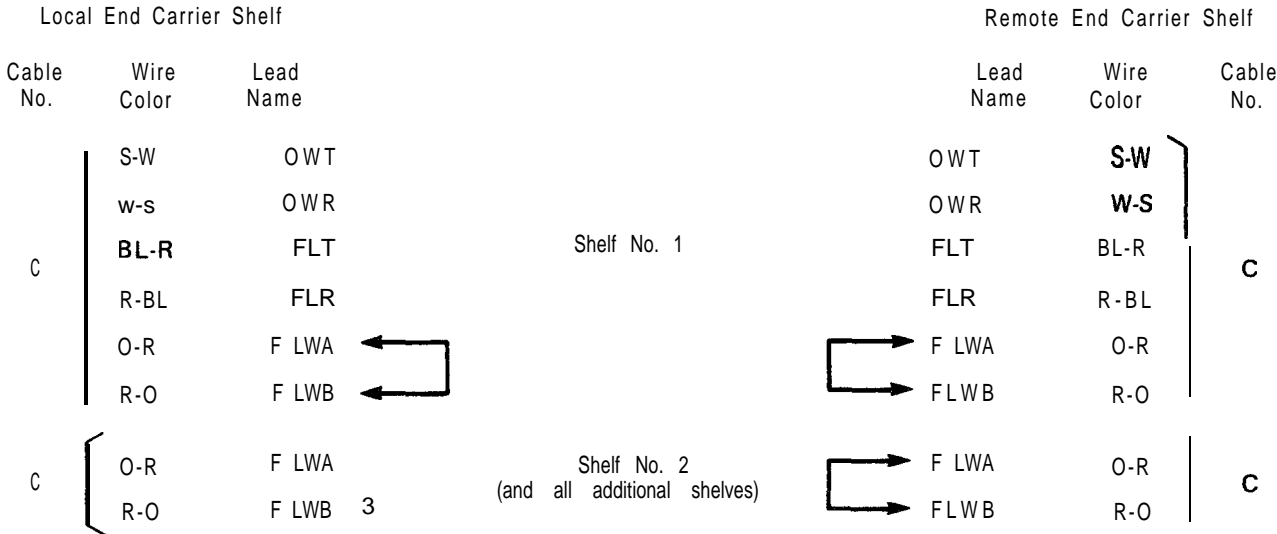
Note 1: Receiving and transmission cable pairs to be entered in these columns. Obtained from Switching Centre personnel and work order.

Note 2:

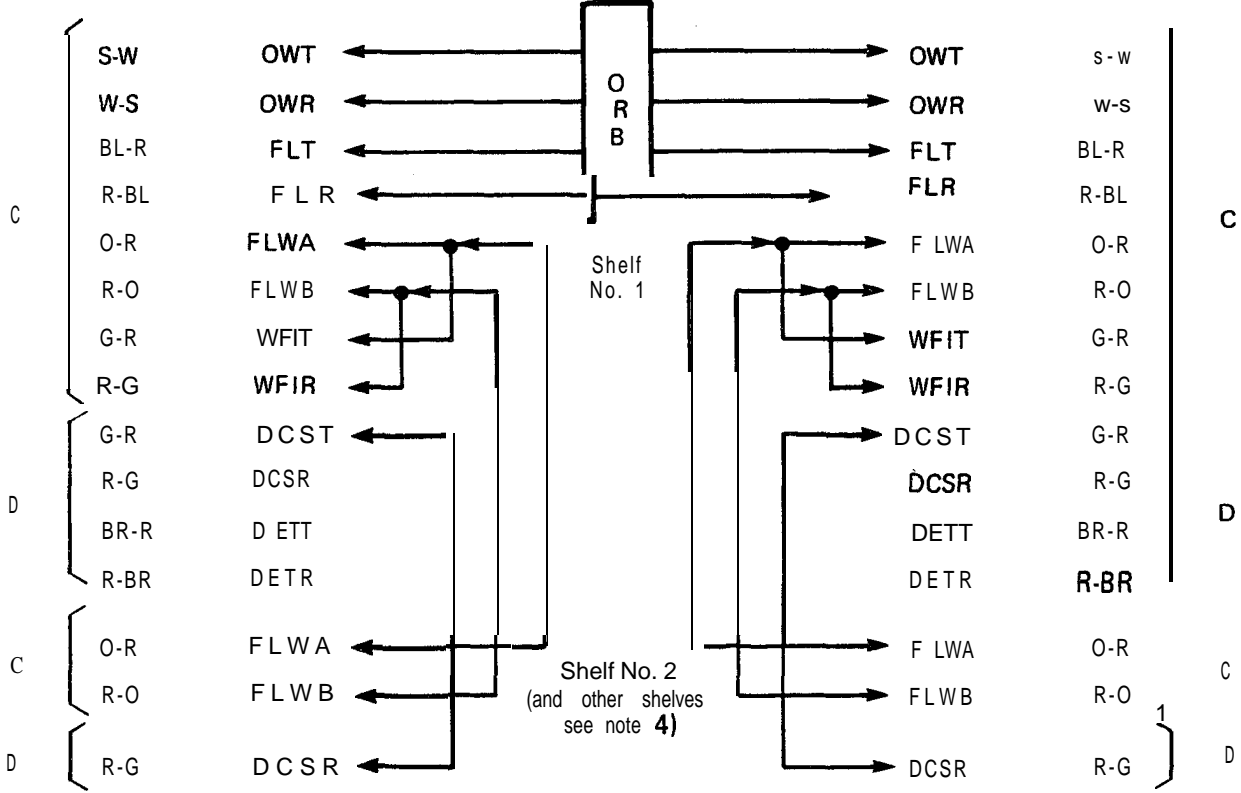
- ➔ Indicates transmission to Service Centre from Local End.
- ➜ Indicates receiving from Service Centre at Local End.
- ➔ indicates **receiving** from Service Centre at Remote End.
- ➜ Indicates transmission to Service Centre from Remote End.

Each set of cable pairs (joined by broken lines) is a separate carrier line (system).

Fig. 6-3
Detail of MDF Cross-Connections (Local and Remote)

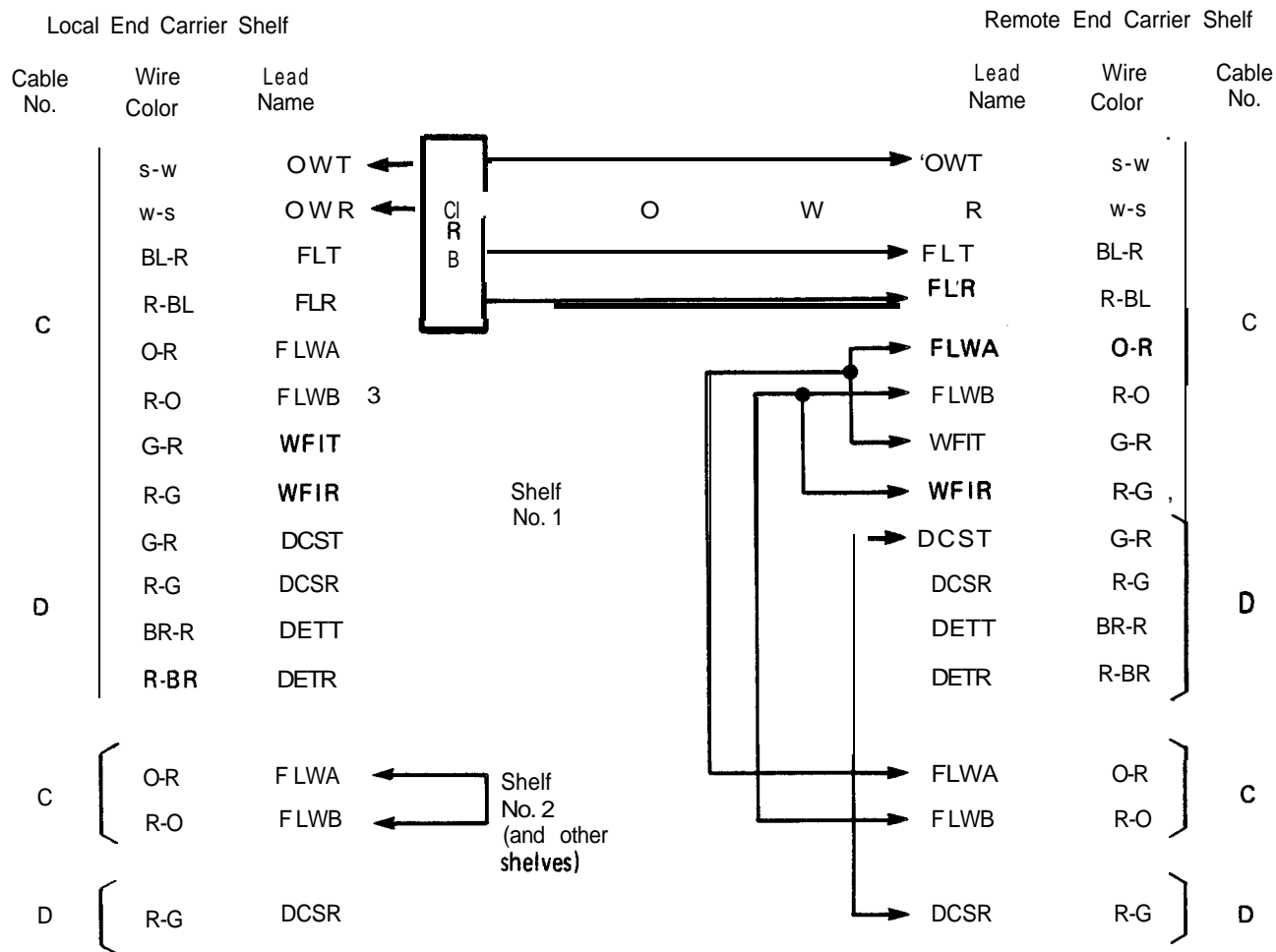


A) RPE SYSTEM NOT EQUIPPED WITH ORB



B) SYSTEM EQUIPPED WITH ORB. ORB NOT LOCATED AT LOCAL OR REMOTE END

Fig. 6-4
Maintenance Leads Cross-Connections



- Notes:
1. Cross-connections are shown when ORB is located at the local end.
 2. If ORB is located at the remote end, the cross-connections shown above for the local end are connected at the remote end and the connections shown for the remote end are connected at the local end.
 3. If there is an ORB at the local and remote ends, cross-connections for both ends are the same as shown above for the local end.
 4. Assuming all the shelves go to the same remote location.

C) SYSTEM EQUIPPED WITH ORB AT LOCAL OR REMOTE OR BOTH LOCATIONS

→ Fig. 6-4 (Continued)
 → Maintenance Leads Cross-Connections

7. TESTING

7.01 The following identifies the testing information for equipment+ associated with RPE. ←

- (a) LD-1 Carrier Equipment. Refer to 368-2101-200.
- (b) Network Loops:
Load the Remote Peripheral Equipment Diagnostic Program 33.
Enter LD 33 on TN
Enter Command LOOP L (**L** is the loop **number**)
Response will be 'OK' if no faults are detected. ←
Enter Command SCAR L to switch primary carriers on loop L
Enter LOOP L again on **TTY**.

Note: Any connection memory or channel faults detected result in the affected channel being disabled. Refer to Program 30 to interpret outputs.

- (c) **500/2500** Sets, SL-1 Sets, Attendant Consoles and Add-on Modules. Refer to **553-2YY1-310**, -305 and -300 respectively or **553-2YY1-315**. ↵
- (d) Emergency Transfer Stations and Trunks. Manually invoke emergency transfer by operating the emergency transfer switch on the faceplate of the QPCM pack. Perform an outgoing and incoming call to each station.

Note: If outgoing trunks are ring ground start, momentarily operate the ground start button on the **500-type** set after lifting the receiver to get dial tone.

7.02 The Remote Peripheral Equipment Diagnostic Program 33 **should** ←
be included in the midnight routines for any system having RPE. ←

TRANSMISSION
QUALITY TEST

7.03 Figure 7-1 shows the monitoring facility built into the **QPC99** Carrier Interface circuit pack. Using these facilities, transmission quality should be tested when the installation is complete.

7.04 There are 4 jacks on the front of the **QPC99**:

- 1 ROUT A and ROUT B are input jacks for channels A and B respectively
- 1 MON A and MON B are output jacks for the monitoring of the regenerated signal. Connecting a test set to the MON jack does not upset an operating RPE system. A test set can be **connected** to a MON jack at any time to test transmission quality.

7.05 Transmission may be checked by closing the MLP switch at the rear end, injecting a test signal at the ROUT jack and monitoring the output at a far end MON jack. This checks the transmission path in one direction. The reverse transmission path should be checked in the same way.

**SPAN LINE FAULT
LOCATING**

7.06 The typical repeatered line maintenance arrangement (Fig. 7-2) includes a series of fault-locating filters and a fault-locating cable pair installed in each span line. One fault locating filter is used for all repeaters installed in a single housing (maximum 25 two-way repeaters in a single housing). A filter with a different audio center frequency is installed in each repeater housing in the span line.

7.07 The fault locating filter is a narrow band selective filter centered at one of 12 audio frequencies. The output of each repeater in the housing is bridged across the input of the respective fault-locating filter. The outputs of all filters in a span are connected to the common fault locating cable pair. This arrangement permits interrogating each repeater in a span from either span terminal using a test signal with an audio frequency component corresponding to the center frequency of the respective fault locating filter.

7.06 The fault locating test is performed using a Pulse Code Modulation (**PCM**) line and repeater test set (**Lair Seigler Sierra 415A** or equivalent), when a span line is producing excessive errors or a failure has occurred, and it is necessary to locate the defective repeater. To accomplish the test, the line is removed from service and the test set is connected to the line at the span terminal. The output of the test set transmit section is connected to the span line while the receive section input is connected to the corresponding fault locating jack.



7.09 A series of test pulses with an audio-frequency component is transmitted down the line. The audio-frequency component is selected in turn to correspond to each of the fault-locating filters in the span. This audio-frequency component appears in the output of each repeater in the span. However, a portion of this signal filters through the appropriate associated fault-locating filter and returns to the test set over the fault locating pair. The amplitude of this test signal is measured on the **dB** meter and is a function of the performance of the repeater under the various test signal conditions. If a repeater has failed completely, no test signal is returned to the test set. By changing the audio-frequency component, each repeater in a span can be tested until the faulty or marginal repeater is located.

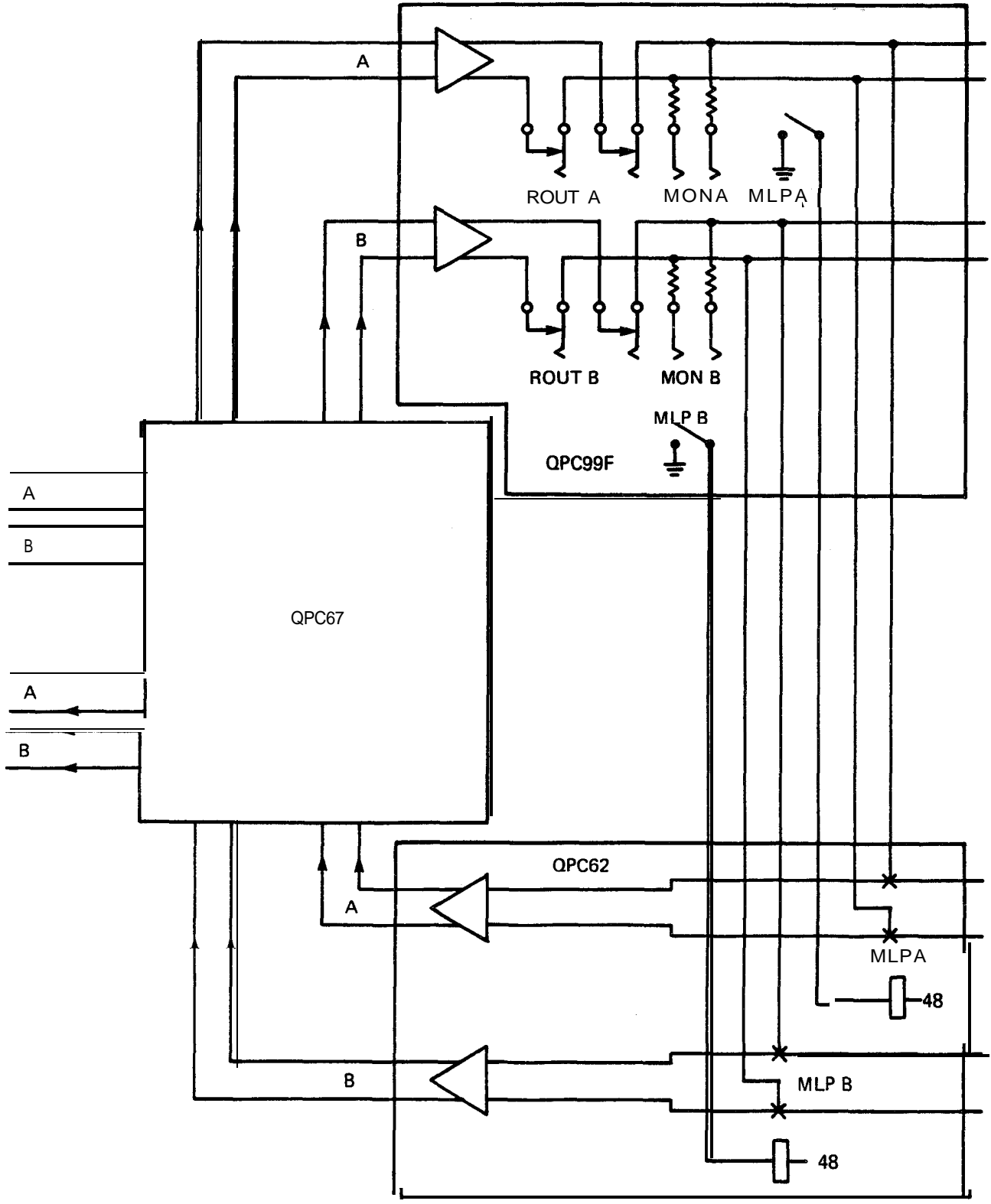
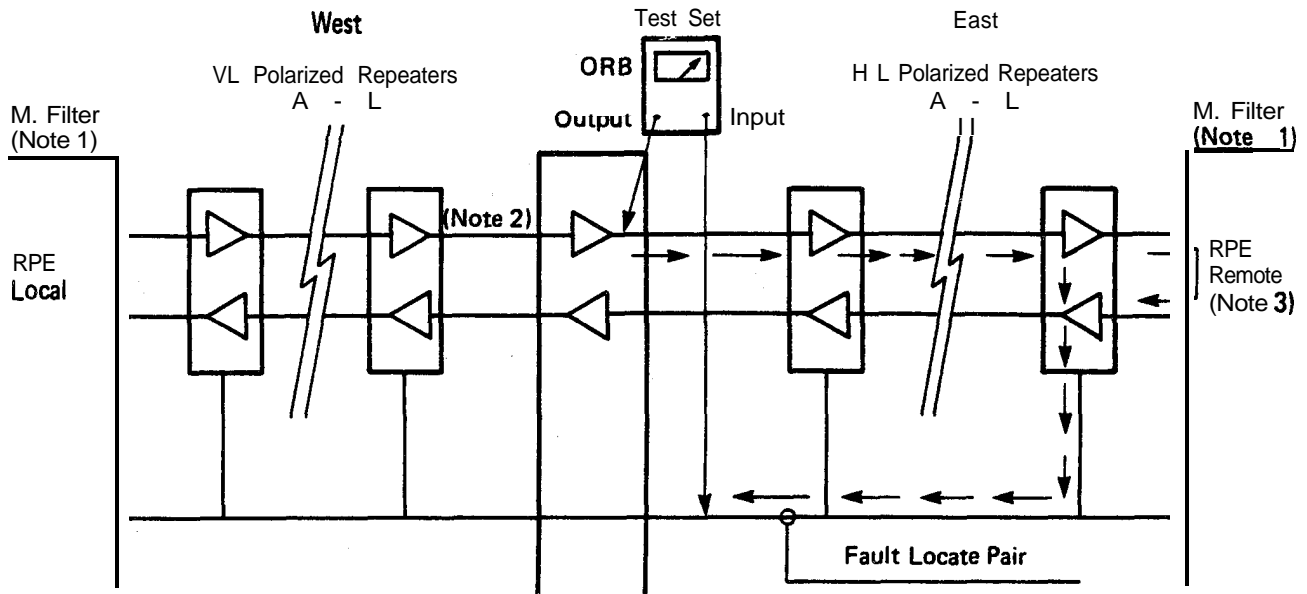


Fig. 7-1
QPC99 Maintenance Jacks and Looping Switches



Note 1: Non-polarized M filter on the **QPC67** pack. There must be no M filters within the repeater span.

Note 2: The previous filter must be good for the signal to continue.

Note 3: **Loopback** conditions, controlled by **QPC99**, occur under the following conditions:

- ┆ DC on Fault Locate Pair and TRIOS present
- **DC** on Fault Locate Pair and excessive Bipolar Violations (**BPV**)
- ┆ TR IOS present

Fig. 7-2
Typical Span Line Fault-Locating Arrangement

SL-1*

BUSINESS COMMUNICATIONS SYSTEM

REMOTE PERIPHERAL EQUIPMENT MAINTENANCE

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1. GENERAL

1.01 Remote Peripheral Equipment (RPE) is a hardware option available for the SL-1 system. RPE is used to extend the serving range of the SL- 1 to a maximum of 70 miles (112 km) from the main SL-1 location.

1.02 A carrier system must be used to link the local and remote RPE locations. This can be provided by Northern Telecom or by the customer. The system should conform to “T1” carrier specifications.

1.03 It must be kept in mind that the SL- 1 RPE and the carrier constitute a unified system. The first stages of the fault-clearing procedure should deal with identifying the location of the fault.

1.04 Before this section is used to diagnose faults and guide the repair of faults, 553-2601-200 should be read to gain familiarity with the SL-1 RPE system and its limits.

This section gives fault-clearing procedures for the SL-1 RPE only. Carrier equipment is referred to, but only in general terms, since the proper documentation for each carrier system should be obtained from the manufacturer.

2. EQUIPMENT DESCRIPTION

2.01 Each RPE system extends a SL-1 network loop to a maximum of 70 miles (112 km). Each loop serves up to a maximum of 4 Peripheral Equipment (PE) shelves.

2.02 A QPC50 network loop pack connects to an RPE carrier shelf at the SL-1 equipment (local end) and the PE shelves connect to an RPE carrier shelf at the remote location (see Fig. 2-1). The two carrier shelves interface with a 1.544 Mb/s multiplexed digital carrier system (such as LD-1) via four cable pairs for signaling and transmission. The two carrier shelves (one local, one remote) will serve two remote network loops.

2.03 An ORB is required in an RPE system if the remote PE equipment is more than 2500 feet (762 m) from the local SL-1 carrier shelf.

An ORB provides the following:

- span line monitoring
- error monitoring
- fault-locate system access
- l order-wire termination with DDD access
- line looping.

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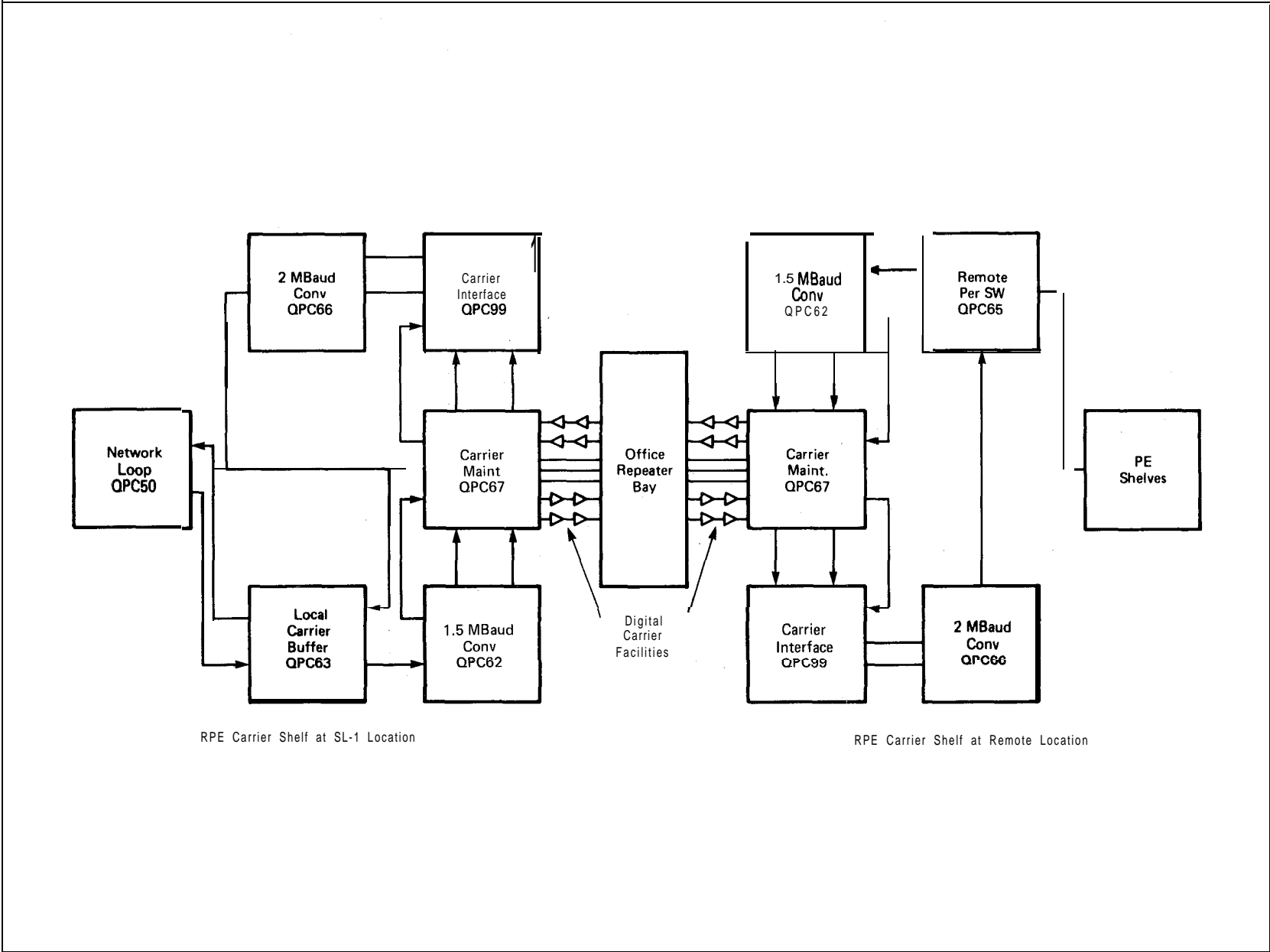


Fig. 2-1 - RPE Block Diagram

3. FAULT DETECTION

OVERLAY PROGRAMS

3.01 Most faults are detected automatically by diagnostic routines that are provided on the system tape.

3.02 Program 45 (Background Signaling Test) is run continuously to test the signaling paths to all PE shelves.

3.03 Program 33 (RPE Diagnostic Program) should be included in the midnight routines, (see 553-2001-220 for procedures) which automatically tests RPE systems every 24 hours. The program may also be loaded manually to test an RPE system.

3.04 Program 62 (RPE Local End Diagnostic) may be loaded manually to test the local equipment associated with an RPE system.

3.05 Programs 45 and 33 may disable parts of RPE systems (network loops) that fail during a test. Program 45 may light the LED on PE shelf packs. Faults are also indicated by an output on a Teletypewriter (TTY) or by a code on the maintenance display on the QPC41 pack located on the CPU common equipment shelf.

3.06 The meaning of the mnemonics output on the TTY is determined by referring to 553-2001-505. The first three characters of the output identify the program and the other characters identify the meaning of the output."

3.07 The suspected faulty pack associated with the maintenance display code is also given in table form in 553-2001-205.

ALARM INDICATIONS

3.08 Local and remote error monitors in the QPC99 indicate when:

- the error rate at one of the two carriers exceeds one in 10^4 bits, or
- the carrier loses framing.

When a remote error monitor detects an error it sends a message to the central control. If the local monitor is in the error condition, or receives an error message from the remote monitor, the background signaling diagnostic takes the following action.

1 minor alarm lamp at the locally served attendant console(s) is steadily lit.

1 time slots served by the faulty line are busied out

- diagnostic messages at all maintenance terminals is printed out.

3.09 Lamps are provided on circuit packs QPC62 and QPC99 to indicate out of frame and/or high error rate conditions. If the errors are in the carrier line going from the remote equipment to the ORB, an alarm is sounded at the ORB.

Errors in peripheral equipment at the remote location raise the same error and alarm indications as local peripheral equipment.

4. FAULT CLEARING

4.01 Fault clearing begins with the flowchart in this section. The steps in the flowchart are designed to isolate the fault with the use of programs 33 and 62 and then to replace the defective apparatus.

4.02 *Precautions.* Circuit packs in an RPE network loop should not be changed without disabling the loop according to the following procedure:

- (1) load overlay 33 (RPD),
- (2) issue DISL L command (see 553-2001-505 for more detail),
- (3) change circuit pack,
- (4) issue ENLL L command.

If a pack has been changed while the loop was enabled, the loop should be re-enabled in two steps:

- (1) disable the loop (DISL L),
- (2) enable the loop (ENL L).

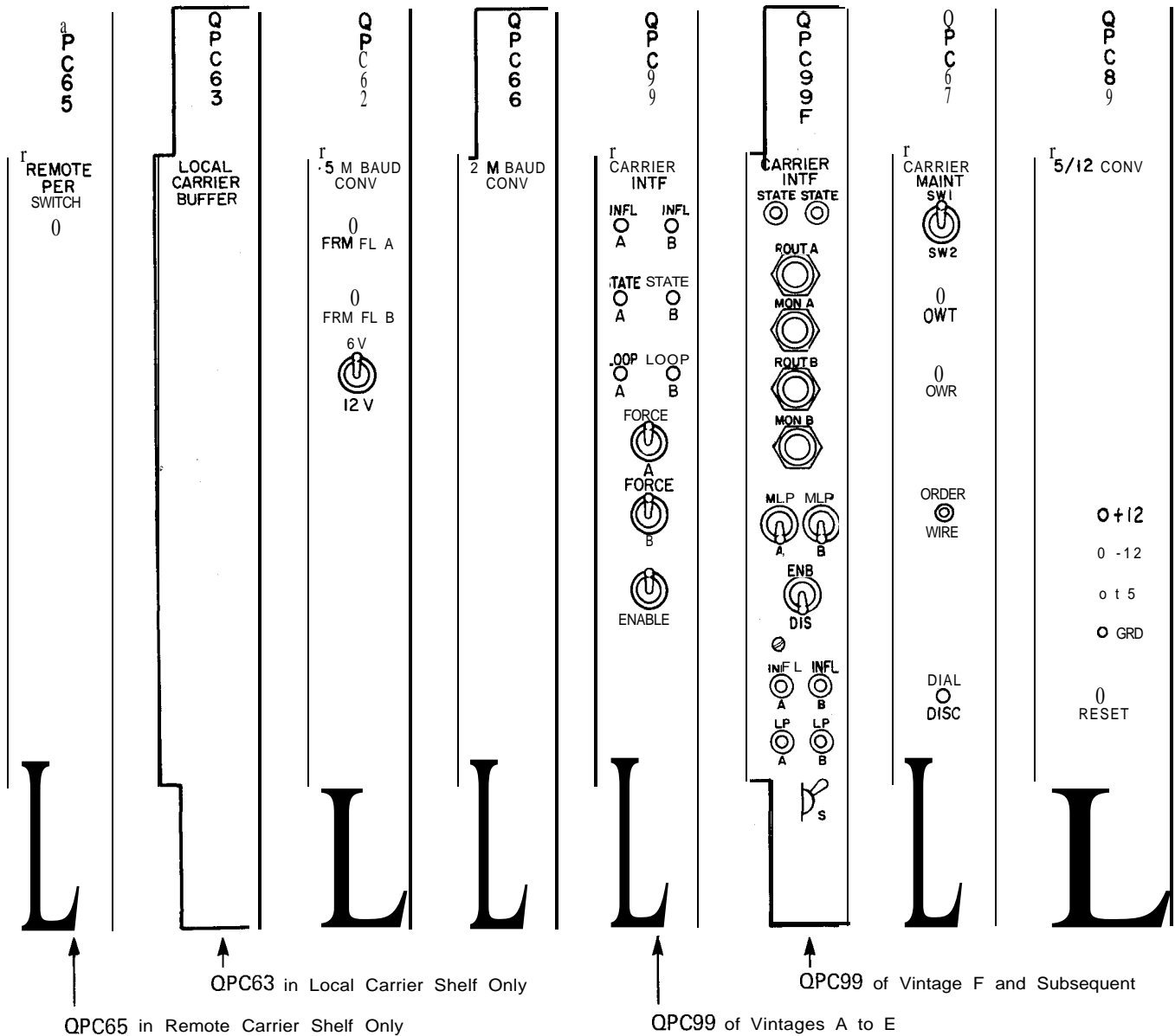
Other precautions are listed in 553-2001-505 and should be followed while replacing faulty apparatus.

4.03 *Option Switches.* 553-2601-200 gives the option switch settings on the QPC62 and QPC99 circuit packs. When replacing these packs, the option switches on the replacement pack must be set accordingly.

4.04 *Power.* Carrier shelves are powered by a QBL 14 power distribution box (see 553-2601-200 for connections and wiring). If the power fault lies elsewhere, refer to the fault-clearing procedures for the power supplies given in 553- 2YY1-520.

4.05 *Faceplate Layout.* Figure 4-1 depicts the faceplates of circuit packs located on the local and remote carrier shelves. Table 4-A gives the functions of the switches and the meanings of the LEDs on each faceplate.

4.06 *Disabling Loop for Testing.* An RPE loop uses two carrier links. If a fault can be isolated to one carrier link, the suspected link should be disabled. In this case the traffic handled by a single link is 360 ccs. Fault clearing can then be done with unnoticeable effects for the end user. If a spare compatible carrier link is available it can be switched into the system until the fault is cleared.

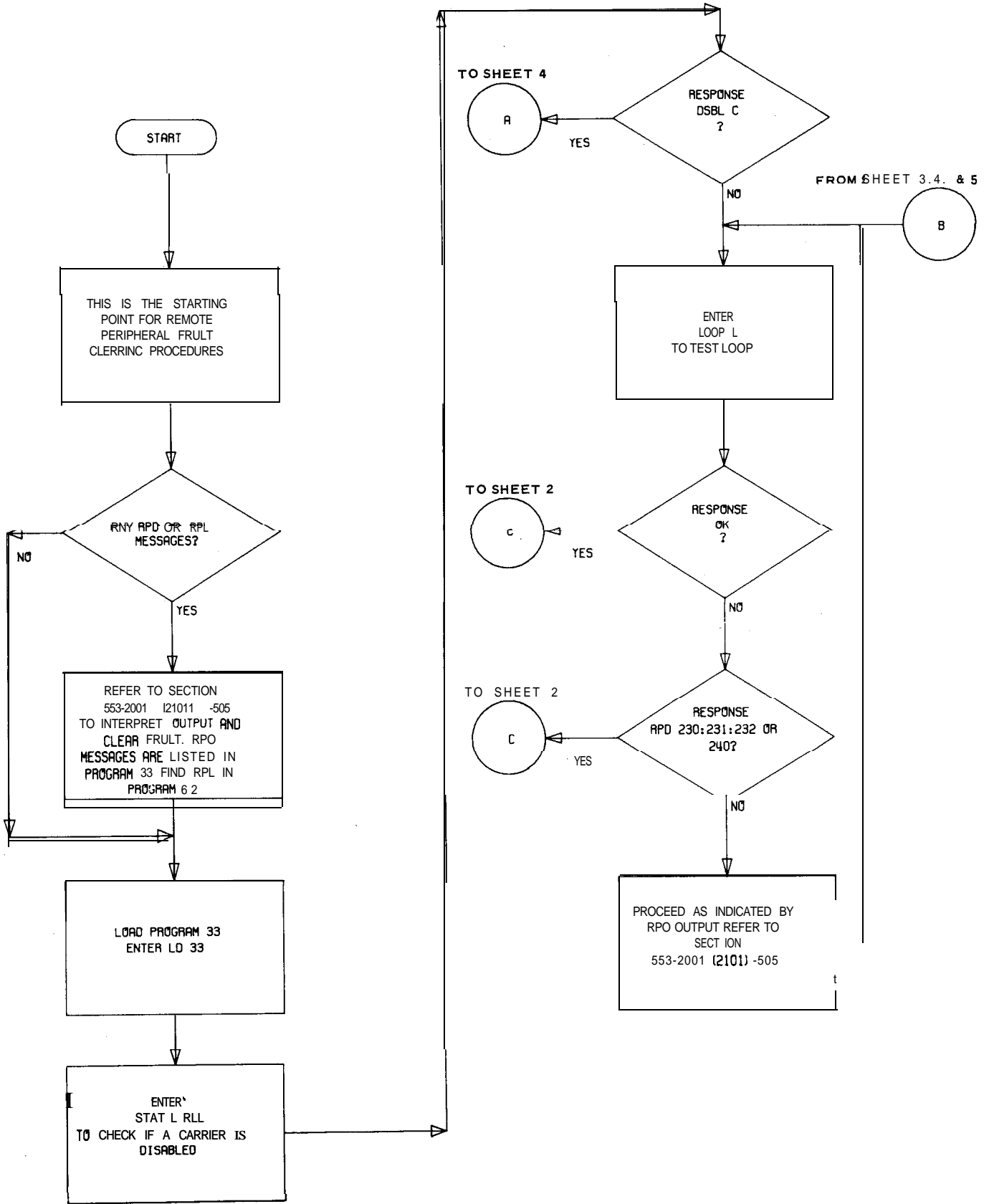


Note : Only one QPC99 is present in the shelf.

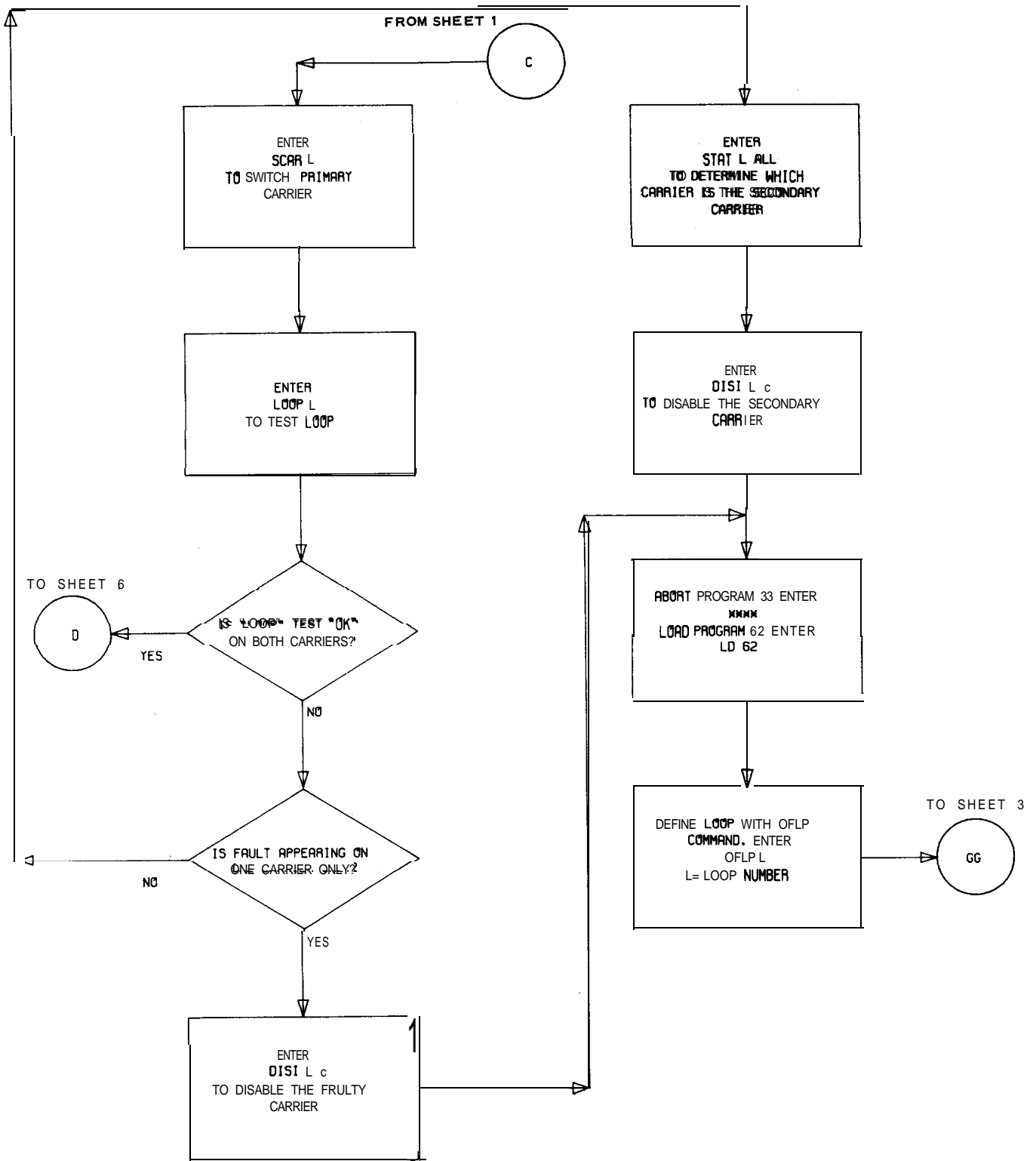
Fig. 4-1 - Carrier Shelf Circuit Pack Faceplates

TABLE 4-A
CIRCUIT PACK LED

Circuit Pack No.								Comments
QPC62		QPC99						
LED Designation								
FRM FL A	FRM FL B	INFL A	INFL B	STATE A	STATE B	LPA	LPB	
Status of LED ● = Lit ○ = Extinguished								
○	○	○	○	●	○	○	○	System is normal and operating on STATE A. If unable to switch to STATE B, outgoing Carrier 0 has failed.
○	○	○	○	○	●	○	○	System is normal and operating on STATE B. If unable to switch to STATE A, outgoing Carrier 1 has failed.
●	○	●	○	○	●	○	○	Incoming Carrier 1 has failed.
○	●	○	●	●	○	○	○	Incoming Carrier 0 has failed.
●	●	●	●	○	○	○	○	Both Incoming Carriers have failed.
○	○	○	○	○	○	○	○	Both incoming Carriers have failed.
●	○	●	○	○	●	●	○	Carrier looping test in progress on Carrier 1.
○	●	○	●	●	○	○	●	Carrier looping test in progress on Carrier 0.
○	○	●	●	○	○	○	○	Loop is manually disabled, disable switch in disable position.

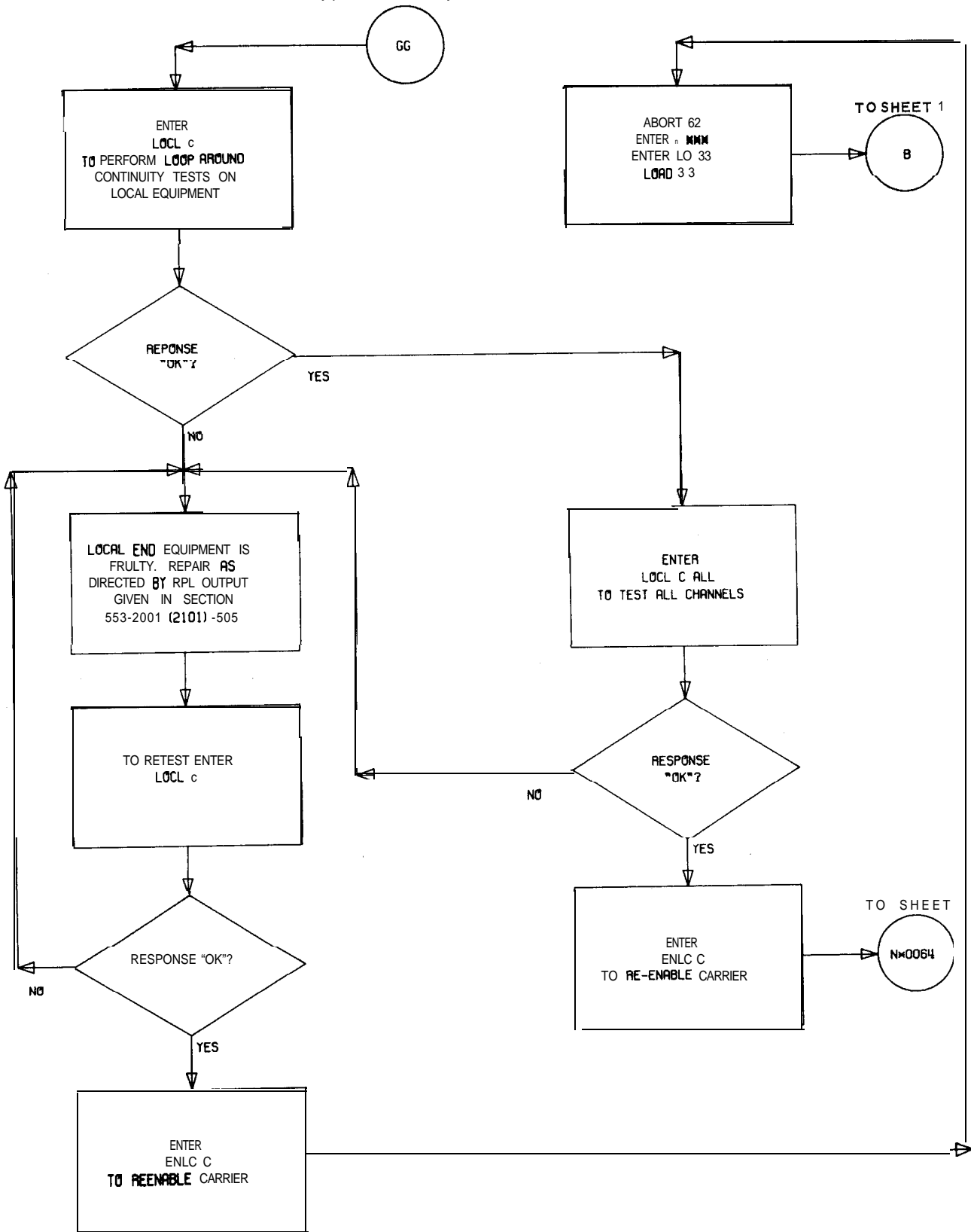


Flowchart 4-1 - RPE Fault-Clearing Procedures

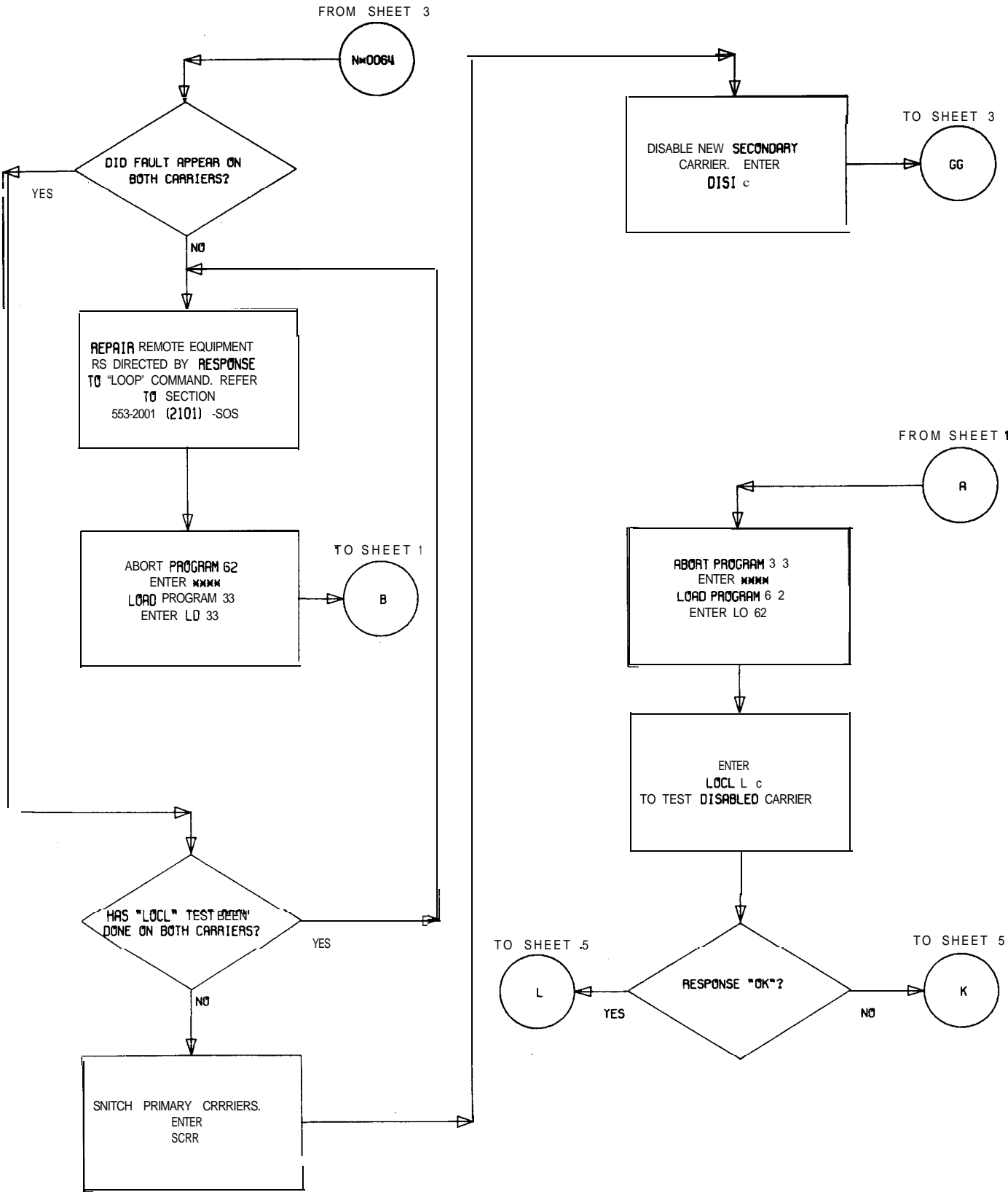


Flowchart 4-1 Continued - RPE Fault-Clearing Procedures

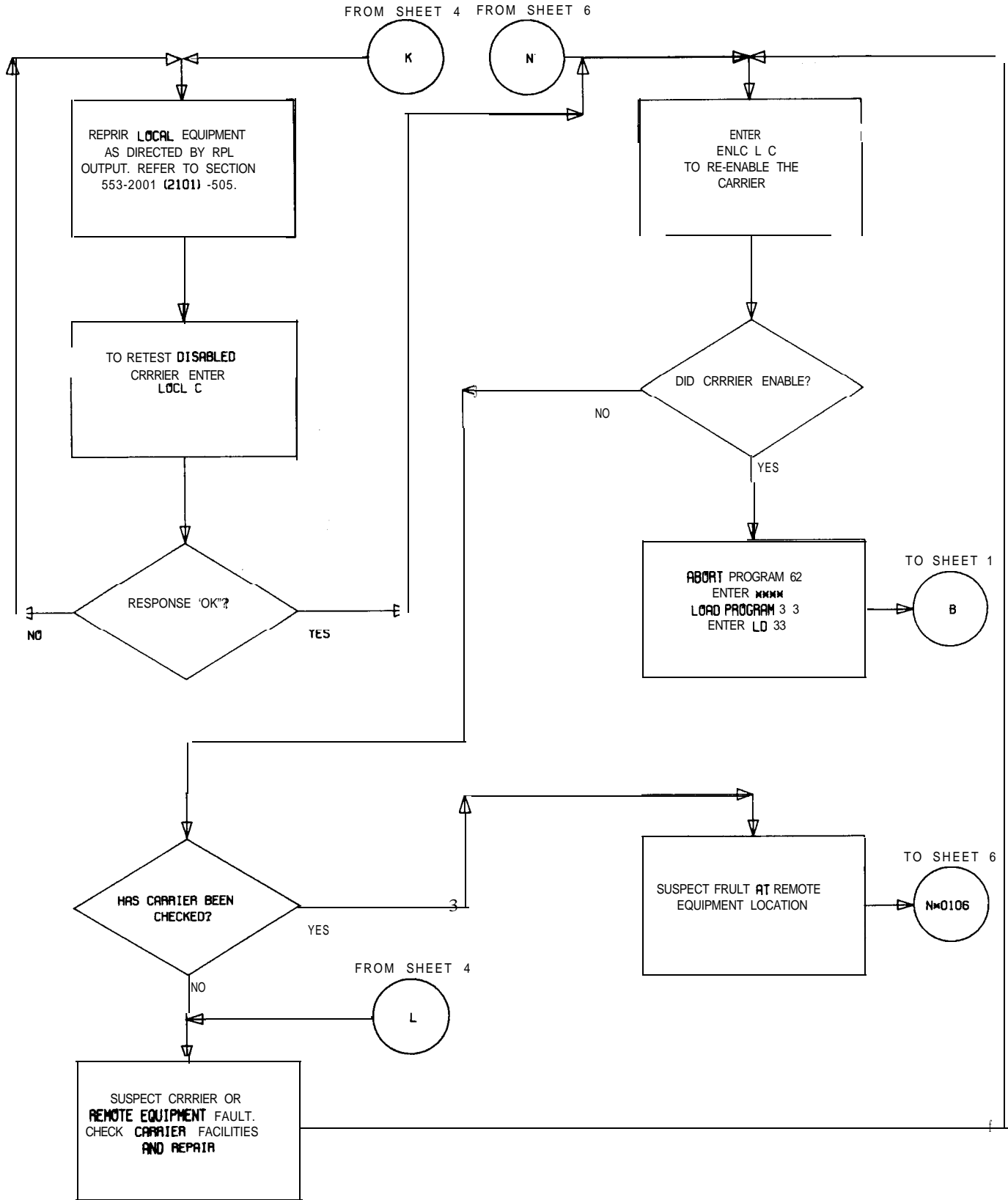
FROM SHEETS 2, & 4



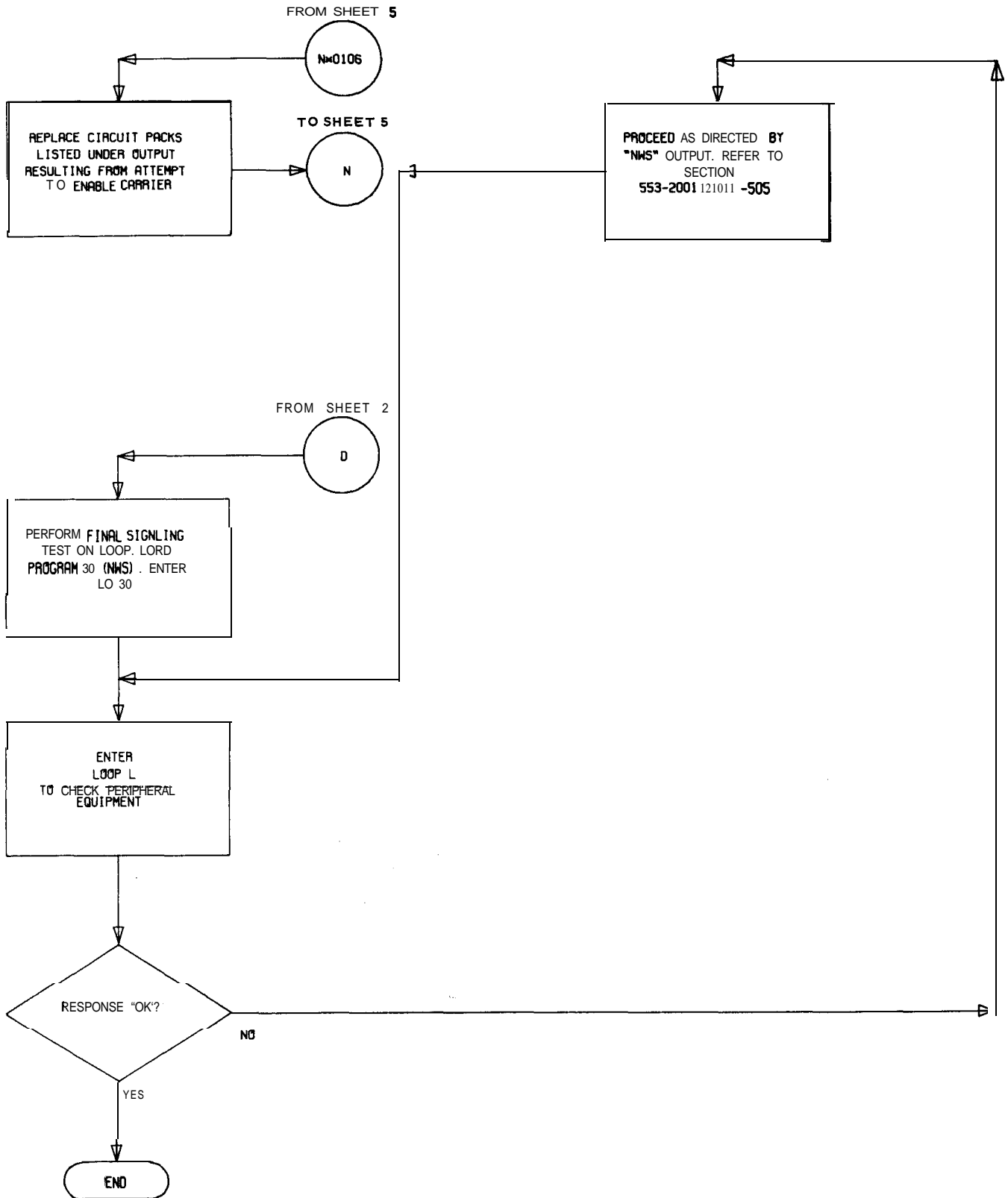
Flowchart 4-1 Continued — RPE Fault-Clearing Procedures



Flowchart 4-1 Continued - RPE Fault-Clearing Procedures



Flowchart 4-1 Continued - RPE Fault-Clearing Procedures



Flowchart 4-1 Continued — RPE Fault-Clearing Procedures

5. CARRIER FAULT CLEARING

RADIO CARRIER SYSTEM

5.01 The considerations for clearing faults in land carrier systems also apply to radio links. The radio link should meet T1 interfacing specifications and any cable connections between the radio and the RPE should meet the same requirements as for ground links.

LAND-LINE CARRIER SYSTEMS

5.02 Due to the number of carrier systems available, this section can only give general ideas for clearing faults in carrier systems. Clearing carrier faults requires:

- | a general understanding of carrier systems
- oscilloscope
- carrier line test set
- complete documentation for the carrier system.

Figure 5- 1 shows a simplified organizational diagram of an RPE carrier system. The system can be partitioned at any of several points to determine whether any component (ORB, line repeater, etc.) is causing the fault. To assist in this, the SL-1 RPE system has a facility to inject test patterns at the ROUT jacks in the QPC99 (Fig. 5-2).

LOOPING OPTIONS

5.03 An RPE system has several looping options available to aid fault location. These options are illustrated in (Fig. 5-1).

RPE Looping Options

- | The system can be tested by looping the signal at the peripheral buffer (D). Load overlay 32 and refer to 553-2001-505 for a list of available commands.

COMMON CAUSES OF FAULTS

5.04 Possible causes of carrier faults can included:

- | **Traffic and Pattern Changes.** Each digital repeater regenerates the clock from the incoming signal. When the loop is idle, a high density of 1s is put into the line, making clock recovery easier. Under traffic conditions the signal density decreases, making clock recovery more difficult. If faults occur when traffic is increased, there may be one or more marginal repeaters in the carrier system, or the clock recovery circuit at one or both RPE ends may be mistuned.
- | Shorted or partially grounded cable pair.
- | The output from the local transmitter can be looped back into the local receiver(A). Load overlay 62 and see 553-2001-505 for the appropriate commands.
- | The carrier can be looped manually and tested from the ORB (B+C), depending on the type of carrier system used.

Carrier Looping Options

- loop when error rate exceeds one in 10^4
- loop when DC is detected on the FLP and error rate exceeds one in 10^4
- loop when the manual loop back switch (MLP) on the QPC99 is closed.

The manual loop back (MLP) switches on the QPC99 allow the signal from the QPC99 to be put directly back into the carrier line and monitored at the far end using the monitor (MON) jacks on the QPC99 at the far end.

These options are switch-selectable on the QPC99. The use of fault-locating filters is outlined in the Northern Telecom Practices for the LD-1 carrier system. Whichever carrier system is used, documentation for it should be obtained and used to assist in the fault clearing process.

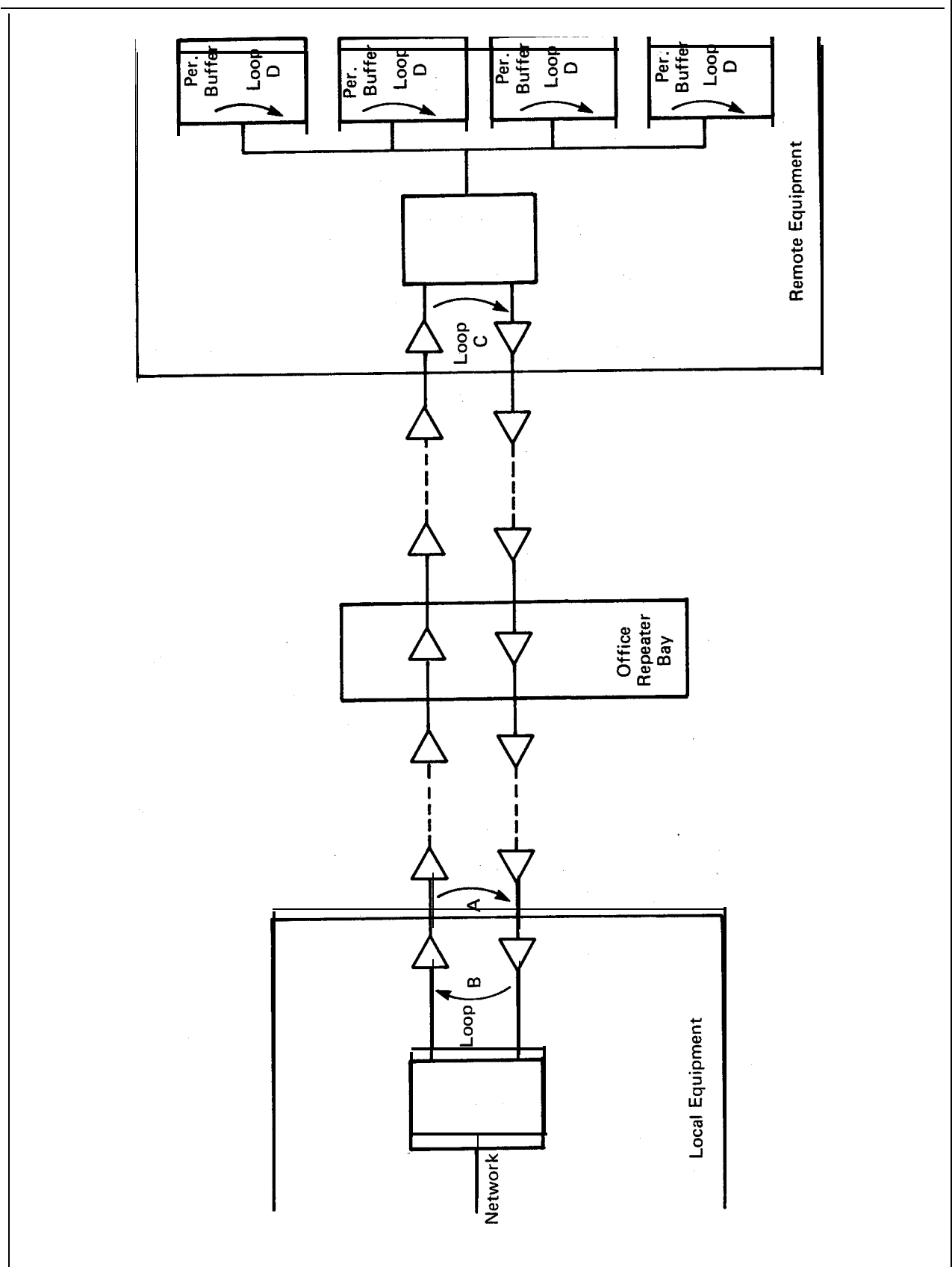


Fig. 5-1 — Carrier Looping Options

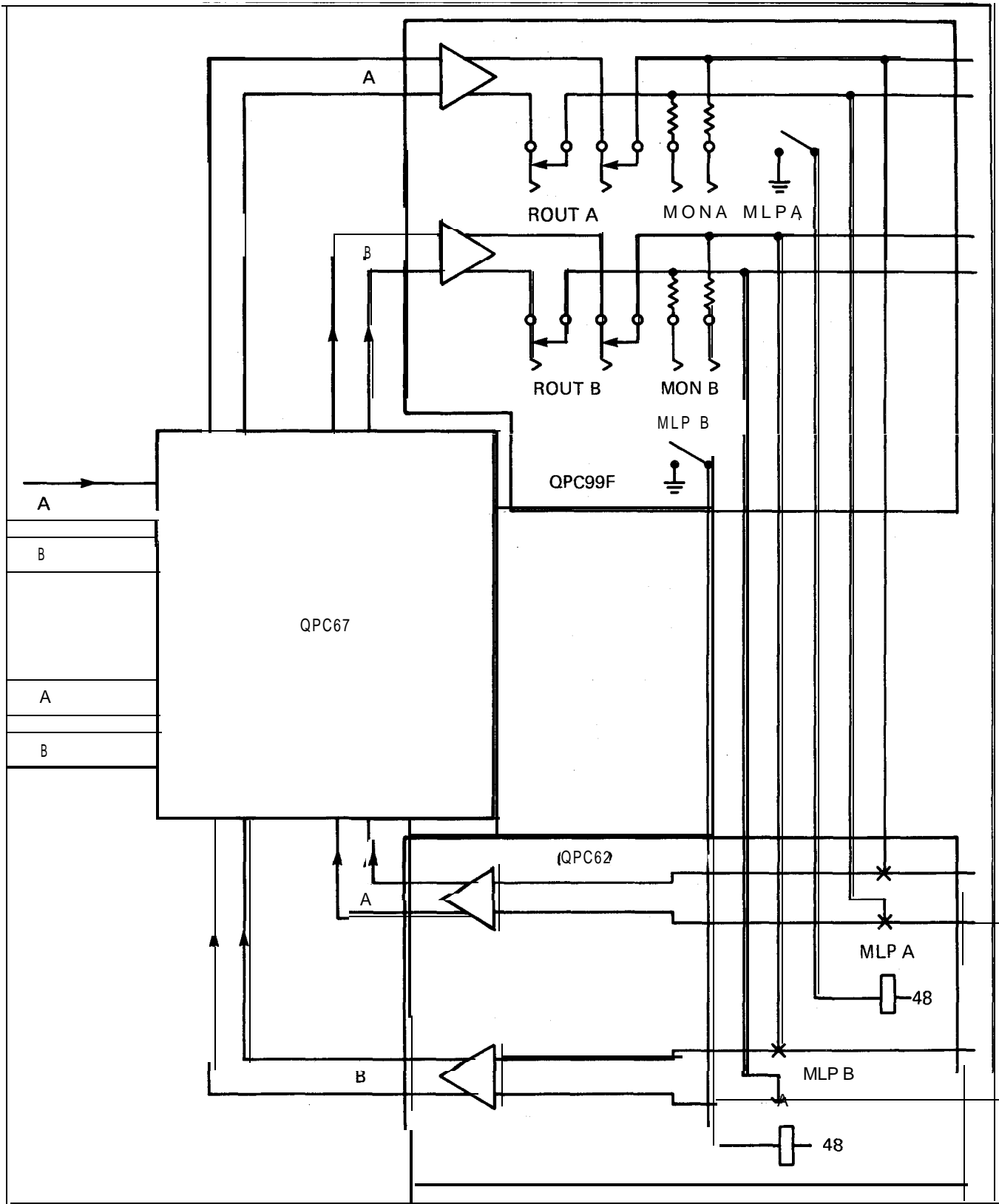


Fig. 5-2 - QPC99F Maintenance Jacks and Looping Switches

5.05 Problems may be encountered at the repeater site itself, such as:

- Faulty Repeater.
- Faulty cable pair at the input to a regenerative repeater, wherein signal errors are arriving from the line.
- Faulty cable pair at the repeater output toward the next repeater location, causing reflections and poor impedance characteristics which result in errors at the repeater.
- Faulty simplex power loop, causing degraded repeater performance or complete failure.
- Faulty wiring in the repeater housing, sometimes encountered during pre-service testing.

TESTING THE CARRIER SYSTEM

5.06 The performance of a PCM system can be observed either by “by eye” using an oscilloscope or by using a test set to measure error density.

5.07 *Error Measurement.* For voice traffic, a PCM system with 10^{-6} (one error in a million bits) performance is generally considered excellent. With 10^{-5} , clicks may be heard, and 10^{-4} performance results in pops, clicks and noise. At 10^{-3} most PCM systems go into alarm **conditon**. (At this point there is one error in only 1000 bits).

5.08 For data traffic, any system performing at less than the 10^{-6} level causes problems for customers.

- 1 **Crosstalk.** Every cable has its limits on the number of PCM systems that can be safely operated in the same sheath without intersystem interference. Crosstalk causes

errors because of the regenerator’s inability to decide if an induced signal constitutes a “pulse” or not. This shows up as “pops”, noise, and data transmission errors. The most common crosstalk path is within the repeater housing and within the cable near the repeater itself due to the large difference (**32 dB** or more) between the outgoing and incoming signal.

- unbalanced cable pairs
- load coils, bridge taps, or building-out capacitors not removed
- moisture in cable
- defective lightning protection devices
- crossed cable pairs
- open or very high resistance cable pairs or splices (continuity tests may help in locating this).
- unbalanced cable pairs
- load coils, bridge taps, or building-out capacitors not removed
- moisture in cable
- defective lightning protection devices
- crossed cable pairs
- open or very high resistance cable pairs or splices (continuity tests may help in locating this).

When inspecting the waveform on the oscilloscope at various points in the carrier span, the requirements listed in 553-2601-200, Carrier Specifications, should be kept in mind, since any deviation from the limits listed there causes errors or contributes to them.

BUSINESS COMMUNICATIONS SYSTEM
SL-1*

AUTOMATIC NUMBER IDENTIFICATION (ANI) FEATURE
DESCRIPTION, OPERATION AND MAINTENANCE

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1. GENERAL

1.01 Certain PBX applications involve automatic message accounting. In these cases, it is necessary to be able to send information automatically which identifies the parties involved in a call to a Central Office (CO) long distance facility. The Automatic Number Identification (**ANI**) feature of SL-1 provides this capability.

1.02 An SL- 1 system with the AN1 feature can identify the stations involved in an outgoing toll call and automatically send this information by Multifrequency (MF) signaling over Central Automatic Message Accounting (**CAMA**) trunks to

toll-switching **CAMA**, Traffic Operator Position System (TOPS), or **Traffic** Service Position System (TSPS) offices. The called number information is sent first, followed by the calling number information.

1.03 This practice outlines the functions, ←
interconnections, characteristics, operation, and maintenance of the Automatic Number Identification (**ANI**) feature of the **SL-1** Business Communications System. The information is intended to be used as a guide when designing or engineering the connection of customer-provided apparatus to the trunk circuit.

2. DESCRIPTION

SIGNALING

2.01 The signaling method used to send AN1 information to the CO can be E&M, DX, or loop signaling. The SL-1 AN1 supports three basic methods: Bell, NT400, and NT500.

- (1) The Bell Method can be used to interface SL-1 to:
 - Bell system TOPS, **TSPS**, or **CAMA** offices,
 - Strowger Automatic Toll Ticketing (**SATT**) systems types 57, 59, 62, and 70A; these systems can be made to accept 1+ and 0+ calls from **SL-1** using MF pulsing by means of customer-provided adapter circuitry,
 - Stromberg **Carlson** Ticketing Systems.
- (2) The NT400 method (modes A and B) provides interface to Northern Telecom NT400 ticketing system. Mode A repeats the toll access code (0 or 1) in the called number, whereas mode B does not.
- (3) The NT500 method (modes A, B, C) provides interface to Northern Telecom NT500 ticketing systems. Mode A is used with central offices that use MF outpulsing, combined trunk groups, and repeats the access code (0 or 1) in the called number format. Mode B does not repeat the access code. Mode C is used in offices with MF outpulsing and trunk groups dedicated only to 1+ or 0+ calls.

2.02 The difference between the Bell method and the **NT400/500** method is in the supervisory signals and in the formatting of called and calling number information. The differences between the NT400 and NT500 method are also in the formatting. Tables 2-A, 2-B, and 2-C summarize the possible combinations of trunk types and AN1 methods.

2.03 The MF sender enables the SL-1 to **output** pulse automatically up to 16 digits (including starting and ending digits, called KP and ST respectively) independently in each of the 30 permissible time slots of a network loop.

2.04 The type of trunk packs required depends on the type of signaling required.

- For **E&M/DX** signaling, the QPC71 E&M/**DX/paging** trunk circuit pack is used. It does not have to be modified for use with the AN1 feature. Refer to 553-2001-187 for a complete description of this trunk circuit pack.
- For loop signaling, the QPC72 loop signaling trunk circuit pack is used. The appropriate switch settings are shown in Fig. 2-4. Connections are the same as those described in 553-2001-186.

2.05 These two types of trunks provide compatibility with the signaling and supervision requirements of **CAMA** and **CCSA** trunks, and provide a path for the eventual analogue transmission of the MF tones, and for speech transmission.

CALLING AND CALLED NUMBER INFORMATION

2.06 The called number information always includes the number dialed (typically 7 or 10 digits). It may also include the toll access code (typically 0 or 1). If MF sending is used, additional control signals are sent such as KP (preparatory for digits), or ST (end of pulsing).

2.07 The calling number information is always sent in MF. It consists of a calling number (always 7 digits), the preparatory and end-of-pulsing signals, and other auxiliary signals; i.e., information digit with the Bell method, and class mark and category digits with the NT methods.

2.08 Each customer in the SL-1 system is assigned an AN1 Listed Directory Number (**ANI LDN**). This number is used to identify the customer to the toll office. Depending on the numbering plan, the AN1 LDN can be a 3-, 4-, or 5-digit number. The calling number for AN1 is obtained by combining the AN1 LDN with one of the following:

- (a) *PBX*: Directory (extension) number of the set.
- (b) *SL-1 Set*: Primary DN of the set.
- (c) *Attendant*: AN1 attendant number, specified on a 'per customer' basis.
- (d) *Tie Trunk*: AN1 trunk number, specified on a 'per trunk group' basis. See 553-200 1-220 for information on how to assign these numbers in software.

TABLE 2-A
CALLED AND CALLING NUMBER INFORMATION FORMAT
BELL METHOD

A. DP Sending of Called Numbers			
CALL TYPE	CALLED NUMBER	CALLING NUMBER	
		REGULAR TRUNK GROUP	SUPER TRUNK GROUP
0	seizure — no digits	KP+ID+7D+STP	KP+ID+ST3P
0+7/10D	7/10D	KP+ID+7D+STP	KP+ID+7D+STP
1+7/10D	7/10D	KP+ID+7D+ST	KP+ID+7D+ST2P
01 1+CC+NN	11+CC+NN	KP+ID+7D+ST	KP+ID+7D+ST2P
01+CC+NN	1+CC+NN	KP+ID+7D+STP	KP+ID+7D+ST3P
010	10	KP+ID+7D+STP	KP+ID+7D+ST3P
B. MF Sending of Called Numbers			
CALL TYPE	CALLED NUMBER		CALLING NUMBER
	REGULAR TRUNK GROUP	SUPER TRUNK GROUP	
0	KP+ STP	KP+ ST3P	KP+ID+7D+ST
0+7/10D	KP+7/10D+STP	KP+7/10D+ST3P	KP+ID+7D+ST
1+7/10D	KP+7/10D+ST	KP+7/10D+ST2P	KP+ID+7D+ST
011+CC+NN	KP+1+CC+NN+ST	KP+1+CC+NN+ST2P	KP+ID+7D+ST
01+CC+NN	KP+1+CC+NN+STP	KP+1+CC+NN+ST3P	
010	KP+ 1+STP or KP+ 10+STP	KP+ 1+ST3P or KP+ 10+ST3P	KP+ID+7D+ST
<p>0+ = operator-assisted call, more digits dialed 0- = operator-assisted call, no other digits dialed 1+ = DDD call CC = country code NN = national number ID = information digit KP = prepare for digits signal ST = end of pulsing STP = premium ST2P = identifier error</p>			

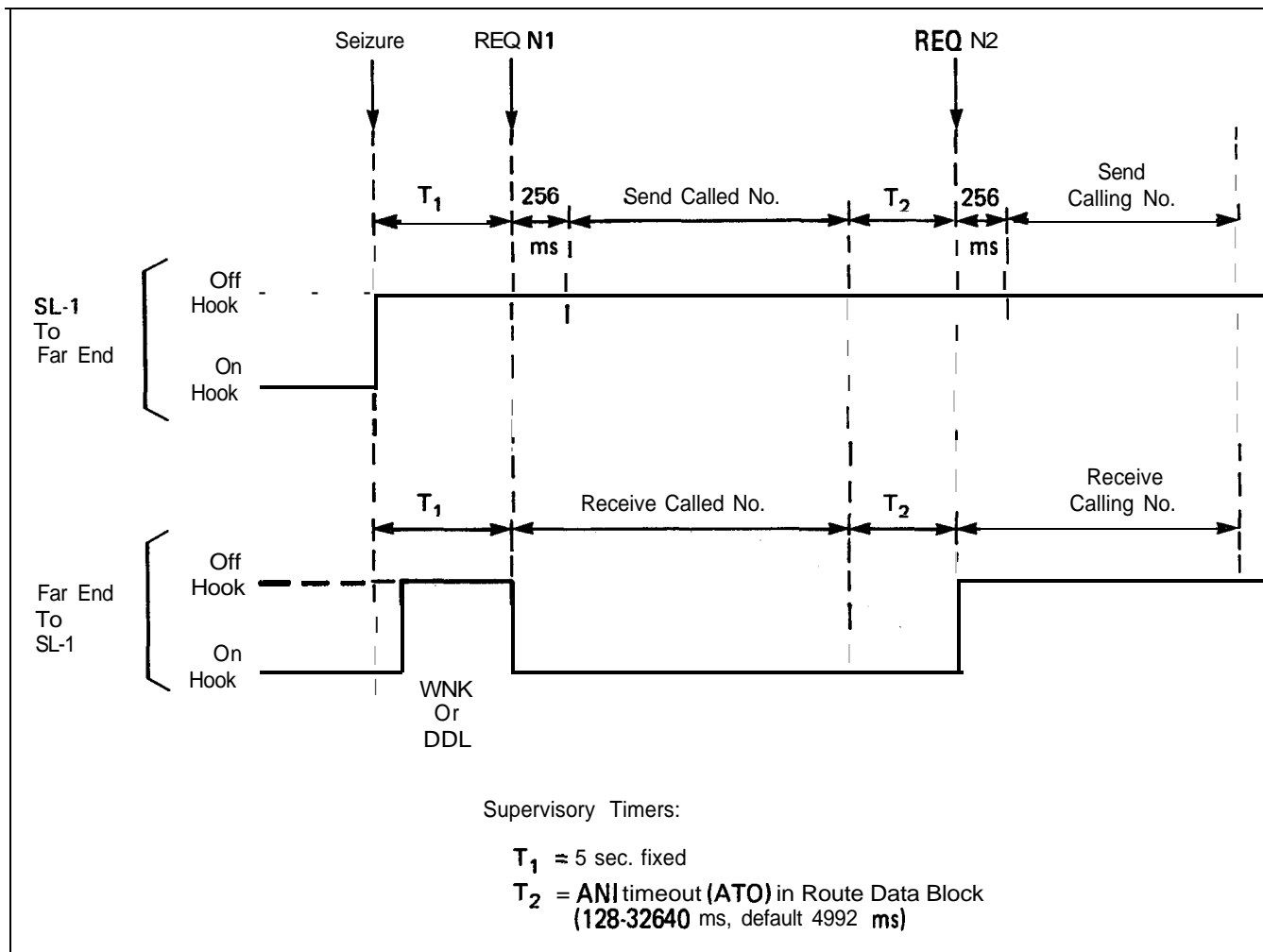


Fig. 2-1 — Supervisory Signals, Bell Method

TABLE 2-B
CALLED AND CALLING NUMBER INFORMATION **FORMAT**
NT400 METHOD

MODE	CALL TYPE	CALLED NUMBER	CALLING NUMBER
A	0+	KP+0+7/10D+ST	KP+CAT+7D+ST ¹
	0-	KP+0+ST	KP+CAT+7D+ST ¹
	1+	KP+1+7/10D+ST	KP+CAT+7D+ST ¹
B	1+	KP+7/10D+ST	KP+CM+CAT+7D+ST ¹
	0-	KP+ST	KP+CM+CAT+7D+ST ¹
	1+	KP+7/10D+ST	KP+CM+CAT+7D+ST ¹

TABLE 2-C
CALLED AND CALLING NUMBER INFORMATION **FORMAT**
NT500 METHOD

MODE	CALL TYPE	CALLED NUMBER		CALLING NUMBER
		DP SENDING	MF SENDING	
A	0+	0+7/10D	KP+0+7/10D+ST	KP+CAT+7D+ST ¹
	0-	0	KP+0+ST	KP+CAT+7D+ST ¹
	1+	1+7/10D	KP+1+7/10D+ST	KP+CAT+7D+ST ¹
B	0+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST ²
	0-	not applicable	KP+ST	KP+CAT+7D+ST ²
	1+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST ²
C	0+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST ¹
	0-	not applicable	KP+ST	KP+CAT+7D+ST ¹
	1+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST ¹

Where:

CM = 1 (for 1+ calls)
 = STP (for 0± calls)
 CAT = XX (category digits)

Where:

x = 0, 1,...,9, and XX is customer-defined data defining the type of long-distance call;

ST¹ = ST (normal),
 = ST2P (identifier failure)
 ST² = ST2P (identifier error),
 = KP (station-to-station 1+),
 = STP (premium Ok).

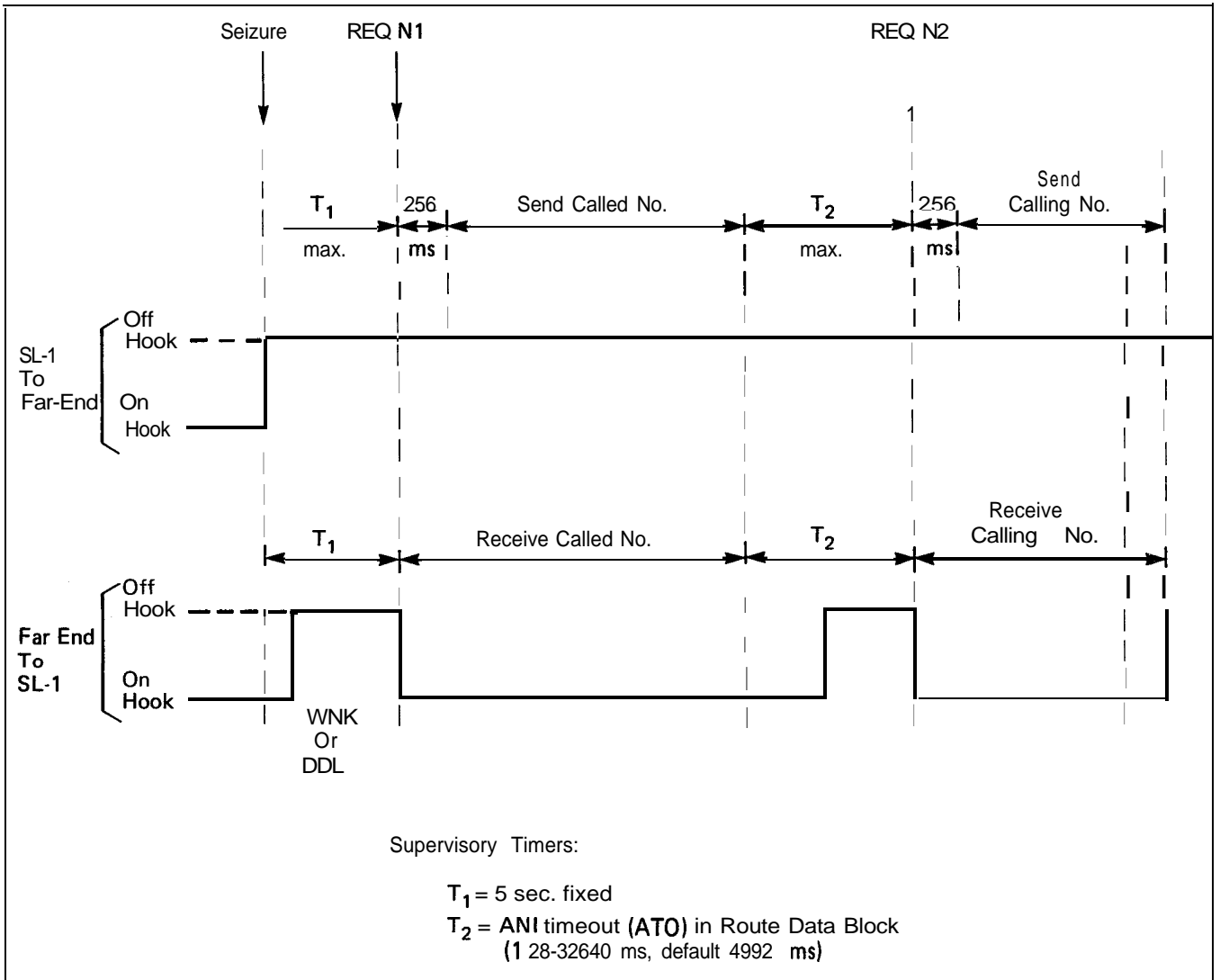


Fig. 2-2 — Supervisory Signals, NT400/500 Method

TABLE 2-D
POSSIBLE COMBINATIONS OF TRUNK TYPES AND AN1 METHODS

TRUNK TYPE	METHODS					
	BELL	NT400 A	NT400 B	NT500 A	NT500 B	NT500 C
CAMA-MF	A	A	A	A	A	A
CAMA-DP	A	N	N	A	N	N
CCSA-MF	A	A	A	A	A	A

A = allowed
N = not allowed

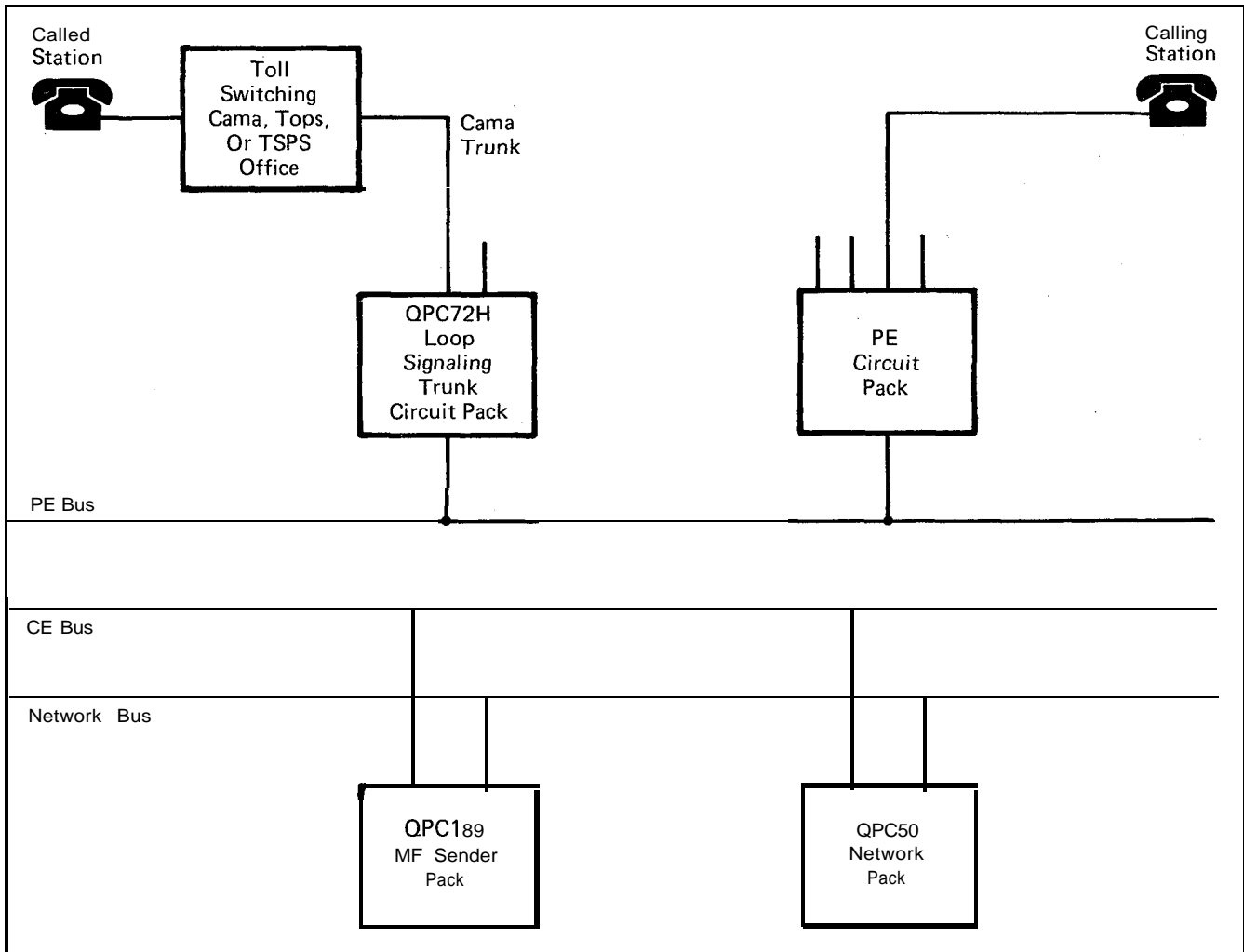


Fig. 2-3 — Hardware for ANI Feature

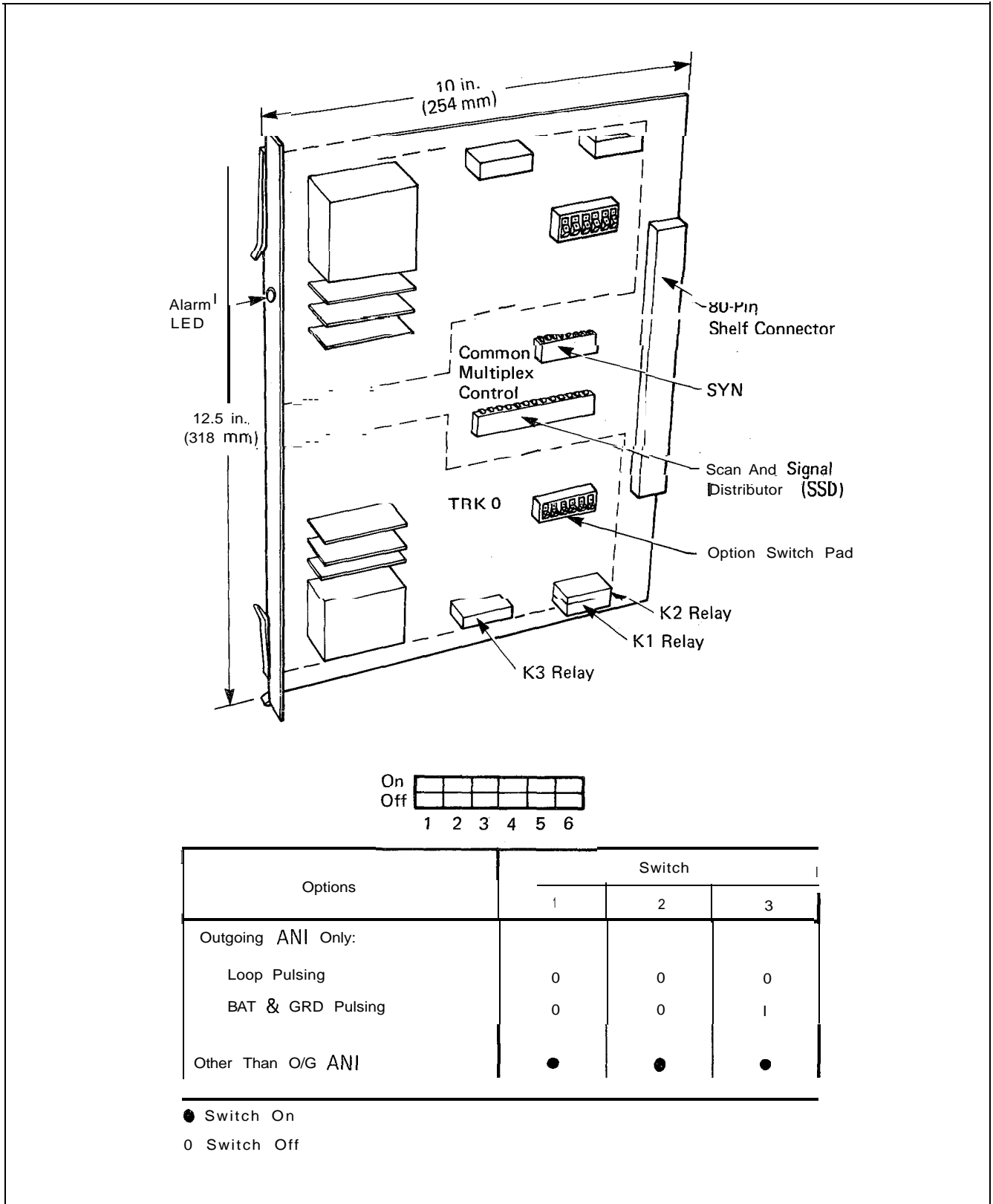


Fig. 2-4 – QPC72 Loop Signaling Trunk Circuit Pack Address Switch Settings

2.09 The calling number information is obtained immediately before being sent. Calls that are subsequently modified (attendant extended calls, transfer, etc.,) are completely billed against the party that initiated the trunk call (this practice is consistent with SL-1 AIOD).

SOFTWARE

2.10 The software portion of the AN1 feature performs the following functions:

- (1) provides identification of an originating, outgoing toll call;
- (2) determines the calling station identification, and controls the signaling and **supervision** of the AN1 trunk circuit according to the call sequence described in Part 3;
- (3) sets up a connection between the MF sender and the AN1 trunk circuit;
- (4) loads up to 16 digits that are to be MF outpulsed over the AN1 trunk into the MF sender;
- (5) orders initiation of the outpulsing;
- (6) takes down the connection between the trunk and the MF sender and sets up the speech path to the trunk, according to the call sequence described in Part 3.

HARDWARE

2.11 The only hardware required for AN1 consists of one QPC 189 MF sender circuit pack and as many trunk circuit packs as needed. Each trunk circuit pack is capable of handling two trunk lines. Each MF sender pack can handle up to 1400 calls an hour. Figure 2-3 shows a block diagram of the hardware.

DP OUTPUTSER

2.12 The capability to **outpulse** at 10 or 20 pps is provided by tone and digit switch circuit packs. A short description of each is given in **553-2YY 1-150**.

MF SENDER

2.13 The MF sender circuit pack QPC 189, located on the CE shelf, provides for MF signaling of AN1 digits over **CAMA** trunks to toll switching **CAMA**, TOPS, or TSPS offices.

2.14 The MF sender designed for the SL-1 system has the following characteristics:

- The maximum number of digits in an AN1 string that can be outpulsed by the MF sender is 16.
- Minimum number of digits in an AN1 string must be 2; i.e., KP + ST.
- No MF signaling is permitted over time slots 0 and 1.
- Any number of MF-AN1 sequences (up to a maximum of 30) can be outpulsed simultaneously over various **CAMA** trunks.
- The **first** digit (usually KP) in an AN1 digit string is pulsed for 136 ms followed by a silent interval of 68 ms. The remaining digits are each pulsed for 68 ms followed by 68 ms of silence.

2.15 The MF sender is located in any one of pack slots 3 through 11 in a QSD2 network shelf.

2.16 The MF sender has a control memory of 12 bits for every time slot and a digit buffer which contains 16 digits (4 bits each) per time slot. The processor can write into four types of registers and read two types of registers. Data store requirements are the same as those of a tone and **digit** switch (2 words protected, and 55 words unprotected). See 553-2001-15 1 for a detailed calculation of memory requirements.

TABLE 2-E
LOOP SIGNALING TRUNK CIRCUIT CHARACTERISTICS

CAPACITY	2trunks
POWER FEED	306/306 ohms balanced, 48 V nominal
LOOP TERMINATION	270 ohms
PACK OPTIONS	(a) 600/900 ohms nominal impedance, (b) ANI or non-ANI , (c) loop, or battery and ground pulsing on ANI .
EXTERNAL CIRCUIT RESISTANCE	2000 ohms max (for 16.2 mA)
GROUND POTENTIAL DIFFERENCE	±10 V max
FAR END BATTERY LIMITS	-42.75 to -52.5 V
LINE LEAKAGE	30 kΩ minimum
PACK APPLICATIONS	(1) Direct In-Dialing Trunk (DID). (2) Direct In-Dialing Direct Out-Dialing (DID/DOD) trunk. † (3) 2-way Dial Repeating (2DR) trunk.* (4) Outgoing Automatic Incoming Dial. (OAI D) trunk. (5) Outgoing Automatic Number Identification (OANI) trunk.

* In applications 2 and 3, battery and ground pulsing is provided.

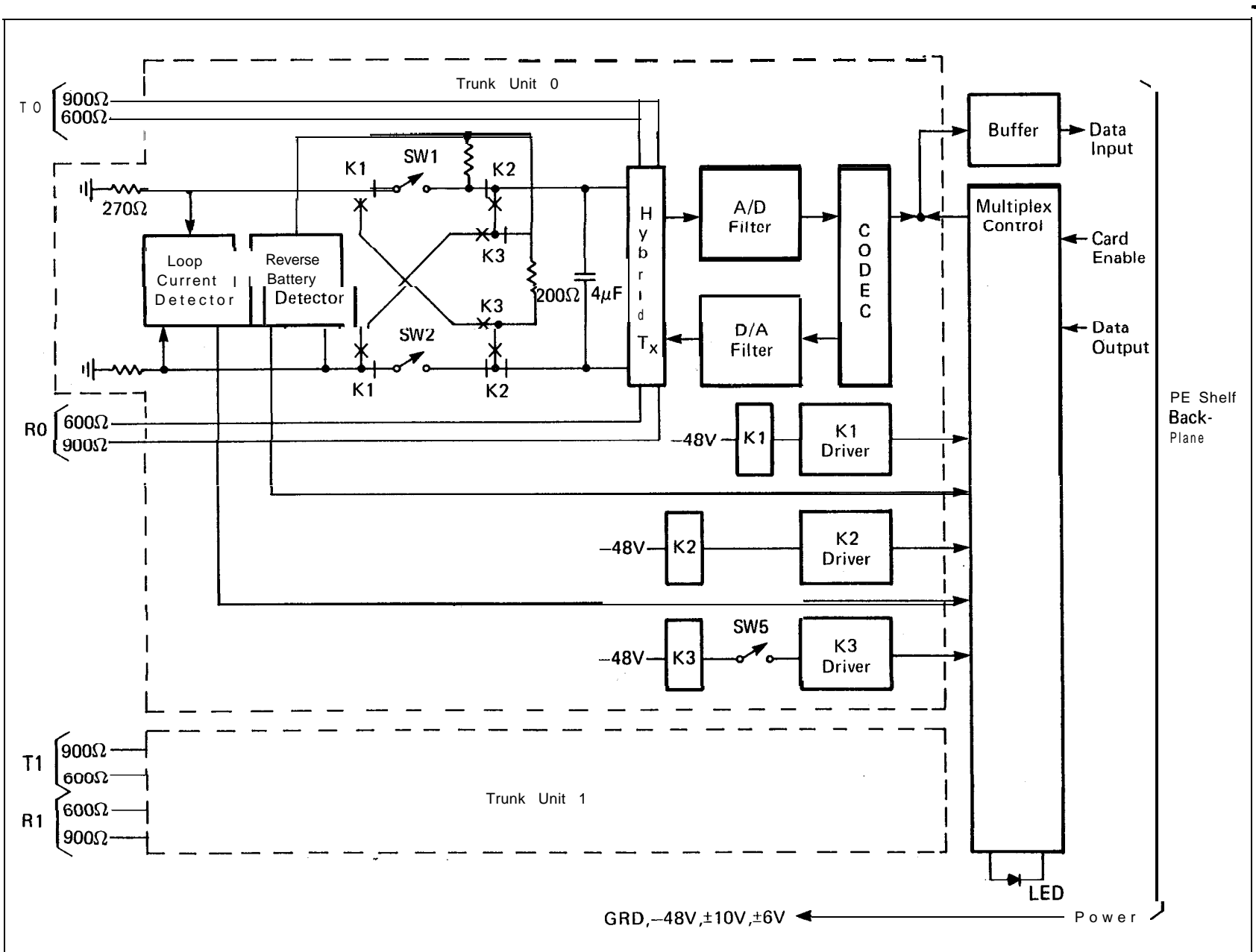


Fig. 2-5 Loop Signaling Trunk Circuit Pack Block Diagram

TABLE 2-F
SIGNAL FREQUENCIES FOR MF OUTPULSING

DIGIT OR SYMBOL	FREQUENCIES (Hz)
1	700 + 900
2	700 + 1100
3	900 + 1100
4	700 + 1300
5	900 + 1300
6	1100 + 1300
7	700 + 1500
8	900 + 1500
9	1100 + 1500
0	1300 + 1500
K P	1100 + 1700
S T	1500 + 1700
S T P	900 + 1700
S T 2 P	1300 + 1700
S T 3 P	700 + 1700

POWER OUTPUT – each frequency is -6 ± 0.7 dBm at the MDF.

POWER VARIATION – less than 0.5 dB between any two output frequencies.

BACKGROUND LEVEL:

Signal transmitted – any frequency other than the two signal frequencies to be 30 dB below the signal frequency level;

Signals not transmitted – 50 dB below signal frequency.

3. OPERATION

3.01 The following charts provide a description of the AN1 call sequence. The SL-1 system intercepts all outgoing calls over **CAMA** trunks, if 1 or 0 is not dialed following the trunk access code. This restriction does not apply to outgoing calls over CCSA AN1 trunks.

CHART 3-1
AN1 CALL SEQUENCE (Via **CAMA** MF Trunk)

STEP	PROCEDURE
1	<p>The calling party accesses the trunk group by:</p> <ul style="list-style-type: none"> dialing the CAMA trunk group access code the optional route selection mechanism for ANI, or Automatic Route Selection (ARS). See 553-265 1-200. <p>The call is intercepted and call sequence terminated, if:</p> <ul style="list-style-type: none"> ● the station is denied access to the CAMA trunk group, or ● the station has toll denied (TLD) class of service and the CAMA trunk group is not code restricted. <p>If call is not intercepted, continue to Step 2.</p>
2	An overflow tone is heard, if all trunks are busy. A free CAMA trunk is assigned, and it is made busy for outgoing calls. An optional second dial tone is provided if the trunk group was accessed directly with other than a four-digit access code. The trunk is not seized at this time. ←
3	The calling party dials the called number. Second dial tone (if given in Step 2) is removed with the first digit dialed. Invalid digits result in an overflow tone and the call sequence is terminated.
4	<p>SL-1 can identify the last digit dialed according to the method in Table 3-A. Partial dial results in line lockout treatment for the calling station. Otherwise, the CAMA trunk is seized by sending the seizure signal. Depending on the class-of-service of the CAMA trunk, sending of the called number is then initiated according to one of the following methods.</p> <p><i>Immediate Start:</i> Wait 1 second, then send the called number (go to Step 5). CAMA MF trunks will not normally be immediate start.</p> <p><i>Wink or Delay Did:</i> A 5 second timer T1 is started. If the wink or delay dial signal is received before T1 expires, reset T1, wait 256 ms, and go to Step 5. If wink or delay dial signal is not received, go to Step 8.</p>
5	The called number information is sent using MF sending. The format of this information depends on the method used. After completing the sending of the called number, the SL-1 number system sets up the speech connection and starts time T2 (adjustable timer, maximum 64 seconds, default is 5 seconds). Continue to Step 8 if the timer expires. If the timer does not expire, go to Step 6.
6	The CAMA office sends the AN1 request signal. Timer T2 is reset. Speech connection is removed, and the calling number information is sent (in MF). Having completed the sending of the calling number information, SL-1 establishes the speech connection.

Chart Continued

CHART 3-1 Continued
AN1 CALL SEQUENCE (Via **CAMA** MF Trunk)

STEP	PROCEDURE
7	AN1 sequence is completed.
8	Timer T1 or T2 expires: the trunk trouble threshold is incremented and the trunk is released (after 3 seconds it is made idle for outgoing calls). A second and subsequent attempts are made.
<i>REMARKS</i>	
1	Auto dial or speed calling may be used to place AN1 calls. When used, second dial tone is not provided.
2	The Bell method may be used to interface SL-1 to Stromberg Carlson Type 40 systems. In such situations, timer T2 is not set in Step 5 of the AN1 call sequence.
3	Some CAMA offices do not send the AN1 request signal if the call is not billable. Examples are calls to long distance directory information and calls intercepted by an operator or recorded announcement. If the call is completed to an operator, the request signal is replaced by answer supervision. If not, no supervisory signal is sent. In systems operating in this way, timer T2 must either not be set at all or be set to a sufficiently large value (maximum 64 seconds) to prevent expiry during an intercepted call. If answer supervision is received, the AN1 sequence is aborted and the call is established.

CHART 3-2
AN1 CALL SEQUENCE (Via **CAMA** DP Trunk)

In order to minimize the post-dialing delay, the sending of the called number starts as soon as the first digit has been dialed, and the toll **office** is ready to receive. The call sequence is the same as that established in Chart 3-1, with the following differences:

STEP	PROCEDURE
1	As soon as the trunk access code has been dialed, the CAMA trunk is assigned 'made busy' for outgoing calls and seized. If specified by class-of-service of the trunk group, a second dial tone is provided locally unless a four-digit access code was used. If not, a talking path is set up .
2	The sending of the called number is performed digit by digit. Similar to what is stated in Chart 3-1, Step 4 sending does not start until the CAMA office is ready.

Recognition of the Last Digit	REQ__N2	Request for the calling number information.
3.02 The method, used by SL-1 system to identify the last digit with CAMA trunk operation, is specified in Table 3-A.	DISCONNECT	This signal results in the call being disconnected, unless the class-of-service specified for the Near End is Originating Party Control .
SUPERVISORY SIGNALS BETWEEN SL-1 AND TOLL OFFICE		
3.03 The signaling method used with AN1 trunks can be E&M, DX or loop signaling. The signals that may be received and recognized by SL-1 are :	HIT	This signal is ignored by the SL-1.
REQ-N1	BUSY__OR__ORIG	This signal is a permanent off-hook from an idle trunk.
Request for the called number information.		

TABLE 3-A
RECOGNITION OF LAST DIGIT - **CAMA** TRUNKS

CALL TYPE	RECOGNITION METHOD
0-	4-second timer after first digit (if 0)
(0+1) X1 X2...X7	X 1 \neq 1, 4-second timer after X7
(0+ 1) X1 X2...X10	X 1 \neq 1, 4-second timer after X7, X8 dialed before timer expires
01 X1 X2...Xn 7 \leq n \leq 12	first 2 digits = 01, X1\neq1 and 4-second timer after X7, X8, X9,...X11
011 X1 X2...Xn 7 \leq n \leq 12	first 3 digits = 011, 4-second timer after X7, X8,...X11
010	first 3 digits = 010

4. INSTALLATION

4.01 Locations of both **QPC71** and **QPC72** packs are the same as those for any trunk circuit pack: in the Peripheral Equipment (PE) shelf. Trunk connections are the same as those shown in 553-2001-186 and 553-2001-187.

4.02 The MF sender pack, however, must be located in a network card position in a Common Equipment (CE) shelf. Installation procedures for all packs are described in **553-2YY1-210**. To test the MF sender pack for proper operation, refer to Part 5 of this section.

4.03 Refer to 553-2001-220 and 553-2001-221 for service change procedures for the SL- 1 system recognition of the AN1 feature.

5. ERROR DETECTION AND MAINTENANCE

5.0 1 The SL- 1 system performs on-line error detection by monitoring signaling irregularities and failures. Such irregularities cause trunk trouble thresholds to be incremented and, if persistent, they result in a diagnostic message on the maintenance Teletypewriter (TTY).

5.02 Testing the **CAMA** trunks used with AN1 can be accomplished **using** the TRK LS C U command in Overlay 36 (trunk diagnostic program 1). Another diagnostic (Overlay 46) is provided to test the MF sender. Refer to Chart 5-1. ←

5.03 All diagnostic output messages and input formats are explained in 553-2001-505. ←

CHART 5-1
TESTING THE MF SENDER

STEP	PROCEDURE
1	Load overlay program 46.
2	Input the command MFS L. If the MF sender pack passes the tests, but is nevertheless suspected of sending incorrect digits, then the TONE command should be used. To use this command, and SL-1 maintenance set input pad is required. The user presses the pad keys and listens to the MF tones being generated.
3	Enter TONE#L#ALL## . This causes the diagnostic to produce a 10-second burst for each digit from 0 through 15. This is a rudimentary check to see that each digit sounds different. If some tones sound the same, use TONE#L## to pursue the problem further. This command enters the user into a special input mode which causes the MF sender at loop L to output a digit burst that corresponds to the SL-1 key depressed (see Table 6-A). This feature allows comparison between arbitrary digits and is useful in comparing to digits suspected of producing the same MF frequencies.
4	To return to the command input environment, either enter a END## sequence on the SL-I set or go off-hook, then on-hook, or press * key.

TABLE 5-A
TONE COMMAND TABLE

SL-1 KEY DEPRESSED	MF DIGIT OUTPUTSED
pad key 1	1
pad key 2	2
pad key 3	3
pad key 4	4
pad key 5	5
pad key 6	6
pad key 7	7
pad key 8	8
pad key 9	9
pad key 0	0
pad key *	END TONE command
strip key 0 (prime DN)	10 (ST-3P)
strip key 1	11 (STP)
strip key 2	12 (KP)
strip key 3	13 (ST-2P)
strip key 4	14 (ST)

6 . ROUTE SELECTION (RS-ANI) OPTION

6.01 The AN1 route selection mechanism is optional and may be used in conjunction with the AN1 feature to route toll calls automatically over specified trunks to toll offices, and local calls over CO trunks to local switching offices.

6.02 The station user places all outgoing CO calls by dialing the RS-ANI access code, typically '9', followed by a CO directory number. If the first digit dialed, following the access code, is 0 or 1, the call is routed over a toll trunk group; if that first digit is not 0 or 1, the call is routed over a CO trunk group.

OPERATION

6.03 After dialing the RS-ANI access code, the user receives second dial tone from SL-1. If no digits are dialed during the next 30 seconds, dial tone is removed and the user hears overflow tone for a further 15 seconds. Otherwise, the second dial tone is removed, when the first digit is dialed. The action taken next depends upon the next digit(s) dialed, and is specified in Table 6-A.

If * is dialed in place of any digit in the table, the * is ignored.

If # is dialed in place of any digit in the table, the call is intercepted by an overflow tone. The exception to this rule is that '0#' acts like a timeout on the 4-second timer.

TABLE 6-A
RS-ANI OPERATION

DIGIT DIALED	ACTION TAKEN BY SYSTEM								
0	A 4-second timer is started to monitor the next digit dialed. Routing is based on this digit, as follows: <table border="1"> <thead> <tr> <th>DIGIT</th> <th>ACTION</th> </tr> </thead> <tbody> <tr> <td>none</td> <td>The timer times out and the call (0-) is routed over the trunk group specified for 0- calls.</td> </tr> <tr> <td>1</td> <td>The timer is canceled and the call (IDD) is routed over the trunk group specified for 1+ of IDDD calls.</td> </tr> <tr> <td>2-9</td> <td>The timer is canceled and the call (0+) is routed over the trunk group specified for 0+ calls.</td> </tr> </tbody> </table>	DIGIT	ACTION	none	The timer times out and the call (0-) is routed over the trunk group specified for 0- calls.	1	The timer is canceled and the call (IDD) is routed over the trunk group specified for 1+ of IDDD calls.	2-9	The timer is canceled and the call (0+) is routed over the trunk group specified for 0+ calls.
DIGIT	ACTION								
none	The timer times out and the call (0-) is routed over the trunk group specified for 0- calls.								
1	The timer is canceled and the call (IDD) is routed over the trunk group specified for 1+ of IDDD calls.								
2-9	The timer is canceled and the call (0+) is routed over the trunk group specified for 0+ calls.								
1	The call is routed over the trunk group specified for 1+ or IDDD calls.								
2-9	The call is routed over the trunk group specified for other (local) calls.								

RESTRICTIONS

6.04 Tie trunks may access RS-ANI in the same way as stations do, but all other trunks are intercepted.

6.05 Any type of trunk may be used for RS-ANI, with the exception of special purpose trunks such as paging, dictation, or recorded announcement. Normally, however, trunk routes of the following types are used:

CALL TYPE	TRUNK TYPE
0± 1+, 011+, 01+, 010- other	CAMA CAMA c o

DATA STORE

6.06 A data block is required for each RS-ANI access code, specifying the trunk routes to be used for each call type. The approximate size is seven words. Program store is approximately 0.2k of the resident programs. This is an optional package. Real time used is approximately 30 ms (basic system) for each RS-ANI call.

CLASS-OF-SERVICE OPTION

6.07 A new station class-of-service option, conditionally unrestricted (CUN), is available to enforce the placing of non-ARS handled toll calls through the ANI process, i.e., over **CAMA** trunk groups. See 553-2001-220 or 553-2001-221 for the implementation of CUN.

CUN = UNR for calls placed through ARS and for ANI calls placed either directly or through RS-ANI;

CUN = TLD for all other calls.

TABLE 6-B
CLASS-OF-SERVICE OPTIONS

OPTION	EXPLANATION
UNR	UNRESTRICTED Allowed to receive calls from, and originate calls to, exchange network (CO, FX, WATS). This includes toll calls.
CUN	CONDITIONALLY UNRESTRICTED = UNR for calls placed through ARS (see 553-2651-200); and for calls placed through ANI = TLD for all other calls.
CTD	CONDITIONALLY TOLL DENIED = UNR for calls placed through ARS. = TLD for all other calls.
TLD	TOLL RESTRICTED SERVICE allowed to receive calls from exchange network; allowed dial access to local exchange network; allowed access to toll network via SL-1 attendant only; denied access to exchange operator.

SL-1
BUSINESS COMMUNICATIONS SYSTEM
 (GENERIC 103,203 AND LATER)
AIOD C25 DATA TRUNK
 (AUTOMATICALLY IDENTIFIED OUTWARD DIALING)
 FUNCTIONAL DESCRIPTION

1. GENERAL

1.01 This section outlines the functions, interconnections, characteristics, and operation of the Automatically Identified Outward Dial (AIOD) C25 data trunk circuit. The information is intended to be used as a guide in designing or engineering the connection between the **SL-1** Business Communications System and the station identification frame in the Central Office (CO).

2. DESCRIPTION

FUNCTION

2.01 The AIOD C25 data trunk used in conjunction with a **118A** interconnecting unit, provides a connection — over a voice-grade cable pair between the **SL-1** automatic number identification equipment and the station identification frame in a Bell System CO. This feature provides the capability of billing outgoing toll calls to individual extension numbers.

2.02 **Each trunk circuit provides the following features:**

- (a) standard EIA serial binary pulses as an input to the **118A** interconnecting unit,
- (b) checks for errors in the message sent from the Central Processing Unit (CPU) of the SL-1 to the AIOD data trunk,
- (c) relays signals regarding the status of the data to the CPU.

PHYSICAL DESCRIPTION

2.03 The AIOD C25 data trunk is mounted on a 12.5 inch (31.8 cm) by 10 inch (25.4 cm) printed circuit board which slides into one of the ten slots in a Peripheral Equipment (PE) shelf. Through an **80-pin** connector at the back of the board, the trunk connects to the switching system and the outside trunk circuits.

2.04 The **118A** interconnecting unit can be powered by the SL-1, but provision is not made for mounting any auxiliary apparatus within the cabinets.

INTERCONNECTION WITH SHELF

2.05 The 80-pin connector on the QPC162, AIOD C25 data trunk plugs into an **80-line** bus system on the back of the PE shelf (Fig. 1). These 80 bus lines feed into 7 **multipin** connectors. Two connectors link the PE shelves to other PE shelves and to the Common Equipment (CE). Four connectors link the line and trunk circuits to the cross-connect terminal or distributing frame and one feeds power to the PE from the converter shelf.

INTERFACE LEADS

2.06 Connection of the trunk circuit to the **118A** interconnecting unit is made via six leads. Clock, data, bid, and transmit leads are passed to the **118A** interconnecting unit which is connected to the CO station identification frame by a single pair of wires.

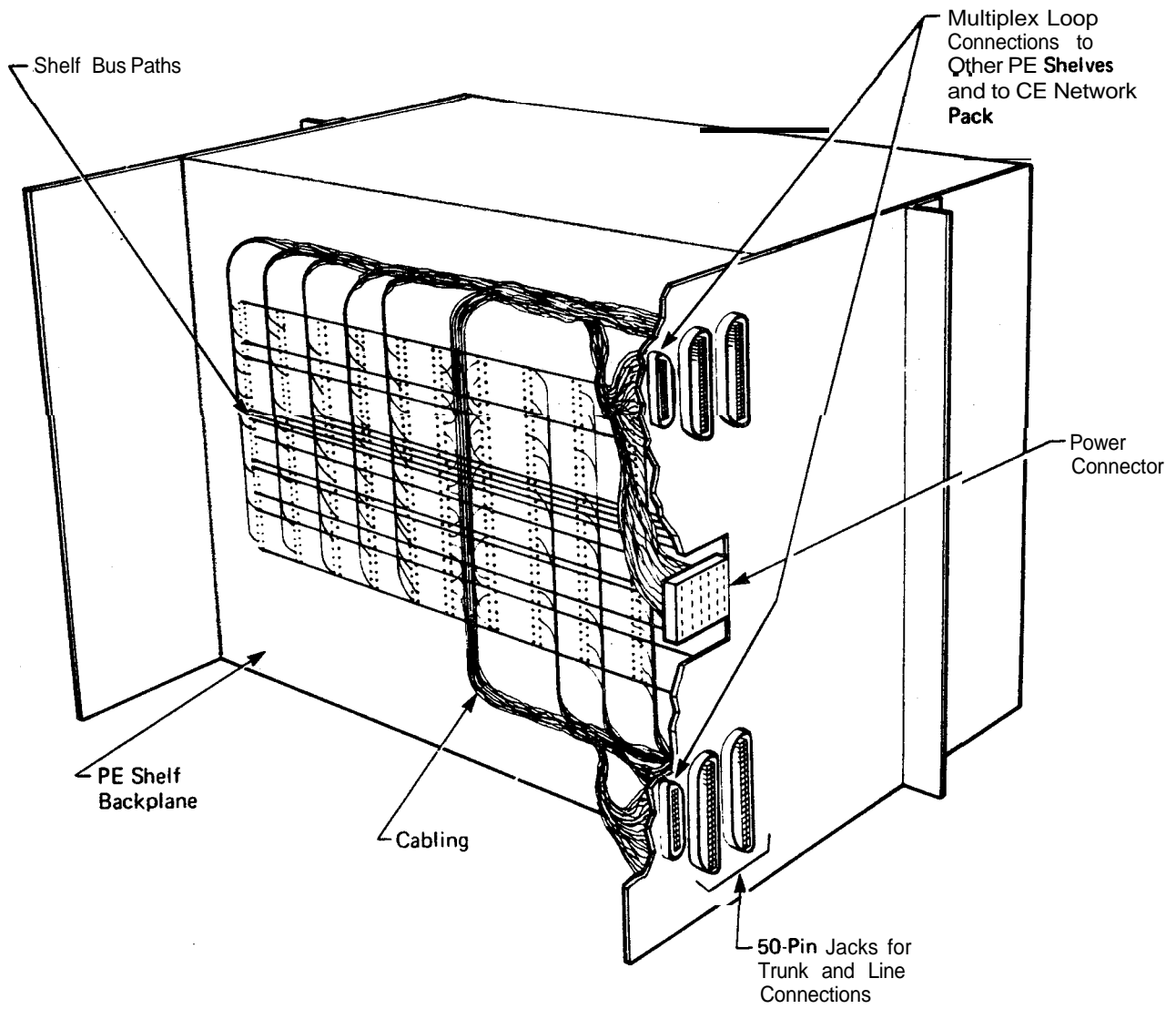
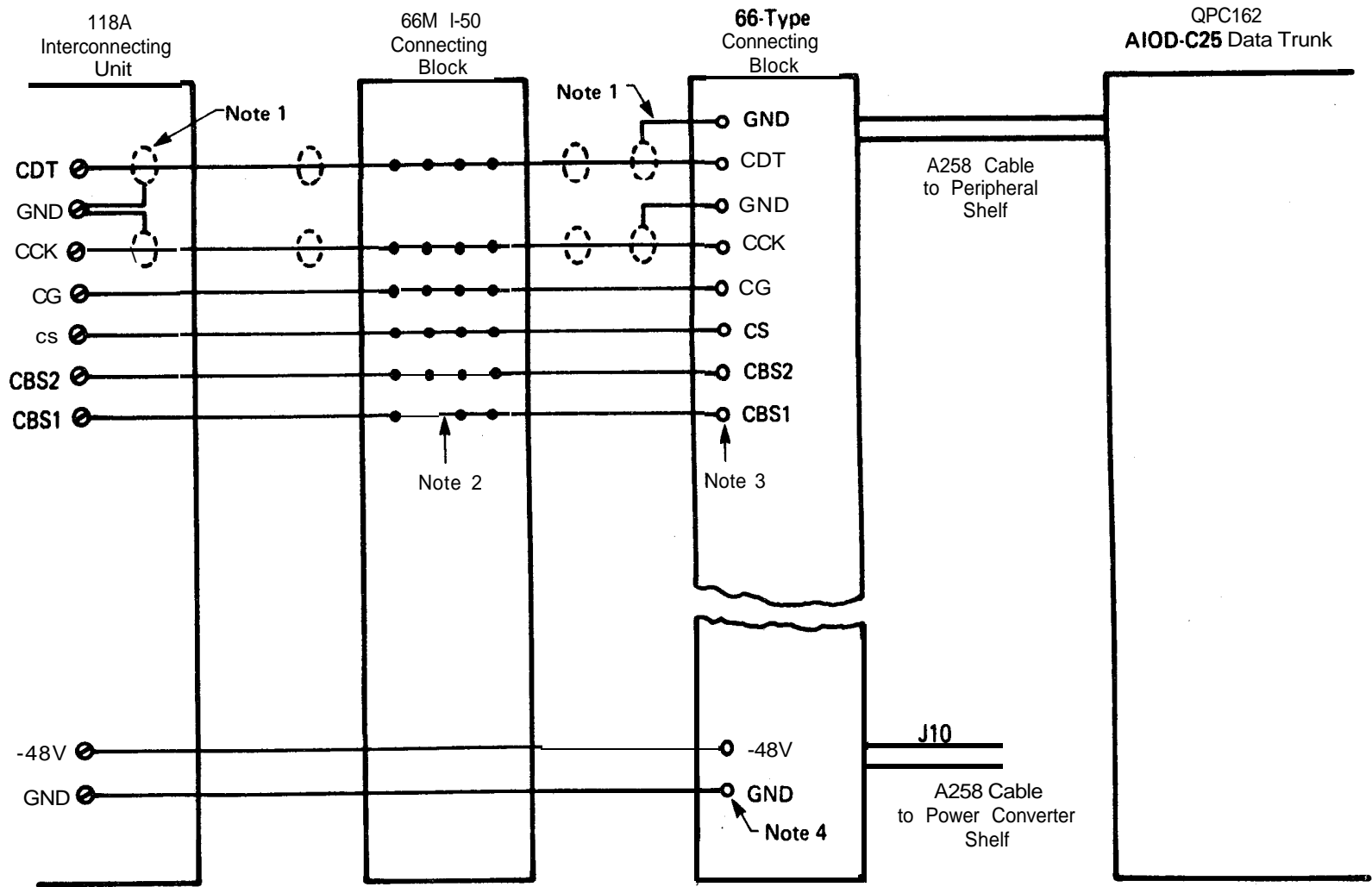


Fig. 1 - Rear View of PE Shelf (right-hand)



- Notes: 1. Connect shields of CDT and CCK leads to ground terminals on 118A IU and SL-1 terminal. Do not connect at interface terminal.
 2. 6 bridging clip.
 3. Location of these terminals depends on location of the QPC162 in the peripheral shelf (Table A).
 4. Pairs 10 through 19 of J10 cable from power converter shelf.

Fig. 2 -- AIOD Trunk Connections

SECTION **553-2621-180**

3. CIRCUIT CHARACTERISTICS AND LIMITATIONS

Circuits per pack **1**
Options none
Signal Range 50 feet (24 AWG cable pair)

Maximum Signal Lead
CCK Lead <**2500** pf
CDT Lead (2500 pf)

4. APPLICATION

4.01' This trunk circuit interfaces with a **118A** interconnecting unit [Bell System Practice (BSP) **463-380-101**] which in turn interfaces with a station identification frame in the CO.

5. OPERATION

5.01 When an extension makes a toll call, the following sequence takes place.

- (a) The CPU transmits a reset command to the data trunk. This resets all logic and enable Signal Scan Distribution (SSD) scan points.
- (b) Depending on the previous status of the trunk, the reset is reported to the CPU.
- (c) The CPU then sends a "ready" message which removes the reset.
- (d) Removal of the reset is reported to the CPU.
- (e) The CPU then sends the trunk and station information to the AIOD trunk in eight messages coded in a 2-out-of-5 code.

(f) During the transmission of the trunk and station number, the message is checked for **2-out-of-5** errors, and the CPU informed of the result.

(g) The AIOD trunk clamps the output to the CO to the "ones" state, if there is a **2-out-of-5** error.

(h) After transmission of the trunk and station information to the AIOD trunk register the CPU will send a "bid" command.

(i) The AIOD trunk initiates a "bid" request to the CO via the **118A** interconnecting unit, and when the transmit signal is received from the CO, the AIOD trunk reports it to the CPU.

(j) Upon receipt of the transmit request, the AIOD trunk begins transmission of trunk and station information and reports the end of transmission to the CPU.

(k) Any loss of transmit is reported to the CPU.

(m) After receiving the loss of transmit signal, the CPU disables the scan points.

6. DESIGN CONSIDERATIONS

6.01 The AIOD trunk must satisfy all the requirements given in BSP 463-380-101 for the Western Electric **118A** unit with which the trunk unit must interface. These requirements, when met, ensure proper transmission of data to the central **office** station identification frame.

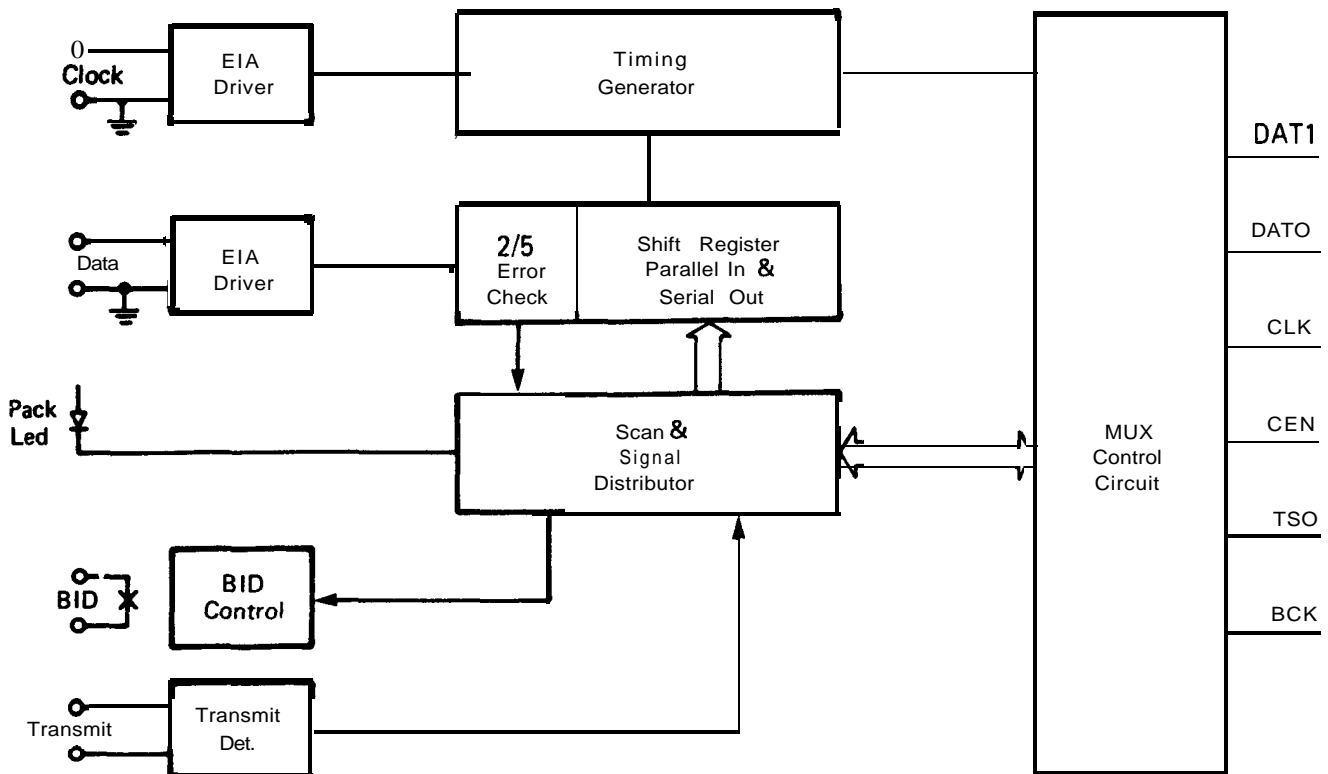


Fig. 3 - Block Diagram of AIOD-C25 Data Trunk

SL-1
BUSINESS COMMUNICATIONS SYSTEM
(GENERIC 103,203 AND LATER)
AIOD C25 DATA TRUNK
(AUTOMATICALLY IDENTIFIED OUTWARD DIALING)
DESCRIPTION, INSTALLATION, AND CONNECTIONS

1. DESCRIPTION

1.01 The AIOD-C25 data trunk, used in conjunction with a 118A interconnecting unit, provides a connection — over a voice-grade cable pair — between the SL-1 automatic number identification equipment and the station identification frame in a Bell System central office. This arrangement provides automatically identified outward dialing capabilities to the SL-1 business communications system. (AIOD is the abbreviation for Automatically Identified Outward Dialing.)

1.02 Connecting arrangement C25, when used with the SL- 1 business communications system, consists of a 118A interconnecting unit [described in

Bell System Practices (BSP), 463-380-101] and a QPC 162 printed circuit pack, both mounted in an SL- 1 peripheral shelf.

1.03 The QPC162, AIOD-C25 data trunk, is a double circuit pack equipped with an 80-contact edge connector for insertion into SL-1 peripheral shelves. A faceplate equipped with two locking devices is riveted to the front of the assembly. The assembly code number and abbreviated name is stenciled on the faceplate.

1.04 A Light Emitting Diode (LED) on the faceplate of the circuit pack is illuminated when the software has disabled the circuit.

2. INSTALLATION

2.01 The QPC 162, **AIOD-C25** data trunk should be installed in position one of any peripheral shelf, if available, to establish a high priority for serving the trunk. If position one is not available, then any other position (except the one dedicated to the QPC64 peripheral buffer) may be used.

2.02 Install the QPC162 circuit pack in the following manner.

- (a) Remove the front panel of the cabinet, and select the shelf and card position.
- (b) Insert the edges of the pack into the upper, and lower aligning guides.
- (c) Push the circuit pack in until resistance is felt.
- (d) Use the locking devices to push the circuit pack into the connector.

2.03 Install the **118A** interconnecting unit (BSP **463-380-101**), and make all the connections given in Part 3 of this section.

SOFTWARE

2.04 If the AIOD trunk is installed with a new system, the software program includes the AIOD trunk. However, if the **AIOD** trunk is added to an existing system, the following changes must be made.

- (a) Create a new *route* data block for the **AIOD** trunk.
- (b) Create a new *trunk* data block for the AIOD trunk.
- (c) Change customer data block to include **AIOD** trunk.
- (d) Change route data blocks.
- (e) Change all trunk data blocks.

Information for creating and changing data blocks is contained in 553-2001/2101-220 553-2001/2102-221

3. CONNECTIONS

3.01 Connections between the QPC 162, **AIOD-C25** data trunk and the **118A** interconnecting unit are made at the distributing frame or **cross-connecting** terminal. Table A shows on which pairs the leads to be cross-connected are found.

3.02 Use a 66M 1-50 connecting block equipped with B-bridging clips to make the connection between the SL-1 cross-connecting terminal and the **118A** interconnecting unit.

3.03 Use D inside-wiring cable for connecting the supervisory leads to the **118A** interconnecting unit. Use DL-1 or equivalent shielded wire for connecting the clock and data leads (CCK and CDT).

4. OPERATION TESTS

4.01 The operation of the AIOD trunk can be tested by loading overlay program 41 (see 55 3-200 1/2 10 1-505 for detailed description) and entering the commands by teletype as follows:

```

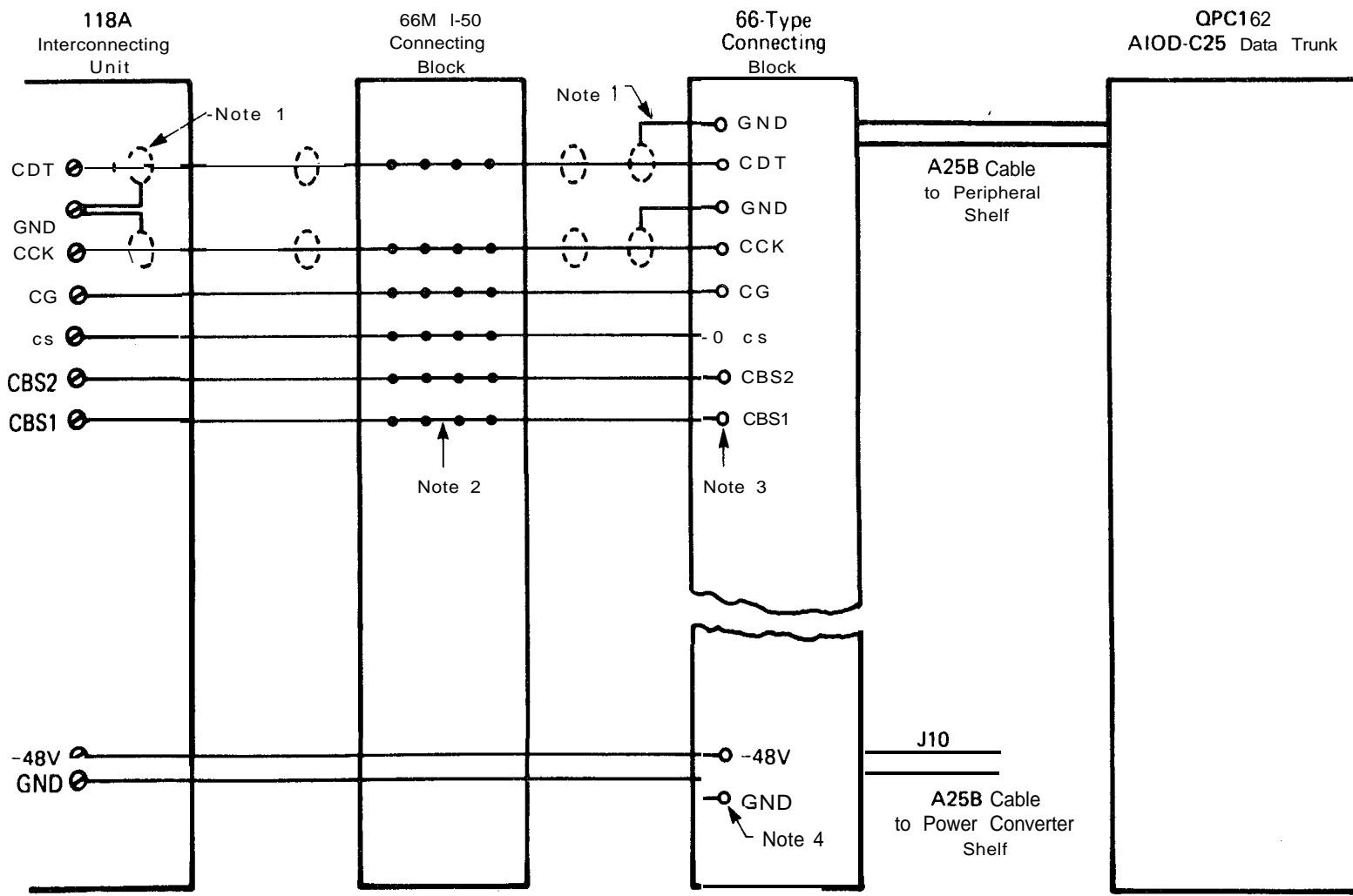
ENTER          PRINTOUT
LD4 1
AIOD L S C      OK

```

Note: L S C is the loop, shelf, and card number of the **AIOD** trunk.

4.02 If the response to the command **AIOD L S C** is one other than OK, refer to 553-2621-500, **AIOD** Trunk Fault-Clearing Procedures.

4.03 Abort the overlay program by typing * * * * and load the background signaling and switching diagnostic program, (see 553-2001/2101-505.)



- Notes: 1. Connect shields of CDT and CCK leads to ground terminals on 118A IU and SL-1 terminal. Do not connect at interface terminal.
 2. B bridging clip.
 3. Location of these terminals depends on location of the QPC162 in the peripheral shelf (Table A).
 4. Pairs 10 through 19 of J10 cable from power converter shelf.

Fig. 1 - AIOD Trunk Connections

TABLE A
PERIPHERAL SHELF PAIR-TERMINATING SEQUENCE

CABLE A

Note: Cable B for pack numbers 4, 5 & 6 and Cable C for pack number 7, 8 & 9 are terminated on Cable A. Similarly Cable D terminates pack 10 only (Pairs 1 thru 8).

Pair	Pin	Pair Color	Pack No.	Unit
				AIOD
1T	26	W-BL	1	GND
R	1	BL-W		CDT
2T	27	W-O		CND
R	2	O-W		CCK
3T	28	W-G		CG
R	3	G-W		CS
4T	29	W-BR		CBS2
R	4	BR-W		CBS1
5T	30	W-S	2	
R	5	S-W		
6T	31	R-BL		
R	6	BL-R		
7T	32	R-O		
R	7	O-R		
8T	33	R-G		
R	8	G-R		
9T	34	R-BR	3	GND
R	9	BR-R		CDT
10T	35	R-S		GND
R	10	S-R		CCK
11T	36	BK-BL		CG
R	11	BL-BK		CS
12T	37	BK-O		CBS2
R	12	O-BK		CBS1
13T	38	BK-G	3	
R	13	G-BK		
14T	39	BK-BR		
R	14	BR-BK		
15T	40	BK-S		
R	15	S-BK		
16T	41	Y-BL		
R	16	BL-Y		
17T	42	Y-O	3	GND
R	17	O-Y		CDT
18T	43	Y-G		GND
R	18	G-Y		CCK
19T	44	Y-BR		CG
R	19	BR-Y		CS
20T	45	Y-S		CBS2
R	20	S-Y		CBS1
21T	46	V-BL	3	
R	21	BL-V		
22T	47	V-O		
R	22	O-V		
23T	48	V-G		
R	23	G-V		
24T	49	V-BR		
R	24	BR-V		
25T	50	V-S	Spare	
R	25	S-V		

SL-1
BUSINESS COMMUNICATIONS SYSTEM
 (GENERIC 103,203, AND LATER)
AIOD C25 DATA TRUNK
 (AUTOMATICALLY IDENTIFIED OUTWARD DIALING)
FAULT-CLEARING PROCEDURES

1. GENERAL

1.01 This section contains information which is used to detect, isolate, and clear faults in the **AIOD-C25** trunk circuit. This circuit includes:

- (a) a QPC162 **AIOD-C25** data trunk,
- (b) a **118A** interconnecting unit,
- (c) a station identification frame, located in a Bell System Central **Office** (CO),
- (d) a cable pair connecting the **118A** interconnecting unit to the station identification frame.

1.02 A fault must be classified according to 553-2001-515 or 553-2101-515 as an Automatically Identified Outward Dialing (**AIOD**) trunk fault, before using the fault-clearing procedures given in this section.

2. CIRCUIT DESCRIPTION

2.01 The QPC162, **AIOD-C25** data trunk is usually located in position one of any Peripheral Equipment (PE) shelf; however, it could be located in any other position on a PE shelf.

2.02 The **AIOD-C25** data trunk used in conjunction with a **118A** interconnecting unit, provides a connection — over a voice grade cable pair — between the SL-I automatic number identification equipment and the station identification frame in a Bell System CO. (This arrangement provides automatically *identified outward dialing capabilities* to the SL-I Business Communications System.)

2.03 When a station set makes an outgoing call on a trunk that is arranged for **AIOD**, the Central Processing Unit (CPU) identifies the station and trunk being used and codes this information. This coded data is passed to the **AIOD-C25** data trunk and stored in a register.

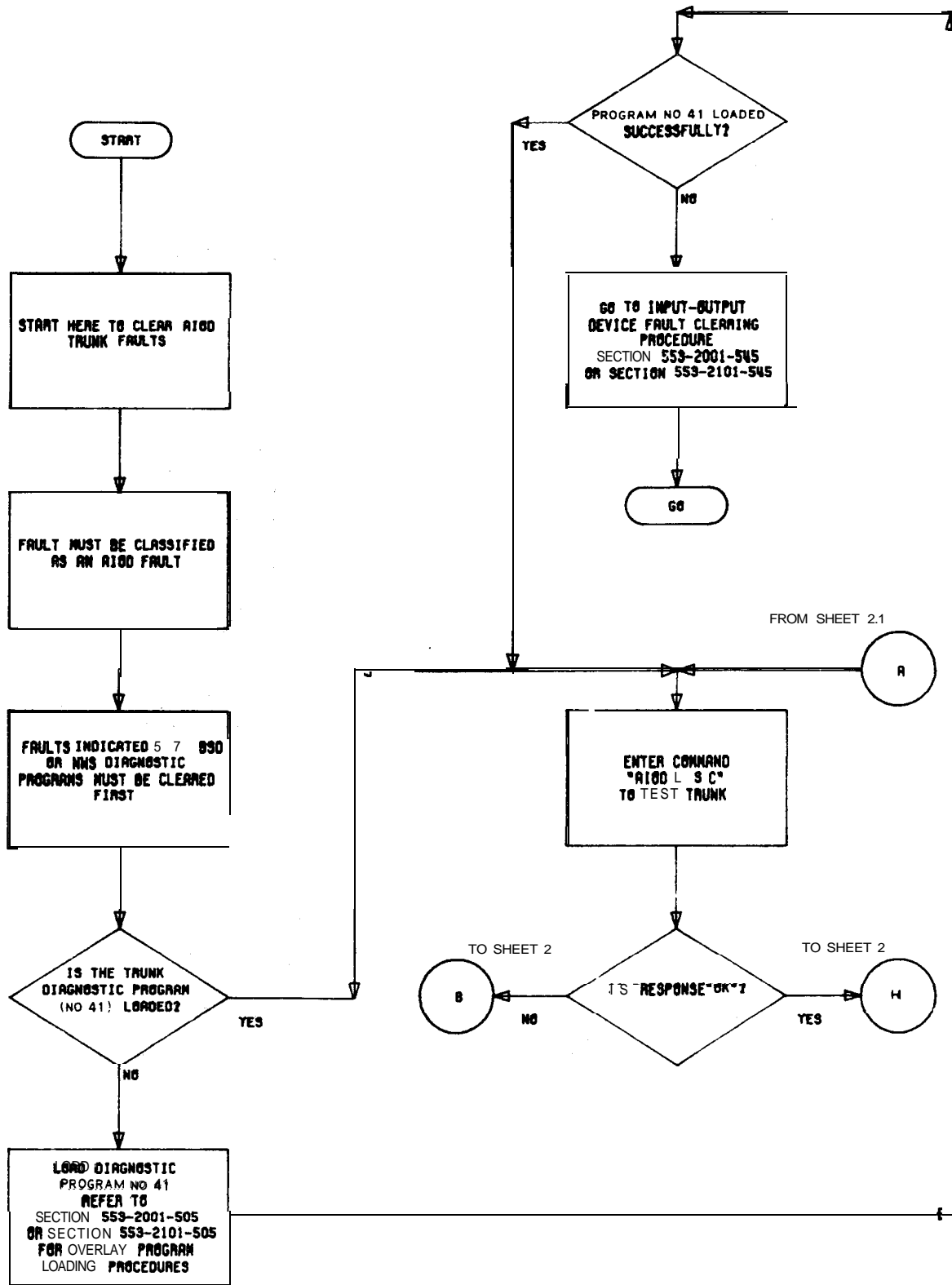
The CPU then sends a bid command, causing a relay to operate on the C25 trunk. The trunk signals the **118A** interconnecting unit via the CS and CG leads to notify the CO that a call is in progress, and station and trunk data is to be sent. The CO, when ready, returns a transmit request which is signaled back to the C25 trunk via the **118A** interconnecting unit by means of a contact closure on the **CBS1** and **CBS2** leads.

On receipt of the transmit request the C25 data trunk transmits binary data over the CCK and **CDT leads** to the interconnecting unit where the information is converted to Frequency Shift Keying (FSK) signals for transmission to the CO. When all the information has been transmitted, the SL-1 CPU and the CO are notified. Any errors in the data being transmitted are also signaled to the CPU and CO. The end of transmission is signaled by the removal of the bid request, which opens contacts across leads CS and CG. The CO acknowledges this by removal of the closure on leads **CBS1** and **CBS2**.

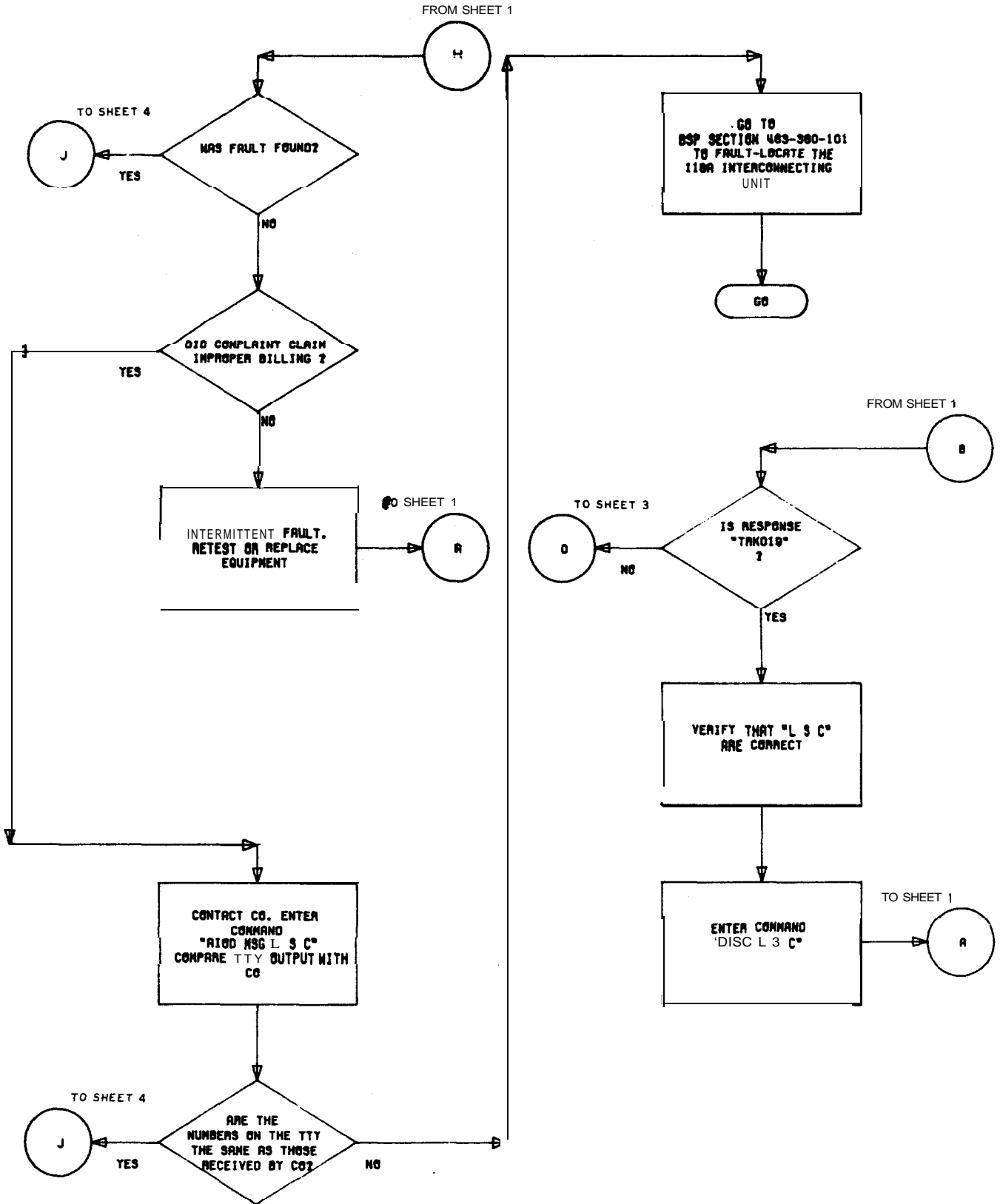
3. FAULT CLEARING

3.01 All SL-1 fault clearing must begin in 553-2001-515 or, for the SL-I VL in 553-2101-515 to ensure that faults are cleared in the most effective manner.

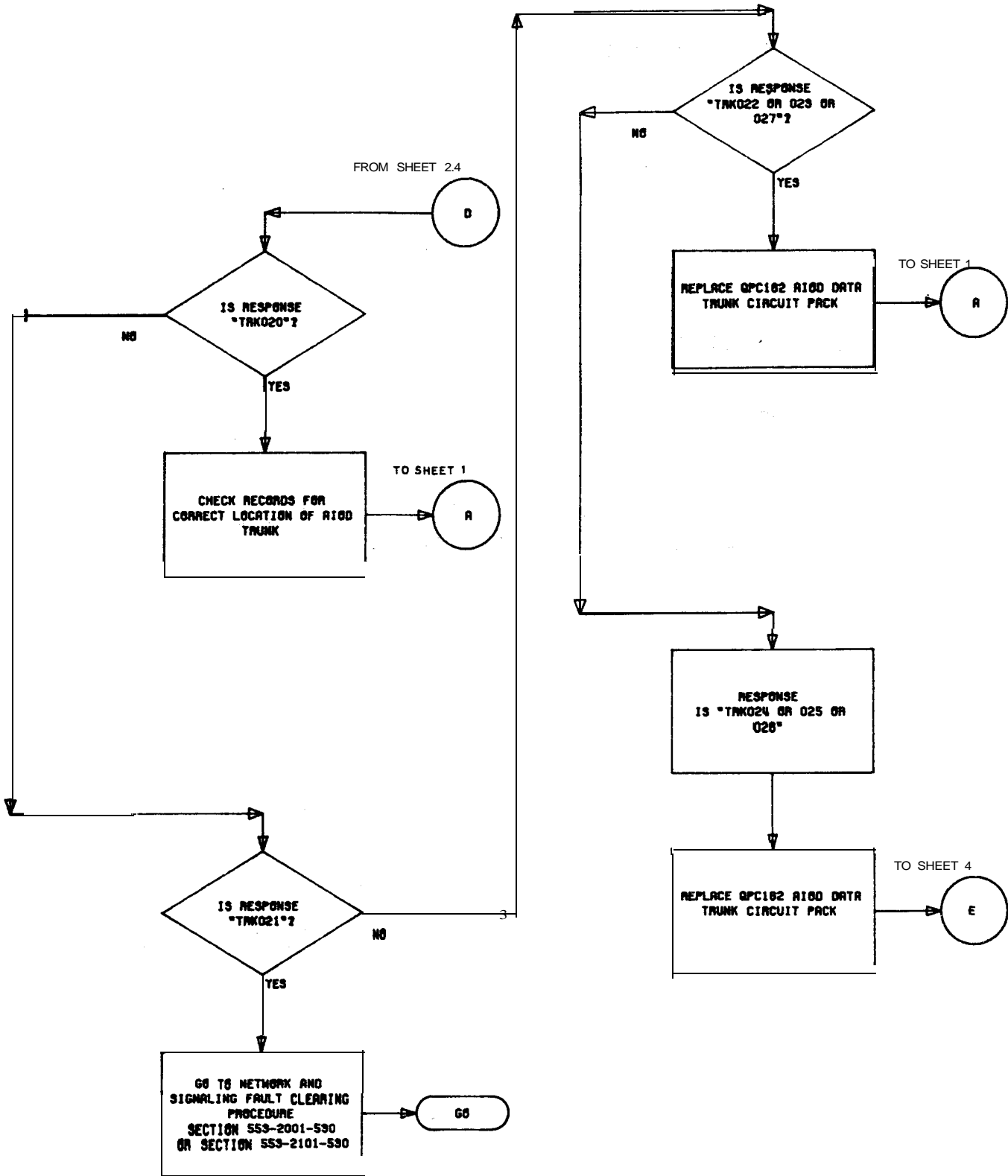
3.02 If the only fault is the **AIOD** trunk, the flowchart leads to this section after checking that there are no power or common equipment faults.



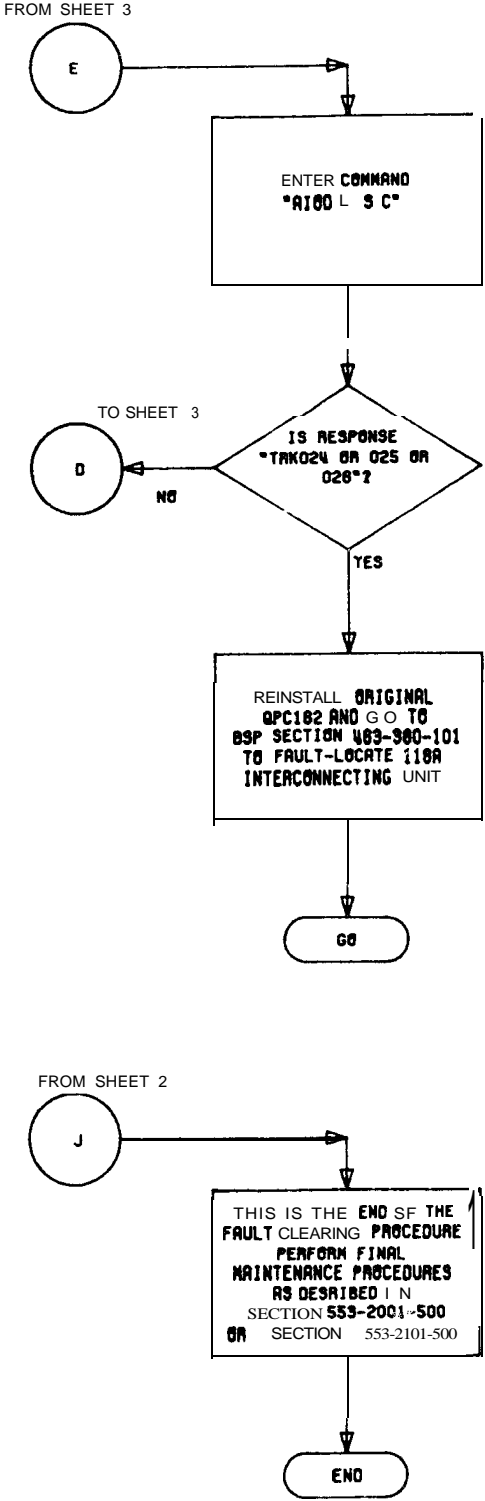
Flowchart 1 - AIOD Trunk Fault-Clearing Procedure



Flowchart 1 Continued - AIOD Trunk Fault-Clearing Procedure



Flowchart 1 Continued - AIOD Trunk Fault-Clearing Procedure



Flowchart 1 Continued - AIOD Trunk Fault-Clearing Procedure

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

CALL DETAIL RECORDING (CDR)
DESCRIPTION

CONTENTS	PAGE
1. GENERAL	1-1
2. DESCRIPTION	2-1

Reason for Reissue: This practice is reissued to create Appendices 1, 2 and 3 for SL-1 software Generics X08, X11 and X37, respectively, which describe the CDR feature as used with those generics.

* SL-1 is a trademark of Northern Telecom Ltd.

1. GENERAL

1.01 The Call Detail Recording (CDR) feature is an optional software package of the SL-1 Business Communications System which provides information on incoming and outgoing calls for accounting and administration purposes. The information is assembled by the SL-1 software and sent out via a Serial Data Interface (SDI) port on the SL-1 to any RS-232-C-compatible output or storage device. Examples of these are:

- TTY or printer output device
 - | Single/Multi-Port CDR storage system
 - | Mini-CDR storage system

1.02 This practice provides an overview of the CDR feature used in the SL-1 system. CDR can be used with all SL-1 software generic streams. Detailed information for CDR information processing for the three current generic streams of SL-1 are contained in the following appendices:

- | Generic X11 - Appendix 1 to 553-2631-100
- Generic X08 - Appendix 2 to 553-2631-100
- | Generic X37 - Appendix 3 to 553-2631-100

ASSOCIATED NTP

1.03 Detailed information on CDR storage systems is provided in the following documents:

Single/Multi-Port CDR Storage Systems.

- | 553-2631-110 - Description
- | 553-2631-210 - Installation
- | 553-2631-310 - Operation
- 553-2631-510 - Maintenance and Fault Clearing

Mini -CDR Storage System.

- | 553-2631-111 - Description
- 553-2631-211 - Installation
- 553-2631-311 - Operation and Maintenance



2. DESCRIPTION

RECORD GENERATION	<p>2.01 The CDR feature in its basic form comprises the CDR software which, if ordered, is delivered with the SL-1 program. This software contains the programs necessary to gather data and produce the call records.</p> <p>2.02 A call record is a sequence of data that describes an event in call processing, such as a call initiation, call transfer or conference, and call termination. For example, if an SL-1 user goes off-hook and dials a toll call (and the CDR feature is configured to record toll calls), a 'start' record is produced showing the time, date, identity of the user, digits dialed, and various other data pertaining to the call. When the user goes on-hook, an 'end' record is generated showing the time, date, identity of the user, etc. By matching the start' and 'end' records for the user, the duration of the call can be calculated, and various other data, such as the number called, can be determined by inspection.</p> <p>2.03 If calls are transferred or modified in some other way that would have an influence on costing or cost assignment, records for these events are also generated as the events take place. (Read the appropriate Appendix to this practice for more information on call modification in CDR.)</p>
RECORD OUTPUT AND STORAGE	<p>2.04 CDR records can be output on any RS-232-C-compatible device, such as a TTY or printer for hard-copy reports (see Fig. 2-1), or stored on tape for downstream processing at some future time.</p> <p>2.05 TTY or printer output uses the standard ASCII character set. In general, one CDR record takes up one line on a printed page, although certain records may require two or more lines. (See the appropriate appendix for details on output formats for the current generics. For previous generics, see Table 2-A.)</p>
PACKAGING	<p>2.06 The CDR software and hardware equipment packages to be ordered by a customer depend on the software generic and release to be used by the customer. Consult 553-2201-150 for packaging and ordering information.</p>
SINGLE/MULTI -PORT CDR TAPE STORAGE SYSTEM	<p>2.07 This hardware option consists of a single equipment cabinet (see Fig. 2-2) that contains a Central Processing Unit (CPU), a 9-track magnetic tape unit and tape control circuitry. The Multi-Port system is capable of storing call records from a maximum of 12 SL-1 systems, and the Single-Port system is restricted to one SL-1 system.</p> <p>2.08 The CDR system collects the call records from the SL-1 system(s), formats them into blocks, and stores the blocks on tape at 1600 bytes per inch in industry-standard Phase-Encoded format. When the tape is full, it is removed from the CDR tape drive and sent for processing into CDR reports for billing, administration, etc.</p> <p>2.09 The SDI cable between the SL-1 and the CDR cabinet can be up to 15.24 m (50 ft.) in length. Beyond this distance, modems are required.</p>

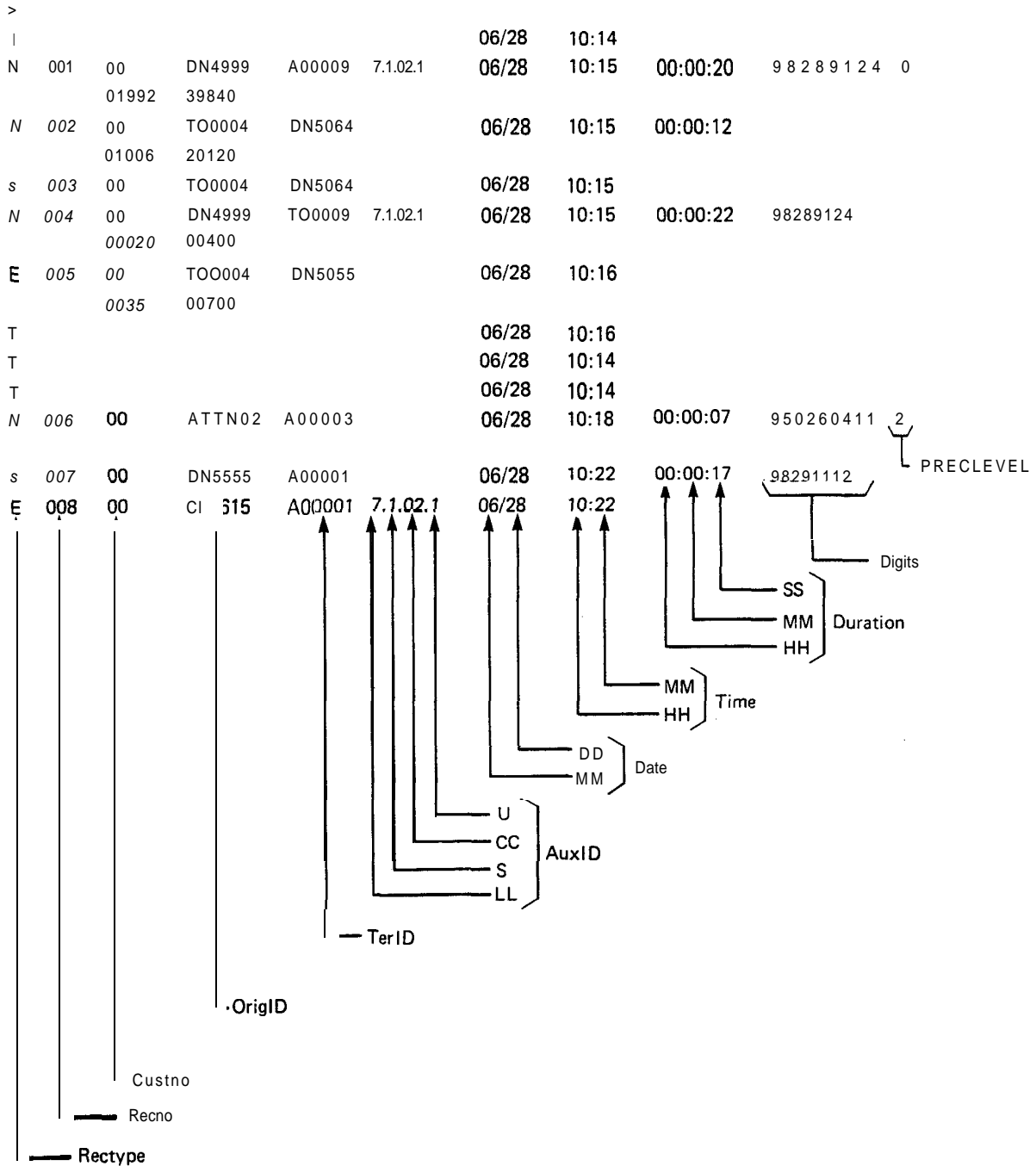


Fig. 2-1
Example of CDR TTY Printout

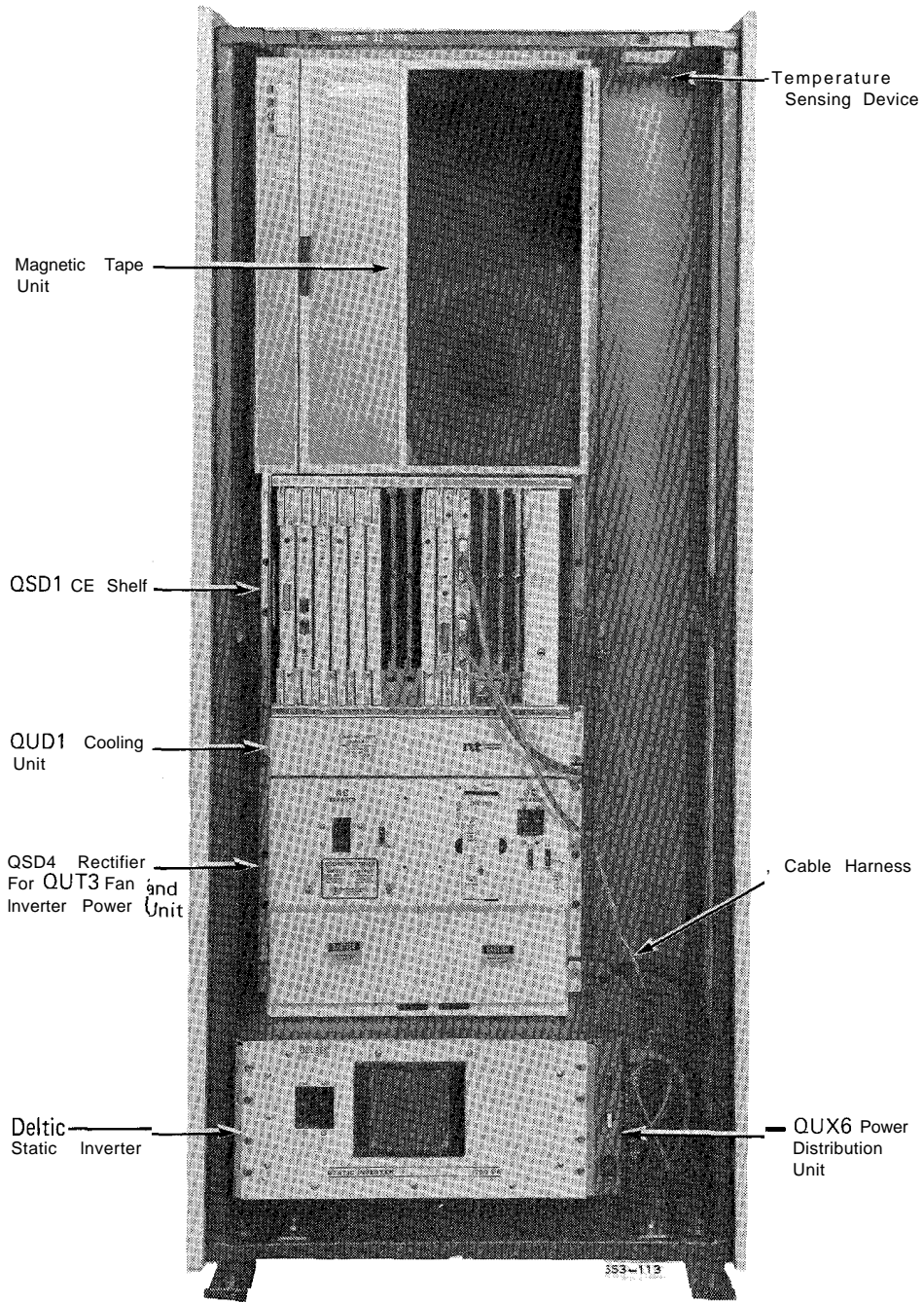


Fig. 2-2
Single/Multi -Port CDR Tape Storage System

Table 2-A
TTY OUTPUT FORMAT (GENERIC **X04**, **X05**, **X07** AND **X09**)

CHARAC - TER POSITION	NAME	FORMAT	DEFINITION
1	RECTYPE	Y	Record Type.
2	<blank>		Blank space
3-5	RECNO	XXX	Record Sequence Number
6	<blank>		Blank space
7-8	CUSTNO	XX	Customer Number
9	<blank>		Blank space
10-15	ORIGID	TRRMMM or DNXXXX or ATTNXX or CFLN	Originating Identification: Trunk Station Directory Number Attendant Number Conference Number
16	<blank>		Blank space
17-22	TERID	TRRMMM or DNXXXX or ATTNXX or CFLN	Terminating Identification: Trunk Station Directory Number Attendant Number Conference Number
23	<blank>		Blank space
24-29	AUXID	S.CC.U	Auxiliary Identification
30	<blank>		Blank space
31-41	TIMESTAMP	MM/DD_HH:MM	Timestamp
42	<blank>		Blank space
43-50	DURATION	HH:MM:SS	Duration
51	<blank>		Blank space
52-76	DIGITS	XXX...X or A_XXX...X	Digits Dialed: Up to 23 normal digits Route Selection was used
43-76	DURATION	XXX...X	14 Authorizaton code digits or 23 Charge Account digits or 23 Calling Party Number digits

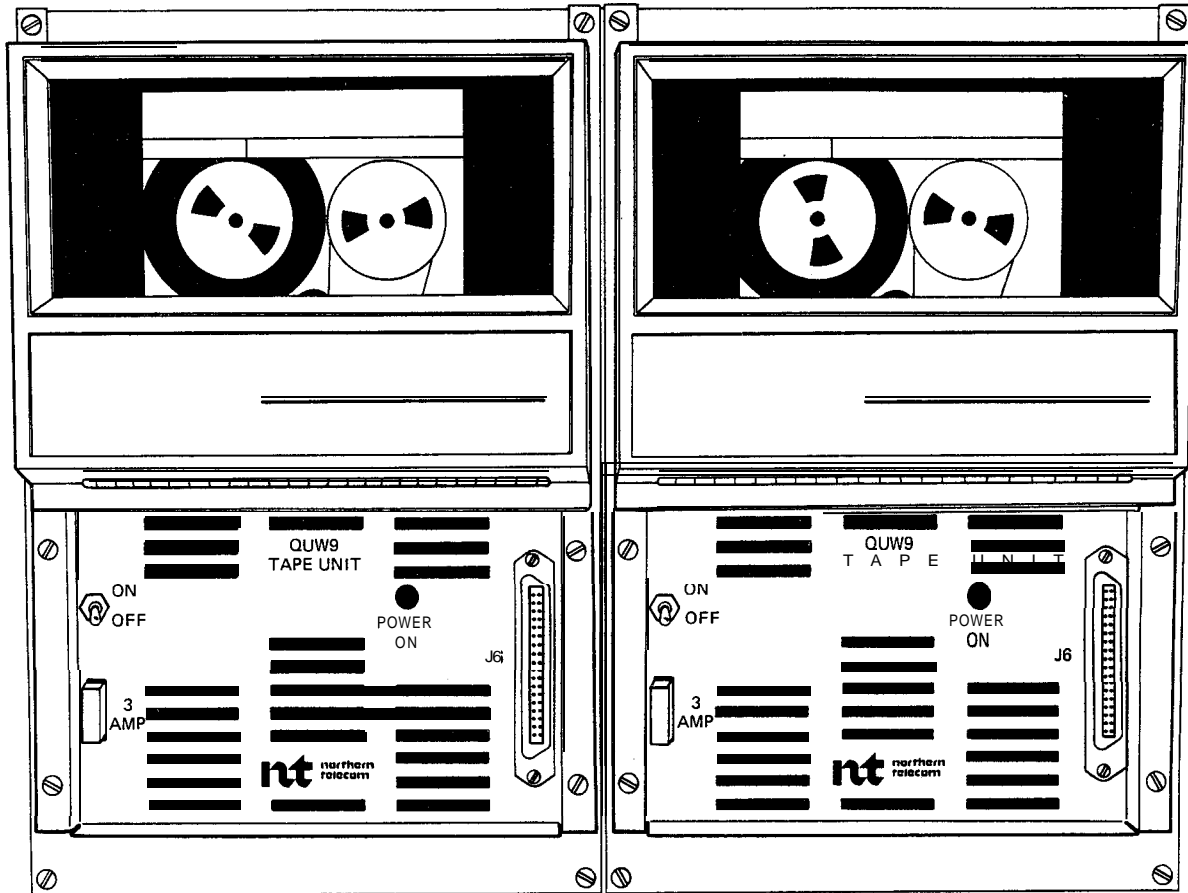


Fig. 2-3
Mini-CDR Tape Storage System

MINI-CDR TAPE STORAGE SYSTEM

2.10 This hardware option is used only in small systems (e.g., SL-1 MS) and consists of a second QUW9 Tape Unit (see Fig. 2-3) installed in the SL-1 CE cabinet. Call records are stored in system memory, then transferred to the Mini-CDR tape unit in block form. The stored records can then be polled from another device capable of storing or printing the data. The same SDI port is used to initiate polling and to transfer the polled information. Once begun, the polling operation must be allowed to complete, or the polling procedure must be repeated.

2.11 The Mini-CDR tape cartridges store information in a special format that is SL-1 standard, not industry standard. Therefore, these tapes can be read only by the SL-1 tape controller circuitry.



 INTEGRATED SERVICES NETWORK

MERIDIAN SL-1.

CALL DETAIL RECORDING

GENERIC X11

DESCRIPTION

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Reason for Reissue: This appendix is reissued to provide information on the Multi-Tenant Service feature.

* Meridian and SL-1 are trademarks of Northern Telecom Limited.

1. GENERAL

1.01 The Call Detail Recording (**CDR**) feature is an optional SL-1 software package which provides a record of selected calls for accounting and administration purposes. The basic CDR call records include the identity of the called and calling parties and the duration of the call. Additional records are generated when certain SL-1 features are used (e.g., conference, authorization codes) or by certain system events (e.g., initialization, change of system time clock).

1.02 The CDR call records are assembled by system software and sent to an EIA **RS-232-C-compatible** device such as a teletypewriter (TTY) or to one of the available CDR storage equipment packages.

1.03 This practice describes the operation of the CDR feature for Generic X11 up to Release 7. A general description of CDR and a list of associated NTP can be found in 553-2631-100.

2. FEATURES AND OPTIONS

BASIC CDR SOFTWARE PACKAGE

2.01 The basic CDR software package is required to generate the call records. This package is always required and has the following features:

- (a) **Multicustomer Operation.** Allows each customer within a single SL-1 system to individually select the CDR feature and its options. The feature is enabled or disabled on a customer basis in the Customer Data Block (overlay 15).
- (b) **Call Types.** Each customer may select for call recording any combination of the following call types on a trunk route basis:
 - | all incoming trunk calls
 - | all outgoing trunk calls
 - | all outgoing toll calls (toll calls are defined as those calls in which the first or second digit dialed after the trunk route access code is a digit 1 or 0)
- (c) **Multiple Appearance Directory Numbers (MADN).** As a customer option, a MADN may be supplemented by an auxiliary identification (AUXID) giving the shelf, card and unit of the originating set. An AUXID field is provided only if the originating set has a MADN as the prime DN. If a call originates from a MADN which is not the prime DN, the prime DN is shown as the originating ID with no AUXID field provided.
- (d) **Priority to CDR.** This option ensures that every call is recorded by having idle call registers selected for CDR before being used for call processing
- (e) **Flexible CDR Digit Suppression.** This option allows a customer to suppress a flexible number of dialed digits in the Digits field of CDR call records. The specified number of digits (0 to 32) are truncated from the end of the dialed digit string on both TTY and magnetic tape outputs.
- (f) **Connection Record.** Connection records (Release 3 and later) are used in the Integrated ACD/CDR Call Processing Interface feature to inform a customer-provided computer in real-time of which Automatic Call Distribution (ACD) agent is connected to which trunk. Connection records are generated on incoming calls on connection between an agent and a trunk and on each conference and transfer connection. The feature is implemented via a CDR-TTY (CTY) port which provides a data link to the customer-provided computer.
- (g) **Terminating Carriage Control.** A carriage return option can be implemented to inform a customer-provided computer that a CDR message is complete. An additional carriage return precedes every CDR record. A single CDR record consists of a carriage return, a line of data then another carriage return, resulting in every other record being a null record. This option is implemented through service change and, when implemented, applies to all CDR TTY ports in the system.

APPENDIX 1 TO 553-2631-100

OPTIONAL CDR
SOFTWARE PACKAGES

2.02 In addition to the basic CDR software package, the following optional packages are available (selected for the type of CDR output required):

- (a) CDR-TTY (**CTY**). This software package is selected when it is required to output call records on one or more RS-232-C-compatible devices. It provides a hard copy of the call records and may be used with the other CDR optional packages.
- (b) CDR Data Link (**CLNK**). This package is required when the Single/Multi-Port CDR storage system is used. The CLNK software formats the call records into a form suitable for storage on magnetic tape.
- (c) Mini-CDR (**MCDR**). This package is required when CDR is used on SL-1 M systems equipped with a second tape unit. The MCDR software sends call records to the Mini-CDR tape unit and allows the Mini-CDR tape unit to be polled from a TTY or storage device.
- (d) CDR with Charge Account (**CHG**). This package provides capability for direct billing of calls to specific charge account numbers.
- (e) CDR AUTOVON Enhancement (**ACDR**). This option records the precedence level of calls made in systems equipped with the AUTOVON feature.
- (f) Multi-Tenant Service (**XII Release 7+**). With the Multi-Tenant package the tenant numbers of the originating and terminating parties are included in CDR records.

CDR STORAGE
OPTIONS

2.03 Two CDR equipment options provide for the storage of call records on magnetic tape.

2.04 Single/Multi-Port CDR Storage System. This hardware option consists of a single equipment cabinet that contains a Central Processing Unit (CPU), a 9-track magnetic tape and tape control circuitry. The Multi-Port system is capable of storing call records from a maximum of 12 different SL-1 installations.

2.05 The CDR system collects the call records from the SL-1 machine(s), formats them into blocks and stores the blocks on tape. When the tape is full, it is removed from the CDR cabinet and sent for downstream processing into CDR reports, billing, etc.

2.06 Mini-CDR. This option is used only in small systems (e.g., the SL-1M) and consists of a second tape unit installed in the SL-1 system CE cabinet. Call records are stored in system memory and then transferred to the Mini-CDR tape unit in block form. The stored records can then be polled from a TTY or any other device capable of handling this data.

3. CALL RECORD OUTPUT

3.01 Call records may be output on TTY (Fig. 3-1) or on tape (Fig. 3-2).

- (a) TTY Output. Each call record is output on a TTY as a single line of data (or two lines if Multi-Tenant Service is enabled). The data is broken up into fields, the significance of which is determined by their position in the line. Fields are separated by one or more blank characters. The unused fields in a given record are left blank.
- (b) Tape Output. When call records are to be stored on magnetic tape, they are collected in the system memory and then transferred to the tape device as a block of 16 bit binary words via an **SDI** port. Null records are used to fill up blocks so that individual call records do not span tape blocks.

3.02 This practice describes the 9-track Single/Multi-Port tape format. **Mini-CDR** can only be read by the SL-1 Tape Controller and then output in the same format as TTY or 9-track tape.

3.03 For 9-track magnetic tape storage, a Tape Information Package (**P0572490**) is available. This package is used to facilitate the design of computer programs to read the tapes, decode the call records and produce accurate accounting reports.

RECORD CONTENTS

3.04 A call record on a TTY or printer contains the following fields except where noted

Record Type

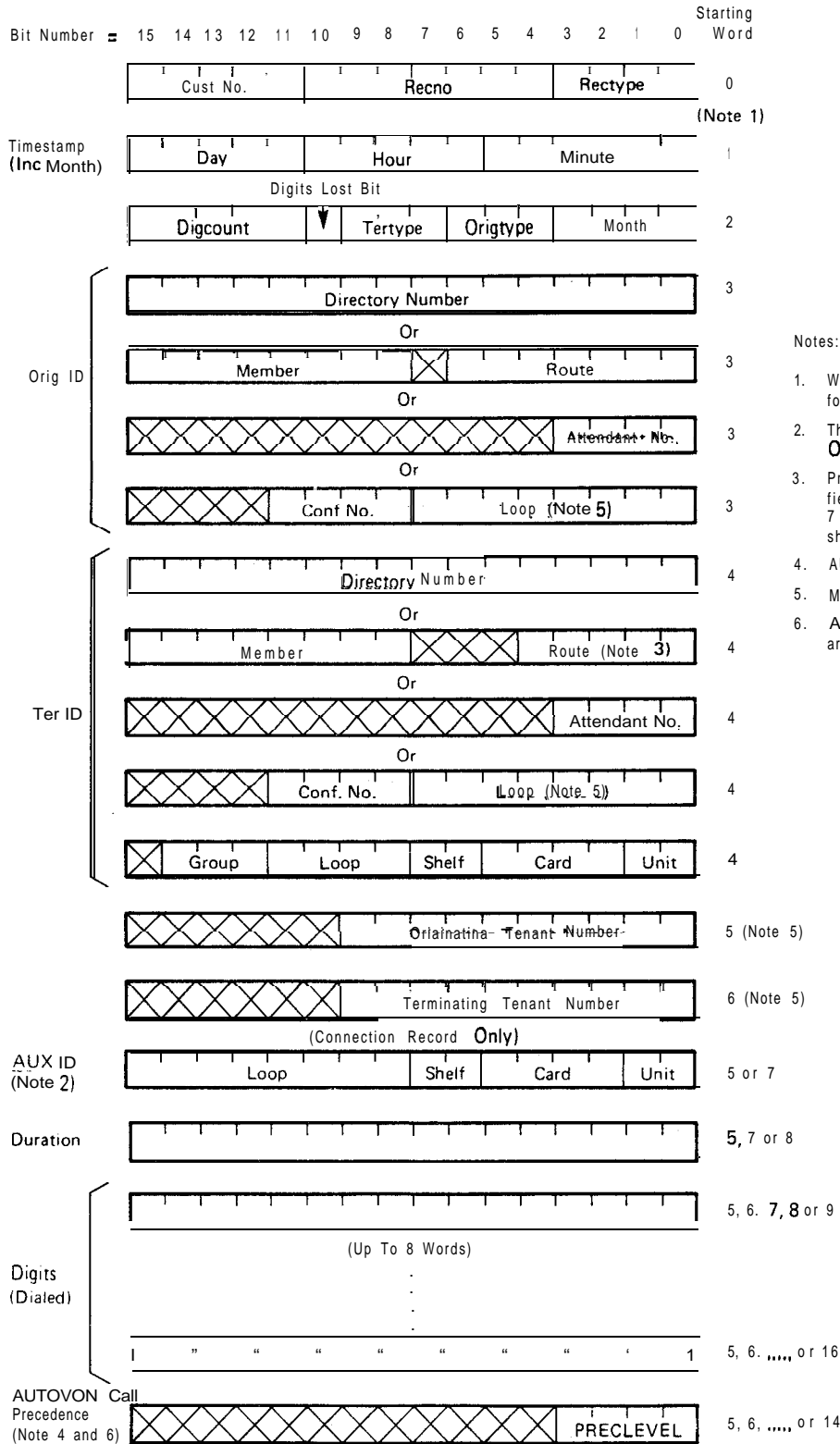
3.05 The Record Type (**RECTYPE**) field indicates the type of call record.

3.06 TTY Output. This field consists of a single letter identifying the type of record:

A	Authorization Code
C	Charge record
E	End record
F	Connection record (Conference connection)
I	Initialization record
M	Charge for Conference
N	Normal record
P	Calling Party Number
Q	Connection record (initial connection)
R	Connection record (Transfer connection)
S	Start Record
T	Timestamp record.

3.07 A **C** record is generated for a charge code entered either before establishing a call, while the call is established or when charge codes are entered prior to conferencing in each party. An **M** record is generated when a charge account code is entered after a conference is completed.

APPENDIX 1 TO 553-2631-100



Generic XI 1

Notes:

- Words 0, 1 and 2 define completely the format of the Balance of the Record.
- The **AuxID** is allocated only if either **Origtype** or **Tertype** requires it.
- Prior to Generic XI 1 Release 4, the loop field in the **OrigID** and **TerID** records is 7 bits. The **CONF** field is therefore shifted one bit right.
- AUTOVON feature must be equipped.
- Multi-Tenant feature must be equipped.
- AUTOVON-CDR and Multi-Tenant packages are mutually exclusive.

Fig. 3-2
Format of a Call Record on Tape

APPENDIX 1 TO 553-2631-100

3.06 Tape Output. This five-bit field can be 0-14 in binary and defines the type of record:

- 0 Null record
- 1 Normal record
- 2 Start record
- 3 End record
- 4 Charge record
- 5 Initialization record
- 6 Timestamp record
- 7 Normal record for RS-ANI, BARS, NARS or CDP
- 8 Start record for RS-ANI, NARS, BARS and CDP
- 9 Calling Party Number (**CPN**)
- 10 Authorization Code
- 11 Charge for Conference
- 12 Normal record for a call on an expensive route (**MARS/BARS**)
- 13 Start record for a call on an expensive route (**NARS/BARS**)
- 14 Connection record.

Record Number

3.09 The Record Number (**RECNO**) field identifies the number of the current record in the CDR sequence.

3.10 TTY Output. This three-character field gives the record sequence number, modulo 128, right-justified.

3.11 Tape Output. This seven-bit field gives the record number in binary, modulo 128.

Customer Number

3.12 The Customer Number (**CUSTNO**) field identifies the customer associated with the call.

3.13 TTY Output. A two-character field identifies the customer number (0-31).

3.14 Tape Output. A five-bit field identifies the customer (0-31 in binary).

Originator Type (Tape Only)

3.15 The three-bit Originator Type (**ORIGTYPE**) field identifies the type of apparatus that originated the call:

- 0 500/2500 set or SL-1 station
- 1 500/2500 set or SL-1 station with which AUXID is associated
- 2 Trunk

- 3 Attendant
- 4 Trunks, END record expected
- 5 Conference.

Terminator Type (Tape Only)

3.16 The Terminator Type (TERTYPE) field identifies the type of apparatus to which the call was terminated. Definition of fields is the same as for ORIGTYPE.

Originator Identification

3.17 The Originator Identification (**ORIGID**) field identifies the apparatus that originated the call.

3.16 TTY Output. The format depends on the type of equipment as follows:

- 1 Stations. Stations are identified in the form **DNxxxx**, where xxxx is the prime DN for the station.
- 1 Trunks. The **ORIGID** field is in the format **Trrrmmm**, where rrr is the route number and mmm is the member number.
- 1 Attendants. Attendants are identified in the form **ATTNxx**, where xx is the console number (1-15).
- 1 Conference Loops. Conference loops are identified in the form **CFillnn**, where 111 is the loop number (0-159, 0-79 prior to Release 4) and nn is the conference number.

3.19 Tape Output. The format of the 16-bit field depends on the type of equipment as follows:

- 1 Stations. If the **ORIGTYPE** or **TERTYPE** fields are 0 or 1 (i.e., a station), this field identifies the **500/2500** set DN or SL-1 prime DN of the set that originated the call. Up to four decimal digits are stored in a binary field of 16 bits.

zero is represented by binary 10 (i.e., 1010)

* is represented by binary 11

is represented by binary 12.

- 1 Trunks. If the **ORIGTYPE** or **TERTYPE** fields are 2 or 4 (i.e., the originator is a trunk), this field is 15 bits long.

- The trunk route number is 7 bits.

- 1 The trunk route member number is 8 bits.

- 1 Attendants. If the **ORIGTYPE** or **TERTYPE** field is 3 (i.e., an attendant), attendant number is identified in 4 bits.

- 1 Conference Loops. If the **ORIGTYPE** or **TERTYPE** field is 5 (i.e., conference), loop and conference number are identified in 12 bits:

8 bits - loop number (7 bits prior to Release 4)

4 bits - conference number.

APPENDIX 1 TO 553-2631-100

Terminator Identif ication	<p>3.20 The Terminator Identification (TERID) field identifies the apparatus on which a call is terminated. Contents are the same as for ORIGID, with the addition of Connection records.</p> <p>3.21 Connection Records. Connection records are identified in the form TNxxxx, where xxxx is the loop, shelf, card and unit in packed format.</p>
Digit Count (Tape Only)	3.22 This field gives the total number of digits dialed in 5 bits,
Digits Lost (Tape Only)	3.23 This field consists of one bit, which is set to 1 if the total digits dialed exceeds 32.
Auxiliary Identif ication	<p>3.24 The Auxiliary Identification (AUXID) field identifies the TN of the originating station when the station has a multiple appearance prime DN, and thus cannot be uniquely identified by ORIGID.</p> <p>3.25 TTY Output. The AUXID field is in the form ll.s.cc.u (X11 Releases 1 to 3) or lll.s.cc.uu (X11 Release 4) and uniquely identifies originating station loop, shelf, card and unit.</p> <p>3.26 Tape Output. This field (16 bits) depends on the card density of the originating station.</p> <ul style="list-style-type: none"> • Single Density - XLLLLLLLSSCCCUU (X = not used) • Double Density - XLLLLLLLSUCCCUU
Timestamp	<p>3.27 The TIMESTAMP field gives the date and time. Its exact definition depends on the type of record:</p> <p>Normal record - end of call</p> <p>Start record - start of call</p> <p>End record - end of call</p> <p>Initialization record - time of system initialization</p> <p>Timestamp - old or new timestamp</p> <p>Transfer record - Completion of transfer</p> <p>Connection record - Time of connection.</p> <p>3.28 If the call started on the last day of a month and stopped on the first day of the next month then the start day is shown as day 0 of the next month.</p> <p>3.29 TTY Output. The format is MM/DD HH:MM where:</p> <p>MM is month (1-12)</p> <p>DD is day (1-31)</p> <p>HH is hour (0-23)</p> <p>MM is minute (0-59)</p>

3.30 Tape Output. The timestamp field occupies 20 bits as follows:

- | MONTH (4 bits) where Jan. = 1 and Dec. = 12
- | DAY (5 bits) where first day of month = 1
- HOUR (5 bits) which can be 0 to 23
- MINUTE (6 bits) which can be 0 to 59.

Call Duration

3.31 The DURATION field measures the call duration in units of 2 s. In a Normal record, this is the duration of the call from start to disconnect. In a Start record, this is the duration of the call from start to first feature usage at time given by TIMESTAMP. An incoming call is deemed to start at the time of presentation to the called party. An outgoing call starts at the time of trunk seizure (dial tone removed).

3.32 TTY Output. The output is in the form HH:MM:SS where:

HH is hour (0-23)

MM is minutes (0-59)

SS is seconds (0-59)

3.33 Start records on TTY output do not contain a duration field.

3.34 Tape Output. DURATION takes up 16 bits, and gives a binary count of the call duration in seconds.

Digits

3.35 The DIGITS field identifies the digits dialed, digits outpulsed, or Charge Account code entered. Up to 32 digits can be recorded. If more digits are dialed, only the first 16 are recorded correctly. Digit 33 and succeeding digits are cycled over digits 17 to 32.

3.36 Route Selection. Route Selection digits (for NARS, BARS, etc.) are indicated if preceded by the letter A. For RS-AN1 or NARS, the digits shown in the digits field are those that are actually outpulsed by the system after route selection and digit manipulation. With BARS and CDP, the digits shown are the actual digits dialed except that the BARS access code and CDP steering code is replaced by the trunk access code. BARS/NARS calls placed over an expensive route are indicated by the letter E.

3.37 Dictation and Paging Trunks. The CDR printout for dictation and paging trunks accessed by a 2500-type set specifies only the trunk access code in the DIGITS field. Dictation trunks require tones to instruct the machines at the other end. The 2500-type set sends these tones directly to the dictation trunk without using a call register to store digits. As these digits are not stored in a call register, the CDR cannot print them out. In the case of 500-type and SL-1 sets, the digits must be stored in a call register and then a DTR translates the digits into tones. The CDR output will therefore show all digits dialed by these sets.

3.38 TTY Output. Up to 32 digits can be output on the terminal.

3.39 Tape Output. Each digit is stored as a four-bit word. Total digit storage is a maximum of eight 16-bit words.

Precedence Level	<p>3.40 The Precedence Level field for AUTOVON (PRECLEVEL) field is output only if the AUTOVON feature package is equipped. It consists of a single digit that represents the precedence level of an AUTOVON call as follows:</p> <ul style="list-style-type: none"> 0 AUTOVON Flash Override 1 AUTOVON Flash 2 AUTOVON Immediate 3 AUTOVON Priority 4 AUTOVON Routine 5 Non-AUTOVON calls <p>3.41 The AUTOVON precedence level is appended to all AUTOVON call records except the time stamp and initialize records.</p> <p>3.42 TTY Output. PRECLEVEL is a single digit appended to the end of call records. Levels 4 and 5 (AUTOVON routine and non-AUTOVON calls) are not output on TTY.</p> <p>3.43 Tape Output. PRECLEVEL occupies a 4-bit field which gives the precedence level (0-5) in binary.</p>
<p>↗ Originating/ Terminating Tenant Number</p>	<p>3.44 When the Multi-Tenant Service feature is present, CDR formats will include the originating and terminating Tenant Numbers.</p> <p>3.45 TTY Output. A second line of data is printed. This consists of the Originating and Terminating Tenant Numbers. The fields are located directly under the Originating and Terminating ID fields of the CDR record.</p>
<p>↳ Terminating Carriage Return (TTY Only)</p>	<p>3.46 Tape Output. The Originating and Terminating Tenant Numbers each occupy a 10 bit binary field (range 0 - 512).</p> <p>3.47 As a customer option, the SL-1 can output a Line Feed (carriage return) character at the end of each call record.</p>
<p>↗ RECORD LENGTH FOR TTY</p>	<p>3.48 The following formulae permit the calculation of lengths of all TTY records in number of characters. (See Tables 3-A for a definition of call record fields for Generic X11 Releases 1, 2 and 3, and Table 3-B for the fields for Release 4.)</p> <p>NORMAL RECORD = C + 44 + T + N + A + M</p> <p>START RECORD = C + 44 + T + N + A + M</p> <p>END RECORD = C + 35 + T + A + M</p> <p>INITIALIZATION RECORD = C + 35 + T</p> <p>TIMESTAMP RECORD = C + 35 + T</p>
<p>↳ AUTHORIZATION CODE RECORD</p>	<p>AUTHORIZATION CODE RECORD = C + 35 + T + N + A + M</p>

$$\text{CHARGE RECORD} = C + 35 + T + N + A + M$$

$$\text{CHARGE-CONFERENCE RECORD} = C + 35 + T + N + A + M$$

$$\text{CPN RECORD} = C + 35 + T + N + A + M$$

$$\text{CONNECTION RECORD} = C + 44 + T + N + A + M$$

3.49 The definition of the variables are as follows:

C is the number of ASCII control characters (CR, LF, NUL) per record, typically 8 (9 if AUTOVON precedence level is printed out or 23 if Multi-Tenant package is equipped).

T is 0 without AUXID; up to 12 with AUXID.

N is the number of digits to be recorded.

A is 1 if AUTOVON is equipped and call has a precedence level of 0 to 3).

M is 6 if Multi-Tenant package is equipped.

RECORD LENGTH FOR TAPE

3.50 To separate call records within a tape block, the length of each record must be known. The following formulae permit the calculation of the lengths of all call records in number of words (16 bits):

$$\text{NORMAL RECORD} = 8 + T + \text{CEILING}(N/4) + A + M$$

$$\text{START RECORD} = 8 + T + \text{CEILING}(N/4) + A + M$$

$$\text{END RECORD} = 7 + T + A + M$$

$$\text{INITIALIZATION RECORD} = 3$$

$$\text{TIMESTAMP RECORD} = 3$$

$$\text{AUTHORIZATION CODE RECORD} = 7 + T + \text{CEILING}(N/4) + A + M$$

$$\text{CHARGE RECORD} = 7 + T + \text{CEILING}(N/4) + A + M$$

$$\text{CHARGE-CONFERENCE RECORD} = 7 + T + \text{CEILING}(N/4) + A + M$$

$$\text{CPN RECORD} = 7 + T + \text{CEILING}(N/4) + A + M$$

$$\text{CONNECTION RECORD} = 6 + T + \text{CEILING}(N/4) + A + M$$

3.51 The definition of the variables are as follows:

N is the number of digits to be recorded.

T is 1 if AUXID is included, 0 if not.

CEILING (N/4) is the smallest integer greater than or equal to $N/4$ (e.g., if $N = 29$, Ceiling = 8).

A is 1 if AUTOVON is equipped.



M is 2 if Multi-Tenant feature is equipped.

Table 3-A
TTY OUTPUT FORMAT (GENERIC X11 RELEASES 1, 2 AND 3)

CHARACTER POSITION	NAME	FORMAT	DEFINITION
1	RECTYPE	Y	Record Type.
2	<blank>		Blank space
3-5	RECNO	XXX	Record Sequence Number
6	<blank>		Blank space
7-8	CUSTNO	x x	Customer Number
9	<blank>		Blank space
10-16	ORIGID	TRRRMMM or DNXXXX_ or ATTNXX_ or CFLN_	Originating Identification: Trunk Station Directory Number Attendant Number Conference Number
17	<blank>		Blank space
18-24	TERID	TRRRMMM or DNXXXX_ or ATTNXX_ or CFLN_	Terminating Identification: Trunk Station Directory Number Attendant Number Conference Number
25	<blank>		Blank space
26-34	AUXID	LL.S.CC.U	Auxiliary Identification
35	<blank>		Blank space
36-46	TIMESTAMP	MM/DD_HH:MM	Timestamp
47	<blank>		Blank space
48-55	DURATION	HH: MM:SS	Duration
56	<blank>		Blank space
57-90	DIGITS	XXX...X or A_XXX...X E_XXX...X	Digits Dialed: Up to 32 normal digits Route Selection was used BARS/NARS call placed over expensive route
48-90	DURATION	XXX...X	14 Authorizaton code digits or 23 Charge Account digits or 23 Calling Party Number digits

Table 3-B
TTY OUTPUT FORMAT (GENERIC X11 RELEASE 4)

CHARACTER POSITION	NAME	FORMAT	DEFINITION
1	RECTYPE	Y	Record Type.
2	<blank>		Blank space
3-5	RECNO	XXX	Record Sequence Number
6	<blank>		Blank space
7-8	CUSTNO	x x	Customer Number
9	<blank>		Blank space
10-16	ORIGID	TRRRMMM or DNXXXX_ or ATTNXX_ or CFLLLN	Originating Identification: Trunk Station Directory Number Attendant Number Conference Number
17	<blank>		Blank space
18-24	TERID	TRRRMMM or DNXXXX_ or ATTNXX_ or CFLLLN	Terminating Identification: Trunk Station Directory Number Attendant Number Conference Number
25	<blank>		Blank space
26-36	AUXID	LLL.S.CC.UU	Auxiliary Identification
37	<blank>		Blank space
38-48	TIMESTAMP	MM/DD_HH:MM	Timestamp
49	<blank>		Blank space
50-57	DURATION	HH:MM:SS	Duration
58	<blank>		Blank space
59-92	DIGITS	XXX...X or A_XXX...X E_XXX...X	Digits Dialed: Up to 32 normal digits Route Selection was used BARS/NARS call placed over expensive route
93	<blank>		Blank space
94	PRECLEVEL	X	AUTOVON Precedence Level

Table 3-B Continued
TTY OUTPUT FORMAT (GENERIC X1 RELEASE 4)

CHARACTER POSITION	NAME	FORMAT	DEFINITION
50-94	DURATION	XXX...X	14 Authorizaton code digits or 23 Charge Account digits or 23 Calling Party Number digits

Table 3-C
TTY INPUT FORMAT (GENERIC X11 RELEASE 7+)

Note 1: This format is used for F, N, Q, R and S records.

Note 2: This format is used for A, M and P records.

CHARACTER POSITION	NAME	FORMAT	DEFINITION
LINE ONE			
1	RECTYPE	Y	Record Type.
2	<blank>		Blank space
3-5	RECNO	XXX	Record Sequence Number
6	<blank>		Blank space
7-8	CUSTNO	XX	Customer Number
9	<blank>		Blank space
10-16	ORIGID	TRRRMMM or DNXXXX_ or ATTNXX_ or CFLLLN	Originating Identification: Trunk Station Directory Number Attendant Number Conference Number
17	<blank>		Blank space
18-24	TERID	TRRRMMM or DNXXXX_ or ATTNXX_ or CFLLLN	Terminating Identification: Trunk Station Directory Number Attendant Number, Conference Number
25	<blank>		Blank space
26-36	AUXID	LLL.S.CC.UU	Auxiliary Identification
37	<blank>		Blank space
38-48	TIMESTAMP	MM/DD_HH:MM	Timestamp
49	<blank>		Blank space
50-57	DURATION	HH: MM:SS	Duration (Note 1 in preamble)
58	<blank>		Blank space
59-92	DIGITS	XXX...X or A_XXX...X E_XXX...X	Digits Dialed: (Note 1 in preamble) Up to 32 normal digits Route Selection was used BARS/NARS call placed over expensive route



Table 3-C Continued
 TTY INPUT FORMAT (GENERIC X11 RELEASE 7+)

CHARACTER POSITION	NAME	FORMAT	DEFINITION
93	<blank>		Blank space
94	PRECLEVEL	X	AUTOVON Precedence Level
5 0 - 9 4	DURATION	XXX...X	14 Authorizaton code digits or 23 Charge Account digits or 23 Calling Party Number digits (Note 2 in preamble)
LINE TWO			
1-9	<blank>		Blank spaces
10-12	ORIGTEN0	XXX	Originating Tenant Number
13-17	<blank>		Blank spaces
18-20	TERTENO	XXX	Terminating Tenant Number

4. CALL RECORD GENERATION

4.01 A simple call generates a single call record. Calls that are modified because of certain features (e.g., call transfer, conference) generate multiple records.

4.02 For every call to be recorded, the system software generates one Normal record or several Start and End records. Additional record types are generated to accommodate certain features.

4.03 The different types of call records and their application are outlined as follows.

Normal Record (N)

4.04 A Normal record is generated when a simple call is established (whether extended or not through the attendant console) and when no other set feature is activated. The Normal record contains the following information:

RECTYPE
 RECNO
CUSTNO
 ORIGID
 TERID
 AUXID (optional)
 TIMESTAMP
 DURATION
 DIGITS
 PRECLEVEL (optional, AUTOVON must be equipped)
 ORIGTEN0 (optional, Multi-Tenant must be equipped) ←
 TERTENO (optional, Multi-Tenant must be equipped). ←

4.05 For a Normal record, all telephone set dial pad input is included in the CDR record until such time as the End-of-Dialing (EOD) timer interval is exceeded or the user enters # from the set keyboard. Thus, the dialed DN portion of the CDR record may include * symbols and unused digits (e.g., in many cases speed call and **autodial** numbers include * symbols, each of which when interpreted by the system causes a 3 s pause to occur). A user may enter useless digits prior to the system receiving an EOD timeout or user-initiated # entry. In such cases, the call is completed to the correct destination but the CDR record contains the useless digits.

4.06 When one of the Route Selection features are used, the letter A precedes the Digits field on TTY outputs. These features are Basic Alternate Route Selection (BARS), Network Alternate Route Selection (NARS), Route Selection-Automatic Number Identification (RS-ANI) and Coordinated Dialing Plan (CDP). In BARS and NARS features, the letter E precedes the TTY Digits field to indicate the call was completed over facilities that are designated (through service change) as expensive. The actual digits that appear in the Normal (or Start) records depend on the Route Selection feature as follows:

- 1 RS-ANI, NARS ~ digits shown are those that are actually outputted by the system after route selection and any required digit manipulation. The actual digits dialed are reflected only if no digit manipulation was required to process the call.

. BARS, CDP - digits shown are those that are dialed with the exception of the BARS access code or CDP steering code which are replaced with the trunk access code.

4.07 Incoming Calls. An incoming call answered by the attendant and extended to a station generates a Normal record, indicating the trunk as the ORIGID and the station as the TERID. No indication is given that the attendant is involved in the call. When the calling party abandons the call before the station answers, and before recall is activated, no record is generated. However, when the call is abandoned during recall, the attendant console is shown as the TERID. Incoming calls answered by the attendant but not extended to a station are shown as terminating at the attendant console.

4.08 Outgoing Calls. An outgoing call placed by the attendant generates a Normal record, indicating the attendant console as the ORIGID and the trunk as the TERID. However, when the call is extended to a station, the Normal record generated indicates the trunk as ORIGID and the station as the TERID. The Digits field includes the station number dialed by the attendant.

4.09 Ring Again. When the Ring Again feature is activated, a record is generated only when a trunk is seized.

4.10 TIE Trunk Operation. Calls placed over tandem TIE trunks are billed from the time the call is answered (i.e., answer supervision is received), rather than from the time the trunk is seized. A Normal (or Start) record is generated only if answer supervision is received. Thus the calling party is not charged for the time taken for outpulsing and ringing.

Start Record (S)

4.11 A Start record is generated when a call receives treatment from certain features of the SL-1 system. A Start record contains the following information:

RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
DURATION (not included in TTY records)
DIGITS
PRECLEVEL (optional, AUTOVON must be equipped)
ORIGTENO (optional, Multi-Tenant must be equipped)
TERTENO (optional, Multi-Tenant must be equipped).

→
→

4.12 Call Transfer. When the Call Transfer feature is activated on an established call, a Start record is generated instead of a Normal record. The record is generated when the transfer is completed and shows the two involved parties immediately before the feature was activated. On termination of the call, an End record is generated showing its final disposition. Start records are not generated for intermediate stations when a call is transferred more than once; CDR records, therefore, do not indicate this occurrence.

4.13 Conference. When the Conference feature is activated on an established call, a Start record is generated as described for the Call Transfer feature. A subsequent Start record is generated for each CDR trunk included in the conference. The Duration field, which appears on CDR tape but not in TTY output, is calculated from the previous Start record timestamp to the current Start record timestamp. Although the sequence of related Start records generated may be altered by the CDR processing routines, the chronological (timestamp) data remains intact and each Start record is generated before its corresponding End record. The End record shows the conference bridge as the ORIGID and the conferenced trunk as TERID.

4.14 Call Forward. When the Call Forward feature is activated and results in a trunk-to-trunk (tandem) call, a consecutive pair of Start records are generated. The first record indicates the incoming trunk as ORIGID and the call forward DN as TERID. The second record indicates the call forwarded DN as the ORIGID and the outgoing trunk as the TERID. Both records indicate the same timestamp and duration data. An End record is generated at the end of the call.

4.15 Other Features. When barge-in, busy verification, privacy release or override is applied to an established call, a Start record is generated. Calls that would normally generate a Normal record are altered to generate a Start record. The record indicates that a feature was activated during the call, as well as any changes to the parties involved. The trunk ID remains consistent throughout. An End record is generated on termination of the call.

End Record (E)

4.16 Each End record is associated with a specific Start record and is generated on termination of the call. The record shows the final disposition of the call. The timestamps on the corresponding Start and End records can be used to calculate the Duration of a call. The End record contains the following information:

```
RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
'PRECLEVEL (optional, AUTOVON must be equipped)
ORIGTENO (optional, Multi-Tenant must be equipped)
TERTENO (optional, Multi-Tenant must be equipped).
```

initialization Record (I)

4.17 After a system initialization, a single record is generated to note this occurrence and its time in the following format:

```
RECTYPE
TIMESTAMP
```

Timestamp Record (T)

4.18 When the system time or date is changed from either the attendant console or a TTY, a consecutive pair of records is generated specifying the old and new timestamps. The first record in such a pair gives the old timestamp; the second record contains the updated timestamp. Both records have the following format:

```
RECTYPE
TIMESTAMP
```

Authorization Code Record **(A)**

4.19 Authorization Code recording is optional and is set using overlay 24. A record is generated when the code is entered and one of the following occurs:

- a trunk is seized
- a local set answers Direct Inward System Access (**DISA**) calls
- a DISA call cannot be connected to a local set
- Ring Again is activated, in which case both the ORIGID and TERID fields are the DN associated with the station entering the authorization code.

4.20 When authorization codes are stored as auto dial or speed call entries, the number stored must contain the access code followed by the authorization code only. All digits after the access code are interpreted as authorization code digits. This record contains the following information:

RECTYPE
 RECNO
 CUSTNO
 ORIGID
 TERID
 AUXID (optional)
 TIMESTAMP
 DIGITS
 PRECLEVEL (optional, AUTOVON must be equipped)
 ORIGTENO (optional, Multi-Tenant must be equipped)
 TERTENO (optional, Multi-Tenant must be equipped).

→
 →

Charge Account Record **(C)**

4.21 The charge account record is designed to allow direct billing of calls to specific charge account numbers. Charge account number lengths (2 to 23 digits) are defined individually for each customer, by using overlay 15. A charge account number, when entered either before dialing or during an established incoming or outgoing call, causes a Charge Account record to be generated which contains the following information:

RECTYPE
 RECNO
 CUSTNO
 ORIGID
 TERID
 AUXID (optional)
 TIMESTAMP
 DIGITS
 PRECLEVEL (optional, AUTOVON must be equipped)
 ORIGTENO (optional, Multi-Tenant must be equipped)
 TERTENO (optional, Multi-Tenant must be equipped).

→
 →

4.22 Numbers of Fixed Length. The system assumes that a charge account number is valid when the number of digits entered corresponds to the account number length as defined in the customer data block.

- (a) When a charge account number is entered before establishing a call and too few digits are entered, the system waits 30 s (15 s for 2500-type sets) for further input. When no further digits are entered, overflow tone is provided for 15 s, then the set is locked out by the system. A Charge record is generated showing the partially entered charge account number.
- (b) When a charge account number is entered during an established call and too few digits are entered, no response is given until the interdigit timeout occurs. Then overflow tone is provided for 15 s. After this time, the previously established call will be reconnected. On 500/2500-type sets, if the user does not wait for a response and has dialed too few digits, then each switchhook flash is interpreted as a digit 1 until the charge account length is reached. Dial tone is then returned and the next switchhook flash will reconnect the call. On SL-1 sets without a charge key, if the user does not wait for a response and has dialed too few digits, then the call is reestablished when the DN key is depressed. However, no charge record will be produced.

4.23 Numbers of Variable Length. SL-1 sets and attendant consoles equipped with a charge (**CHG**) key are permitted to enter charge account numbers containing less digits than defined in the customer data block. The Charge account number entered is accepted by the system when the CHG key is operated. Charge account number entry is also accepted by the system by operating a DN, call transfer or conference key that was active before the CHG key was operated. The CHG key may also be used to terminate an entry to correct an error or to enter multiple charge account numbers.

4.24 Deletion of Number. A charge account record is not output by the system unless the call involves a trunk and meets the criteria set for CDR in the route data block.

4.25 Call Transfer. A call transferred from one set (**A**) to another set (**B**) generates a Start record for set A and an End record for set B when the call is terminated. However, if set B, instead of terminating the call, enters charge account number and subsequently transfers the call to another set (**C**), a Charge record is generated for set B. The resulting Start and End records do not indicate set B or any other intermediate set as being involved in the call. The Start, Charge and End records can only be associated on a timestamp and trunk basis, the trunk remaining consistent on all three records.

Charge Conference Record **(M)**

4.26 Charge Conference records allow the assignment of one or more charge account numbers to one, several or all members of a conference. Individual Start, Charge Conference and End records are generated for each individual trunk participating in a conference call. Individual End records are generated for each trunk as it disconnects from the conference. The Charge Conference record contains the following information:

RECTYPE
 RECNO
 CUSTNO
 ORIGID
 TERID
 AUXID (optional)
 TIMESTAMP
 DIGITS
 PRECLEVEL (optional, AUTOVON must be equipped)
 ORIGTENO (optional, Multi-Tenant must be equipped)
 TERTENO (optional, Multi-Tenant must be equipped).

→
 →

4.27 Different Account Numbers. To assign portions of a conference call to different charge account numbers, the account numbers must be entered when adding a trunk to a conference and before conferencing is completed. For-500/2500-type sets, the account number is entered after the switchhook flash and before the trunk is dialed. For SL-1 sets, the account number is entered after the conference key is operated, either before or after the trunk is dialed and before the conference key is operated a second time. The Charge Conference record generated shows the set performing the entry, the trunk added and the charge account number. A charge account number is entered for each trunk added to the conference.

4.28 Single Account Numbers. When all participants in a conference call are assigned the same charge account number, only one entry is required. Once all the trunks are included in the conference, the charge account number is entered in the usual manner. Separate Charge Conference records are generated for each trunk in the conference call.

Calling Party Number **(CPN) Record (P)**

4.29 The CPN record is useful in matching telephone company billing records of collect calls against call detail records. By operating a CPN key, which may be assigned to attendant consoles and SL-1 sets, and entering the calling party number (1 to 23 digits), a CPN record is generated. A CPN record is generated each time the CPN key is depressed, permitting the creation of multiple CPN records. The CPN record contains the following information:

RECTYPE
 RECNO
 CUSTNO
 ORIGID
 TERID
 AUXID (optional)
 TIMESTAMP
 DIGITS
 PRECLEVEL (optional, AUTOVON must be equipped)
 ORIGTENO (optional, Multi-Tenant must be equipped)
 TERTENO (optional, Multi-Tenant must be equipped).

→
 →

Connection Record

4.30 There are three types of Connection record (Release 3 and later):

- (a) **Q Record.** Generated when a connection is made between a trunk and an ACD agent.
- (b) **R Record.** Generated when a call is transferred by an ACD agent.
- (c) **F Record.** Generated when a conference is set up by an ACD agent.

4.31 The appropriate Connection record is generated for each connection made with an ACD agent linked to a customer-provided computer. The Connection record allows agents to establish and modify a profile of a call in progress (e.g., the caller's name, address or other information) which can be passed on with the call each time the call is transferred or conferenced. This profile is entered in the customer-provided computer and can be displayed on ACD agent terminals. If the call is transferred or conferenced to another agent, the call profile is output on the second agent's terminal. The second agent can then change the call profile as required. Further transferring or conferencing is treated in the same manner. When the customer-provided computer detects a connection record, it scans the CDR records for other connection records related to that call. Any previously entered call profiles will be displayed on the ACD agent's screen. This procedure is repeated on all subsequent transfers and conferences until the call is terminated by an agent or transferred to a non-agent.

4.32 The Q record may be used in Voice Response Unit (VRU) applications (e.g., 411 and 555-1212 calls) to inform the customer-provided computer as to which trunk to connect the VRU to play out an electronically constructed message to the caller. The agent's work time for the call is therefore reduced.

4.33 Connection records contain the following information:

```

RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
DURATION
DIGITS
PRECLEVEL (optional, AUTOVON must be equipped)
ORIGTENO (optional, Multi-Tenant must be equipped) ←
TERTENO (optional, Multi-Tenant must be equipped). ←
    
```

4.34 The CDR format of Connection records is similar to the N record except that the TERID field contains the TN (packed format) of the agent station.

4.35 The Duration field of Connection records contains the length of time a call had to wait before being served. Optionally, the customer-provided computer can read this information and provide RUSH displays on agent screens to signal that calls are waiting for an excessive length of time.

Note: Connection records are not recorded on magnetic tape.

4.36 Priority Connection Record. The primary use of Connection records is to integrate the SL-1 ACD feature with the customer-provided computer to transfer the call profile from agent to agent. In the integrated environment, call completion consists of three parts:

- (a) switching through a voice path
- (b) informing the computer of the connection (Connection record)
- (c) having the customer-provided computer output the call profile on the terminal of the agent receiving the call.

4.37 The time required to process all three parts of the call is cumulative. To reduce the real-time required to complete the call, connection records are given priority over other types of CDR records. Priorities are assigned as follows:

- | 128 ms high priority timing queue
- | Network input messages
- | 128 ms regular timing queue
- | Ringing
- | TTY input
- | CDR Connection Call Processing records
- other functions at lower priority levels (including other CDR records).

4.38 CDR Record Sequencing. The simplest type of call to an ACD DN is an incoming call from a trunk to an agent: i.e., the agent answers and terminates the call, without transferring or conferencing. The typical sequence of CDR record for this type of call is:

- | Q record - indicates a connection has been established.
- | N record - indicates that the call is terminated.

4.39 Calls involving transferring and conferencing generate a sequence of Connection, Start and End records. A typical sequence of records for transfer calls is:

- | Q record - Indicates that a connection has been established.
- | S record - Indicates that the first leg of the call has been completed. Timing and other data is provided for this portion of the call. This record is only generated for the first leg of a call.
- | R record - Indicates that the call has been transferred to a second agent. This record may appear before the S record when the first agent disconnects after the second agent answers. Another R record is generated each time the call is transferred to subsequent agents. There can be any number of additional R records.

- E record - indicates that the call is ended. It provides timing and other data for that call.

4.40 A typical sequence of records for conference calls is:

- | Q record - Indicates that a connection has been established.
- F record - Indicates that a new connection has been established between two parties for a conference.
- | S record - Indicates that the conference has been set up (third party **conferenced** to the call). It provides timing and other information for the connection. Only one **S** record is generated.
- F record - an F record is generated each time a conference connection is made.
- | E record - indicates that the call is ended. It provides timing and data for that call.

4.41 Because Connection records are processed at a higher priority than other CDR records, the actual sequence of CDR records may vary. Connection records are always output before other types of records. Under heavy traffic conditions, Connection records may be output five or more seconds before other types of records. The actual sequence of the CDR records can be determined by their serial numbers.

 5. EXAMPLES OF CALL RECORDS

5.01 This part provides examples of call records produced by various call sequences (Table 5-A). For each example, the expected TTY output is given.

Table 5-A
LIST OF SAMPLE CALLS

SAMPLE CALL	FEATURES
1	Charge Account, AUXID
2	CPN
3	Authorization Code
4	Authorization Code, Call Transfer
5	Charge Conference
6	Charge Conference
7	Charge Conference
8	Charge Account, Call Transfer

Table 5-B
OUTPUT OF SAMPLE CALL #1

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID ll.s.cc.uu	DATE mm/dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS
C	008	00	DN7234	T04016	007.2.04.03	09/07	10:07	123456	Note 2
N	017	00	DN7234	T04016	007.2.04.03	09/07	10:07	00:02:10	99361212

Note 1: Prior to Release 4, the AUXID format was ll.s.cc.u

Note 2: When DURATION is not recorded, the digits dialed are output in that column.

Note 3: Other unrelated call records may be inserted between these records.

Sample Call 1

5.02 Table 5-B gives an example involving the Charge Account and AUXID features.

- (1) DN 7234, a multiple appearance prime DN, dials 9-936-1212 and enters a charge account (123456). The call is completed on trunk group 4, member 16.
- (2) The Charge Account record is generated as soon as the account number is fully entered. The Normal record with AUXID (loop/shelf/card/unit) is generated at the termination of the call.

Table 5-C
OUTPUT OF SAMPLE CALL #2

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/dd	TIME h h : m m	DURATION hh:mm:ss	DIGITS
P	025	00	TO0005	DN9876		09/07	11:10	2019493000	Note 1
N	027	00	TO0005	DN9876		09/07	11:09	00:12:05	

Note 1: When DURATION is not recorded, the calling party number digits are output in that column.

Note 2: Other unrelated call records may be inserted between these records.

Sample Call 2

5.03 Table 5-C gives an example involving the Calling Party Number (CPN) feature.

- (1) DN 9876 receives a collect call on trunk group 0, member 5 and enters CPN 201-949-3000.
- (2) The CPN record is generated after the calling party number is entered and the caller has returned to the call. The Normal record is generated at the termination of the call.

Table 5-D
 OUTPUT OF SAMPLE CALL #3

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/dd	TIME h h : m m	DURATION hh:mm:ss	DIGITS
A	039	00	DN3456	T00045		09/07	11:49	12345678	Note 1
N	051	00	DN3456	T00045		09/07	11:51	00:07:15	92126823333

Note 1: When DURATION is not recorded, the authorization code is output in that column.

Note 2: Other unrelated call records may be inserted between these records.

Sample Call 3

5.04 Table 5-D gives an example involving the Authorization Code feature.

- (1) DN 3456 enters an authorization code number 12345678 and dials 9-212-682-3333. The call is completed on trunk group 0, member 45.
- (2) The Authorization Code record is generated after the code is entered and accepted. The Normal call record is generated at the end of the call.

Table 5-E
OUTPUT OF SAMPLE CALL #4

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/dd	TIME hh:mm	DIGITS
A	057	00	DN7865	T00019		09/07	13:07	87654321
S	059	00	DN7865	T00019		09/07	13:10	9*7145559292
E	079	00	T00019	DN3131		09/07	13:18	

Note 1: Other call records may be inserted between these records.

Note 2: DURATION is not supplied but must be calculated from the TIME field of S and E records.

Sample Call 4

5.05 Table 5-E gives an example involving the Authorization Code and Call Transfer features.

- (1) DN 7865 dials **9--*--714-555-9292#** and the call completes through trunk 19. The call is transferred to DN 3131 but an authorization code (87654321) is entered prior to the transfer. The call terminates shortly after the transfer is made.
- (2) An Authorization Code record is generated after the entry is complete. When the call is transferred, a Start record is generated. When the call is terminated, an End record is generated.

Table 5-F
OUTPUT OF SAMPLE CALL #5

REC TYP	REC NO	C U S NO	ORIGID Note 1	TERID	AUXID	DATE mm/dd	TIME hh:mm	DIGITS
C	061	00	DN6543	T00016		09/07	15:10	123456
C	063	00	DN6543	T00045		09/07	15:11	123457
S	071	00	DN6543	T00016		09/07	15:12	99699170
S	072	00	DN6543	T00045		09/07	15:10	9*212626170
c	073	00	DN6543	T00067		09/07	15:12	123458
S	079	00	DN6543	T00067		09/07	15:13	92015425747
E	087	00	CF01980	T00067		09/07	15:25	
E	088	00	CF01980	T00016		09/07	15:31	
E	091	00	CF01980	T00045		09/07	15:31	

Note 1: Prior to Release 4, the conference number in the ORIGID field had a format of CFLN, e.g., CF1980.

Note 2: Other call records may be inserted between these records.

Note 3: DURATION is not supplied, but must be calculated from S and E records using the TIME field.

Sample Call 5

5.06 Table 5-F gives an example involving the Charge Conference feature.

- (1) DN 6543 establishes a conference call with 3 other parties, entering an account code prior to connecting each party. The parties were connected sequentially on trunks 16, 45 and 67. The parties disconnected in the order 67, 16 and 45. Conference bridge 1980 was used.
- (2) The account codes entered were 123456, 123457 and 123458 in that order.
- (3) The digits dialed were 9-969-9170 for the first party, 9-*--212-262-6170 for the second and 9-201-542-5747 for the third.
- (4) A Charge Account record is generated after each entry is completed. A Start record is generated after each party is added; however, the first two Start records are generated together after the system recognizes the conference situation exists. An End record is generated as each trunk disconnects.

Table 5-G
OUTPUT OF SAMPLE CALL #6

REC TYP	REC NO	CUS NO	ORIGID Note 1	TERID	AUXID	DATE mm/dd	TIME hh:mm	DIGITS
S	103	00	DN6543	T00016		09/07	11:17	99699170
S	104	00	DN6543	TO0045		09/07	11:17	9*2122626170
S	107	00	DN6543	T00067		09/07	11:18	92015425749
M	112	00	T00045	DN6543		09/07	11:19	123456
M	113	00	T00016	DN6543		09/07	11:19	123456
M	115	00	T00067	DN6543		09/07	11:19	123456
E	121	00	CF01122	T00067		09/07	11:40	
E	126	00	CF01122	T00016		09/07	11:45	
E	127	00	CF01122	TO0045		09/07	11:47	

Note 1: In Generic X11 Release 4, the conference number in the ORIGID field has a format of CFLLNN, e.g., CF01122.

Note 2: Other call records may be inserted between these records.

Note 3: DURATION is not supplied, but must be calculated using the TIME fields of S and E records.

Sample Call 6

5.07 Table 5-G gives an example involving the Charge Conference feature.

- (1) DN 6543 places the same conference call as in sample call 4, except this time the account code is entered after the conference has been established and the same account code (123456) is intended to apply to all 3 conferees. Conference bridge 1122 is used.
- (2) A Start record is generated as each party is connected. The first two Start records are generated at the same time as the system recognizes a conference situation.
- (3) After the account code is entered a separate Charge Conference record is generated for each trunk involved.

Table 5-H
OUTPUT OF SAMPLE CALL #7

REC TYP	REC NO	CUS NO	ORIGID Note 1	TERID	AUXID	DATE mm/dd	TIME hh:mm	DIGITS
S	017	00	DN8765	TO0027		09/08	10:10	99291123
S	018	00	DN8765	T00037		09/08	10:11	99461130
M	021	00	T00037	DN8765		09/08	10:12	123456
M	023	00	TO0027	DN8765		09/08	10:12	123456
C	037	00	DN8765	T00047		09/08	10:27	123457
S	039	00	DN8765	T00047		09/08	10:29	9*9299170
E	051	00	CF01122	T00037		09/08	11:10	
E	053	00	CF01122	T00047		09/08	11:10	
E	055	00	CF01122	TO0027		09/08	11:11	

Note 1: Prior to Release 4, the conference number in the ORIGID field had a format of CFLLNN, e.g., CF1122.

Note 2: Other call records may be inserted between these records.

Note 3: DURATION is not given, but must be calculated from the TIME fields of S and E records.

Sample Call 7

5.08 Table 5-H gives an example involving the Charge Conference feature.

- (1) DN 8765 places a conference call with 2 other parties on trunks 27 and 37. A charge account (123456) is entered after the conference is established. Conference bridge 1122 is used.
- (2) Later during the conference, a fourth party on trunk 47 is added, but a new charge number (123457) is entered prior to adding the fourth party.
- (3) Two Start records are generated after the conference is established
- (4) Two Charge Conference records are generated after the charge account entry is made.
- (5) A Charge record is generated after the new charge number is entered.
- (6) A Start record is generated after the fourth party is added.
- (7) End records are generated as the trunks disconnect from the conference.

Table 5-I
OUTPUT OF SAMPLE CALL #8

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/dd	TIME hh:mm	DIGITS
C	076	00	DN6789	T00006		09/08	11:15	123451
S	081	00	DN6789	T00006		09/08	11:16	9*2329169166
E	097	00	T00006	DN6789		09/08	11:31	

Note 1: Other call records may be inserted between these records.

Note 2: DURATION is not given, but must be calculated from the TIME fields of S and E records.

Sample Call 8

5.09 Table 5-I gives an example involving the Call Transfer and Charge Account features.

- (1) DN 6789 enters account code 123451 and dials **9*232-916-9166**. The call is completed on trunk 6 and later transferred to DN 5600.
- (2) Later the call is transferred back to DN6789 and terminated shortly after.
- (3) An Account Code record is generated when the entry of the code is complete.
- (4) When the call is transferred, a Start record is generated.
- (5) When the second transfer takes place, no record is generated as a Start record already exists for the trunk in use.
- (6) When the call terminates, an End record is generated.
- (7) Note that the identity of the intermediate party (DN 5600) is lost because an additional account code was not entered during or prior to the call transfer.

↗ **Table 5-J**
 OUTPUT SAMPLE CALLS (MULTI -TENANT SERVICE)

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE	TIME	DURATION	DIGITS DIALED
N	001	05	DN4999	A001009	27.1.02.1	06/28	10:14	00:00:20	98289124
			042	000					
N	001	04	T002010	DN5000		06/28	10:15	00:00:40	
			000	000					
S	002	05	T002004	DN5064		06/28	10:18		
			000	000					

Note: Other call records may be inserted between these records.

Sample Call
 (Multi -Tenant)

5.10 Table 5-J gives examples of calls in the Multi-Tenant Service environment.

Record 1. DN 4999, a member of Tenant group 042, Customer 5, dials 9-828-9124. The trunk is a shared customer resource.

Record 2. DN 5000 of Customer 4 receives an incoming call. Customer 4 has not enabled Tenant Service, so all resources contain a 000 in the Tenant Number field.

Record 3. DN 5064 of Customer 5 receives an incoming call. Customer 5 has Tenant Service enabled but the station does not belong to a Tenant group so all resources contain a 000 in the Tenant Number fields.



INTEGRATED SERVICES NETWORK ←

MERIDIAN SL-1* ←

CALL DETAIL RECORDING
GENERIC X37
DESCRIPTION

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Reason for Reissue: To add the CDR Improved Duration Resolution feature for Release 5 of Generic X37. Changes are indicated by revision marks in the margin. ↗ ↘

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1. GENERAL

1.01, The Call Detail Recording (**CDR**) feature is an optional software package which provides a record of selected calls for accounting and administration purposes. The basic CDR call records include the identity of the called and calling parties and the duration of the call. Additional records are generated when certain features are used (e.g., conference, authorization codes) or by certain system events (e.g., initialization, change of system time clock).

1.02 The CDR call records are assembled by system software and sent to an EIA RS-232-C-compatible device such as a teletypewriter (**TTY**) or to one of the available CDR storage equipment packages.

1.03 This practice describes the operation of the CDR feature for Generic X37 Release 3+. A general description of CDR and a list of associated NTP can be found in 553-2631-100.

2. FEATURES AND OPTIONS

BASIC CDR SOFTWARE PACKAGE

2.01 The basic CDR software package is required to generate the call records. This package is always required and has the following features:

- (a) **Multicustomer Operation.** This feature allows each customer to individually select the CDR feature and its options. The feature is enabled or disabled on a customer basis in the Customer Data Block (overlay 15).
- (b) **Call Types.** Each customer may select for call recording any combination of the following call types on a trunk route basis:
 - all calls
 - | all answered calls
 - all toll calls, North American definition (first digit after the trunk access code is 0 or 1)
 - | all answered toll calls, North American definition
 - | all toll calls, international definition (first digit after the trunk access code is a customer-defined toll digit)
 - | all answered toll calls, international definition
- (c) **Multiple Appearance Directory Numbers (MADN).** As a customer option, a MADN may be supplemented by an auxiliary identification (AUXID) giving the shelf, card and unit of the originating set. An AUXID field is provided only if the originating set has a MADN as the prime DN. If a call originates from a MADN which is not the prime DN, the prime DN is shown as the originating ID with no AUXID field provided.
- (d) **Priority to CDR.** This option ensures that every call is recorded by having idle call registers selected for CDR before being used for call processing

OPTIONAL CDR SOFTWARE PACKAGES

2.02 In addition to the basic CDR software package, the following optional packages are available (selected for the type of CDR output required):

- (a) **CDR-TTY (CTY).** This software package is selected when it is required to output call records on one or more RS-232-C-compatible devices. It provides a hard copy of the call records and may be used with the other CDR optional packages.
- (b) **CDR Data Link (CLNK).** This package is required when the Single/Multi-Port CDR storage system is used. The CLNK software formats the call records into a form suitable for storage on magnetic tape.
- (c) **CDR with Charge Account (CHG).** This package provides capability for direct billing of calls to specific charge account numbers.



(d) CDR Improved Duration Resolution (**CDR5**).

This feature can be added only to the CTY package. It is applicable to all outgoing CO calls on loop start trunks with answer supervision only, providing an accuracy of 0.5 seconds on CTY output records. The duration field, HH:MM:SS:X where X=0 or 5, is always rounded up to a multiple of 0.5 seconds. All other calls remain rounded in multiples of 2 seconds.

If this option is equipped but is not selected in customer overlay 15, then the duration on loop start trunks with answer supervision is rounded to the nearest second. All other calls are in 2 second multiples.



CDR STORAGE
OPTIONS

2.03 Two CDR equipment options provide for the storage of call records on magnetic tape.

2.04 Single/Multi-Port CDR Storage System. This hardware option consists of a single equipment cabinet that contains a Central Processing Unit (CPU), a 9-track magnetic tape and tape control circuitry. The Multi-Port system is capable of storing call records from a maximum of 12 different Meridian SL-1 installations.

2.05 The CDR system collects the call records from the Meridian SL-1 machine(s), formats them into blocks and stores the blocks on tape. When the tape is full, it is removed from the CDR cabinet and sent for downstream processing into CDR reports.

3. CALL RECORD OUTPUT

3.01 Call records may be output on **TTY** (Fig. 3-1) or on tape (Fig. 3-2).

(a) **TTY Output.** Each call record is output on a TTY as a single line of data. The data is broken up into fields, the significance of which is determined by their position in the line. Fields are separated by one or more blank characters. The unused fields in a given record are left blank.

(b) **Tape Output.** *When* call records are to be stored on magnetic tape, they are collected in the system memory and then transferred to the tape device as a block via an **SDI** port. Null records are used to **fill up** blocks so that individual call records do not span tape **blocks**.

3.02 For 9-track magnetic tape storage, a Tape Information Package (**P0572490**) is available. This package is used to facilitate the design of computer programs to read the tapes, decode the **call** records and produce accurate accounting reports.

RECORD CONTENTS

3.03 A call record on a TTY or printer contains the following fields except where noted.

Record Type

3.04 The Record Type (**RECTYPE**) field indicates the type of call record.

3.05 TTY Output. This field consists of a single letter identifying the type of **record**:

A Authorization Code

C Charge record

E End record

I Initialization record

N Normal record

M Charge for Conference

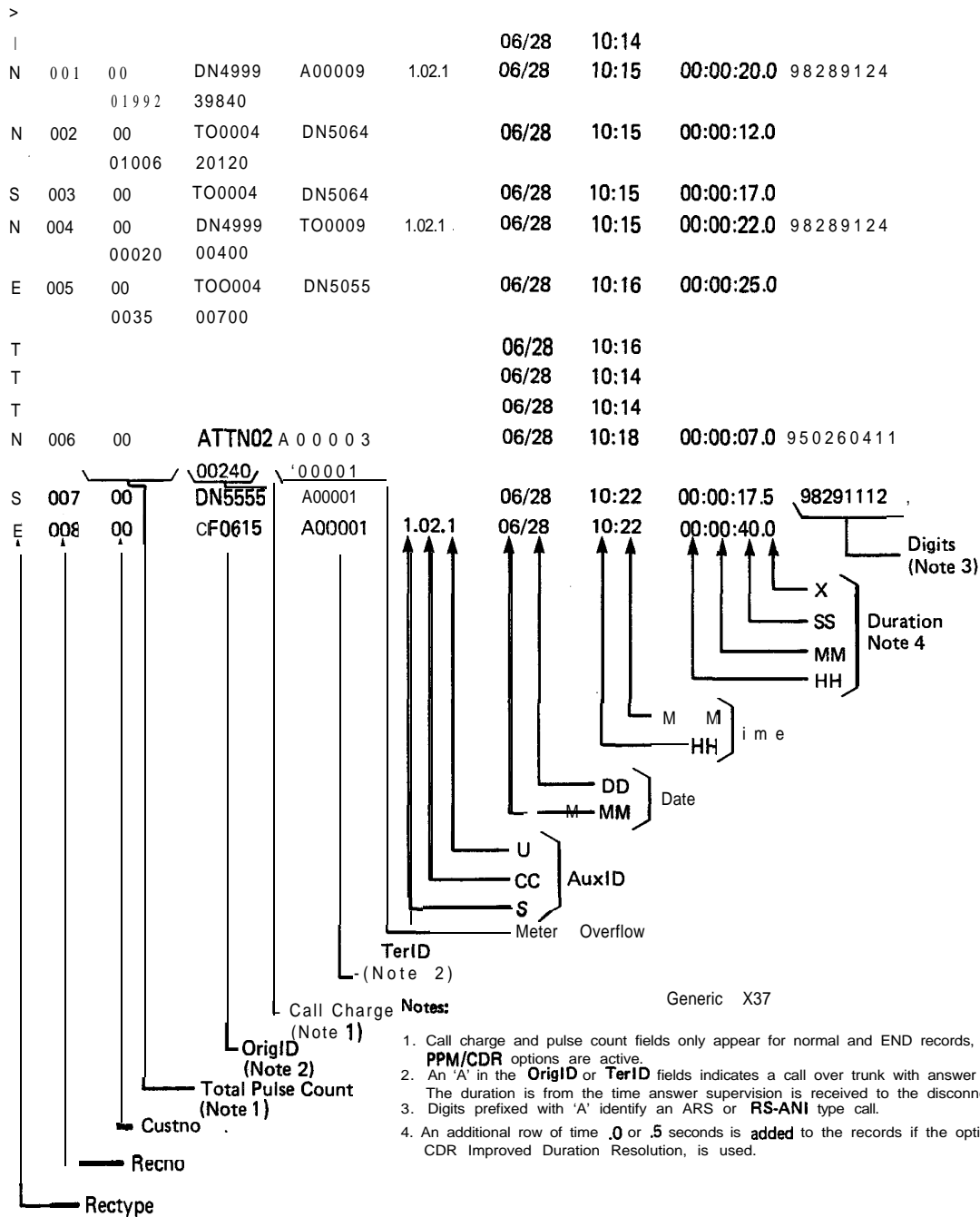
P Calling Party Number

S Start Record

T **Timestamp** record.

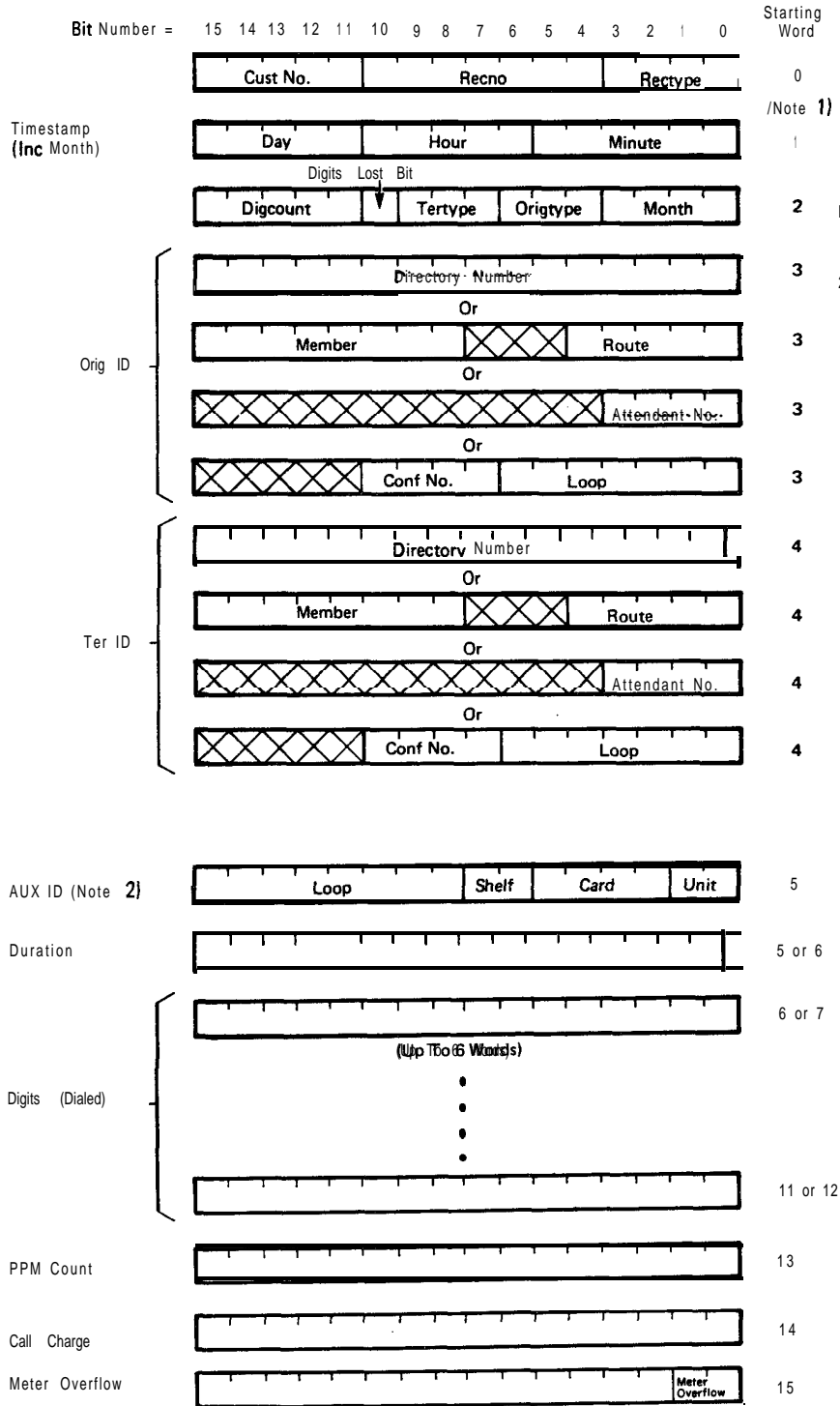
3.06 A C record is generated for a charge code entered before establishing a call, while the call is established or when charge codes are entered prior to conferencing in each party. An M record is generated when a charge account code is entered after a conference is completed.

3.07 Tape Output. This five-bit field can be O-14 in binary and defines the type of record:



(III. 1158)

Fig. 3-1
→ Format of Cal I Records on TTY



Starting Word

/Note 1)

Generic X37

Notes:

1. Words **0, 1** and 2 define completely the format of the Balance of the Record.
2. The Aux ID is allocated only if either **Origtype** or **Tertype** requires it.

Fig. 3-2
Format of a Call Record on Tape

APPENDIX 3 TO 553-2631-100

- 0 Null record
- 1 Normal record
- 2 Start record
- 3 End record
- 4 Charge record
- 5 Initialization record
- 6 Timestamp record
- 7 Normal record for AN1 or RS-AN1
- 8 Start record for AN1 or RS-AN1**
- 9 Calling Party Number (**CPN**)
- 10 Authorization Code
- 11 Charge for Conference

Record Number

3.08 The Record Number (**RECNO**) field identifies the number of the current record in the CDR sequence.

3.09 TTY Output. This three-character field gives the record sequence number, modulo 128, right-justified.

3.10 Tape Output. This seven-bit field gives the record number in binary, modulo 128.

Customer Number

3.11 The Customer Number (**CUSTNO**) field identifies the customer associated with the call.

3.12 TTY Output. A two-character field identifies the customer number (0-31).

3.13 Tape Output. A five-bit field identifies the customer (0-31 in binary).

Originator Type (**Tape Only**)

3.14 The three-bit Originator Type (**ORIGTYPE**) field identifies the type of apparatus that originated the call:

- 0 **500/2500** set or SL-1 station
- 1 **500/2500** set or SL-1 station with which AUXID is associated
- 2 Trunk
- 3 Attendant
- 4 Trunks, END record expected
- 5 Conference.

Terminator Type (Tape Only)	3.15 The Terminator Type (TERTYPE) field identifies the type of apparatus to which the call was terminated. Definition of fields is the same as for ORIGTYPE .
Originator Identification	3.16 The Originator Identification (ORIGID) field identifies the apparatus that originated the call. 3.17 TTY Output. The format depends on the type of equipment as follows: <ul style="list-style-type: none"> ● Stations. Stations are identified in the form DNxxxxxx, where xxxxxx is the prime DN for the station. Trunks. The ORIGID field is in the format Trmmmm, where rr is the route number and mmm is the member number. Attendants. Attendants are identified in the form ATTNxx, where xx is the console number (1-15). Conference Loops. Conference loops are identified in the form CFllnn, where ll is the loop number (0-79) and nn is the conference number.
Terminator Identification	3.15 Tape Output. The format of the 24-bit field depends on the type of equipment as follows: <ul style="list-style-type: none"> ● Stations. If the ORIGTYPE or TERTYPE fields are 0 or 1 (i.e., a station), this field identifies the 500/2500 set DN or prime DN of the SL-1 set that originated the call. Up to six decimal digits are stored in normal hexadecimal format, with the exception of: <ul style="list-style-type: none"> zero, which is represented by hex A *, which is represented by hex B #, which is represented by hex C. Trunks. If the ORIGTYPE or TERTYPE fields are 2 or 4 (i.e., the originator is a trunk), this field is 15 bits long. <ul style="list-style-type: none"> The trunk route number is 7 bits. The trunk route member number is 8 bits. Attendants. If the ORIGTYPE or TERTYPE field is 3 (i.e., an attendant), attendant number is identified in 4 bits. Conference Loops. If the ORIGTYPE or TERTYPE field is 5 (i.e., conference), loop and conference number are identified in 11 bits: <ul style="list-style-type: none"> 7 bits ■ loop number 4 bits ■ conference number.
Digit Count (Tape Only)	3.19 The Terminator Identification (TERID) field identifies the apparatus on which a call is terminated. Contents are the same as for ORIGID .
Digits Lost (Tape Only)	3.20 This field gives the total number of digits dialed in 5 bits. 3.21 This field consists of one bit, which is set to 1 if the total digits dialed exceeds 23.

Auxiliary Identification 3.22 The Auxiliary Identification (AUXID) field identifies the TN of the originating station when the station has a multiple appearance prime DN, and thus cannot be uniquely identified by ORIGID.

3.23 TTY Output. The AUXID field is in the form **s.cc.u** and uniquely identifies originating station shelf, card and unit.

3.24 Tape Output. This field (16 bits) depends on the card density of the originating station.

- | Single Density - **XXXXXXXXSSCCCCU** (X = not used)
- | Double Density - **XXXXXXXXSUCCCCU**

Timestamp 3.25 The **TIMESTAMP** field gives the date and time. Its exact definition depends on the type of record:

- Normal record - end of call
- Start record - start of call
- End record - end of call
- Initialization record - time of system initialization
- Timestamp** - old or new timestamp

3.26 If the call started on the last day of a month and stopped on the first day of the next month then the start day is shown as day 0 of the next month.

3.27 TTY Output. The format is **MM/DD HH:MM** where:

- MM** is month (1-12)
- DD** is day (1-31)
- HH** is hour (0-23)
- MM** is minute (0-59)

3.28 Tape Output. The timestamp field occupies 20 bits as follows:

- MONTH (four bits) where Jan. = 1 and Dec. = 12
- DAY (five bits) where first day of month = 1
- HOUR (five bits) which can be 0 to 23
- MINUTE (six bits) which can be 0 to 59.

Call Duration	<p>3.29 The DURATION field measures the call duration in units of 2 seconds or 0.5 seconds for optional CDR software package CDR5. In a Normal record, this is the duration of the call from start to disconnect. In a Start record, this is the duration of the call from start to first feature usage at the time given by TIMESTAMP. An incoming call is deemed to start at the time of presentation to the called party. An outgoing call starts at the time of trunk seizure (dial tone removed). In an End record, the time the call is presented to the last party until disconnect of that call is given in the duration field.</p> <p>3.30 TTY Output. The output is in the form HH:MM:SS.X where:</p> <p>HH is hour (0-23)</p> <p>MM is minutes (0-59)</p> <p>SS is seconds (0-59)</p> <p>X is one half second (0 or 5) for optional package CDR5.</p> <p>3.31 Tape Output. DURATION takes up 16 bits, and gives a binary count of the call duration in seconds.</p>
Digits	<p>3.32 The DIGITS field identifies the digits dialed, digits outpulsed, or Charge Account code entered. Up to 23 digits can be recorded. If more digits are dialed, only the first 16 are recorded correctly. Digit 24 and succeeding digits are cycled over digits 17 to 23.</p> <p>3.33 Route Selection. Route Selection digits (for ARS or RS-ANI) are indicated if preceded by the letter A. The digits shown in the digits field are those that are actually outpulsed by the system after route selection and digit manipulation.</p> <p>3.34 Dictation and Paging Trunks. The CDR printout for dictation and paging trunks accessed by a 2500-type set specifies only the trunk access code in the DIGITS field. Dictation trunks require tones to instruct the machines at the other end. The 2500-type set sends these tones directly to the dictation trunk without using a call register to store digits. As these digits are not stored in a call register, the CDR cannot print them out. In the case of 500-type and SL-1 sets, the digits must be stored in a call register and then a DTR translates the digits into tones. The CDR output will therefore show all digits dialed by these sets.</p> <p>3.35 TTY Output. Up to 23 digits can be output on the terminal.</p> <p>3.36 Tape Output. Each digit is stored as a four-bit word. Total digit storage is a maximum of eight 16-bit words.</p>
End of First Line (TTY Only)	<p>3.37 After the DIGITS field is padded out to the end of the line, the Meridian SL-1 outputs a Carriage Return character. The remaining fields, if required, are output on a new line.</p>
PPM Count	<p>3.38 The PPM Count (PPMCNT) field records the PPM count (0 to 32767) received for call on metered CO routes, for NORMAL and END records only. If the PPM count goes beyond this value, the residual PPM count is printed.</p> <p>3.39 TTY Output. Up to 5 digits are output, right justified.</p>

3.40 Tape Output. This field occupies a 16-bit word.

Cal | Charge 3.41 The Call Charge (**CALLCHG**) field is the number of PPM pulses received multiplied by a customer-defined value to a maximum of 0 to 65535. In the case of a value beyond this, the field shows OVF.

3.42 TTY Output. Up to 5 digits, right justified.

3.43 Tape Output. This field occupies a 16-bit word.

Meter Overflow 3.44 The Meter Overflow (**MTROVFL**) field counts the number of PPM meter overflows that have **occurred**(to a maximum of three).

3.45 **TTY** Output. The output is up to five digits, preceded by *.

3.46 Tape Output. This field occupies a 16-bit word.

Line Feed (**TTY** Only) 3.47 As a customer option, up to 9 blank lines can be output at the end of each CDR record.

RECORD LENGTH FOR TTY 3.48 The following formulae permit the calculation of lengths of all TTY records in number of characters. (See Tables 3-A, 3-B and 3-C for definitions of the call record fields for, respectively, Generic X37 Release 1 (**ORIGID** as **DN**), X37 Release 1 (**TERID** as **DN**) and X37 Release 3+.)

$$\text{NORMAL RECORD} = C + 55 + P + N$$

$$\text{START RECORD} = C + 55 + N$$

$$\text{END RECORD} = C + 55 + P$$

$$\text{INITIALIZATION RECORD} = C + 46$$

$$\text{TIMESTAMP RECORD} = C + 46$$

$$\text{AUTHORIZATION CODE RECORD} = C + 46 + P + N$$

$$\text{CHARGE RECORD} = C + 46 + N$$

$$\text{CHARGE-CONFERENCE RECORD} = C + 46 + N$$

$$\text{CPN RECORD} = C + 46 + N$$

3.49 The definition of the variables are as follows:

C is the number of control characters (CR, LF, **NUL**) per record, typically 8

N is the number of digits to be recorded.

P is 12 if PPM data is included (plus 7 if meter overflows); 0 if not.

RECORD LENGTH FOR TAPE 3.50 To separate call records within a tape block, the length of each record must be known. The following formulae permit the calculation of the lengths of all call records in number of 16-bit words:

$$\text{NORMAL RECORD} = 6 + D + T + \text{CEILING}(N/4) + P$$

START RECORD = 6 + D + T + CEILING (N/4) + P

END RECORD = 6 + D + T + P

INITIALIZATION RECORD = 3

TIMESTAMP RECORD = 3

AUTHORIZATION CODE RECORD = 5 + D + T + CEILING (N/4) + P

CHARGE RECORD = 5 + D + T + CEILING (N/4)

CHARGE-CONFERENCE RECORD = 5 + D + T + CEILING (N/4) + P

CPN RECORD = 5 + D + T + CEILING (N/4) + P

3.51 The definition of the variables are as follows:

D is 1 if **ORID** or TERID is a DN, 0 if not, 2 if BOTH are DN

N is the number of digits to be recorded

T is 1 if AUXID is included, 0 if not.

P is 3 if PPM data is included, 0 if not.

CEILING (X) is the smallest integer greater than or equal to X.

Table 3-A
 TTY OUTPUT FORMAT (GENERIC X37 RELEASE 1 WITH **ORIGID AS DN**)

CHARACTER POSITION	NAME	FORMAT	DEFINITION
1	RECTYPE	Y	Record Type.
2	<blank>		Blank space
3-5	RECNO	XXX	Record Sequence Number
6	<blank>		Blank space
7-8	CUSTNO	XX	Customer Number
9	<blank>		Blank space
10-17	ORIGID	DNXXXXX__	Originating Identification: Station Directory Number
18	<blank>		Blank space
19-24	TERID	TRRMMM or ARRMMM or ATTNXX or CFLN	Terminating Identification: Trunk Trunk with Answer Supervision Attendant Number Conference Number
25	<blank>		Blank space
26-31	AUXID	S.CC.U	Auxiliary Identification
32	<blank>		Blank space
33-43	TIMESTAMP	MM/DD_HH:MM	Timestamp
44	<blank>		Blank space
45-52	DURATION	HH:MM:SS	Duration
53	<blank>		Blank space
54-78	DIGITS	XXX...X or A_XXX...X	Digits Dialed: Up to 23 normal digits Route Selection was used
→ 45-78	DURATION	XXX...X	14 Authorizaton code digits or 23 Charge Account digits or 23 Calling Party Number digits

Table 3-B
TTY OUTPUT FORMAT (GENERIC X37 RELEASE 1 WITH **TERID AS DN**)

CHARACTER POSITION	NAME	FORMAT	DEFINITION
1	RECTYPE	Y	Record Type.
2	<blank>		Blank space
3-5	RECNO	XXX	Record Sequence Number
6	<blank>		Blank space
7-8	CUSTNO	XX	Customer Number
9	<blank>		Blank space
10-15	ORIGID	TRRRMM or ARRMM or ATTNXX or CFLN	Originating Identification: Trunk Trunk with Answer Supervision Attendant Number Conference Number
16	<blank>		Blank space
17-24	TERID	DNXXXX_	Terminating Identification: Station Directory Number
25	<blank>		Blank space
26-31	AUXID	S.CC.U	Auxiliary Identification
32	<blank>		Blank space
33-43	TIMESTAMP	MM/DD_HH:MM	Timestamp
44	<blank>		Blank space
45-52	DURATION	HH:MM:SS	Duration
53	<blank>		Blank space
54-78	DIGITS	XXX...X or A_XXX...X	Digits Dialed: Up to 23 normal digits Route Selection was used ←
45-78	DURATION	XXX...X	14 Authorizaton code digits or 23 Charge Account digits or 23 Calling Party Number digits

Table 3-C
TTY OUTPUT FORMAT (GENERIC X37 RELEASE 3+)

CHARACTER POSITION	NAME	FORMAT	DEFINITION
1	RECTYPE	Y	Record Type.
2	<blank>		Blank space
3-5	RECNO	XXX	Record Sequence Number
6	<blank>		Blank space
7-8	CUSTNO	XX	Customer Number
9	<blank>		Blank space
10-17	ORIGID	TRRMMM__ or ARRMMM__ or DNXXXXXX or ATTNXX__ or CFLLNN__	Originating Identification : Trunk Trunk with Answer Supervision Station Directory Number Attendant Number Conference Number
18	<blank>		Blank space
19-26	TERID	TRRMMM__ or ARRMMM__ or DNXXXXXX or ATTNXX__ or CFLLNN__	Terminating Identificarion : Trunk Trunk with Answer Supervision Station Directory N urn ber Attendant Number Conference Number
27	<blank>		Blank space
28-33	AUXID	S.CC.U	Auxiliary Identification
34	<blank>		Blank space
35-45	TIMESTAMP	MM/DD_HH:MM	Timestamp
46	<blank>		Blank space
→ 47-56	DURATION	HH:MM:SS.X	Duration
→ 57	<blank>		Blank space
→ 58-80	DIGITS	XXX...X or A_XXX...X	Digits Dialed: Up to 21 normal digits Route Selection was used
→			
47-80	DURATION	x x x - x	14 Authorizat on code digits or 23 Charge Account digits or 23 Calling Party Number digits

4. CALL RECORD GENERATION

4.01 A simple call generates a single call record. Calls that are modified because of certain features (e.g., call transfer, conference) generate multiple records.

4.02 For every call to be recorded, the system software generates one Normal record or several Start and End records. Additional record types are generated to accommodate certain features.

4.03 The different types of call records and their application are outlined below.

Normal Record

4.04 A Normal record is generated when a simple call is established, whether or not extended through the attendant console, and when no other set feature is activated. The Normal record contains the following information:

RECTYPE
 RECNO
CUSTNO
 ORIGID
 TERID
 AUXID (optional)
 TIMESTAMP
 DURATION
 DIGITS

4.05 If PPM information is required and transfer records are allowed:

PPMCNT
CALLCHG
 MTROVFL

4.06 For a Normal record, all telephone set dial pad input is included in the CDR record until such time as the End-of-Dialing (**EOD**) timer interval is exceeded or the user enters **#** from the set keyboard. Thus, the dialed DN portion of the CDR record may include * symbols and unused digits (e.g., in many cases speed call and **autodial** numbers include * symbols, each of which when interpreted by the system causes a 3 s pause to occur). A user may enter useless digits prior to the system receiving an EOD timeout or user-initiated # entry. In such cases, the call is completed to the correct destination but the CDR record contains the useless digits.

4.07 Route Selection. When one of the Route Selection features are used, the letter A precedes the Digits field on TTY outputs. These features are Automatic Route Selection (**ARS**) and Route Selection-Automatic Number Identification (**RS-ANI**). The actual digits that appear in the Normal (or Start) records are those that are actually outputted by the system after route selection and any required digit manipulation. The actual digits dialed are reflected only if no digit manipulation was required to process the call.

4.08 Incoming Calls. An incoming call answered by the attendant and extended to a station generates a Normal record, indicating the trunk as the ORIGID and the station as the TERID. No indication is given that the attendant is involved in the call. When the calling party abandons the call before the station answers, and before recall is activated, no record is generated. However, when the call is abandoned during recall, the attendant console is shown as the TERID. Incoming calls answered by the attendant but not extended to a station are shown as terminating at the attendant console.

4.09 Outgoing Calls. An outgoing call placed by the attendant generates a Normal record, indicating the attendant console as the ORIGID and the trunk as the TERID. However, when the call is extended to a station, the Normal record generated indicates the trunk as ORIGID and the station as the TERID. The Digits field includes the station number dialed by the attendant.

4.10 Ring Again. When the Ring Again feature is activated, a record is generated only when a trunk is seized.

4.11 TIE Trunk Operation. Calls placed over tandem TIE trunks are billed from the time the call is answered (i.e., answer supervision is received), rather than from the time the trunk is seized. A Normal (or Start record) is generated only if answer supervision is received. Thus the calling party is not charged for the time taken for outpulsing and ringing.

Start Record

4.12 A Start record is generated when a call receives treatment from certain features of the Meridian SL-1. A Start record contains the following information:

RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
DURATION
DIGITS

4.13 If PPM information is required and transfer records are allowed:

PPMCNT
CALLCHG
MTROVFL

4.14 Call Transfer. When the Call Transfer feature is activated on an established call, a Start record is generated instead of a Normal record. The record is generated when the transfer is completed and shows the two involved parties immediately before the feature was activated. On termination of the call, an End record is generated showing its final disposition. Start records are not generated for intermediate stations when a call is transferred more than once: CDR records, therefore, do not indicate this occurrence.

4.15 Conference. When the Conference feature is activated on an established call, a Start record is generated as described for the Call Transfer feature. A subsequent Start record is generated for each CDR trunk included in the conference. The Duration field is calculated from the previous Start record timestamp to the current Start record timestamp. Although the sequence of related Start records generated may be altered by the CDR processing routines, the chronological **(timestamp)** data remains intact and each Start record is generated before its corresponding End record. The End record shows the conference bridge as the ORIGID and the **conferenced** trunk as TERID.

4.16 Call Forward. When the Call Forward feature is activated and results in a trunk-to-trunk (tandem) call, a consecutive pair of Start records are generated. The first record indicates the incoming trunk as ORIGID and the call forward DN as TERID. The second record indicates the call forwarded DN as the ORIGID and the outgoing trunk as the TERID. Both records indicate the same timestamp and duration data. An End record is generated at the end of the call.

4.17 Other Features. When barge-in, busy verification, privacy release or override is applied to an established call, a Start record is generated. Calls that would normally generate a Normal record are altered to generate a Start record. The record indicates that a feature was activated during the call, as well as any changes to the parties involved. The trunk ID remains consistent throughout. An End record is generated on termination of the call.

End Record

4.16 Each End record is associated with a specific Start record and is generated on termination of the call. The record shows the final disposition of the call. The timestamps on the corresponding Start and End records can be used to calculate the approximate total Duration of a call. The duration field in the End record contains the length of the call from the time the modified call is presented to the last party until call termination. The End record contains the following information:

RECTYPE
 RECNO
CUSTNO
 ORIGID
 TERID
 AUXID (optional)
 TIMESTAMP
 DURATION

4.19 If PPM information is required and transfer records are allowed:

PPMCNT
 CALLCHG
 MTROVFL

Initialization Record

4.20 After a system initialization, a single record is generated to note this occurrence and its time in the following **format**:

RECTYPE
 TIMESTAMP

Timestamp Record 4.21 When the system time or date is changed from either the attendant console or a TTY, a consecutive pair of records is generated specifying the old and new timestamps. The first record in such a pair gives the old timestamp; the second record contains the updated timestamp. Both records have the following format:

```
RECTYPE
TIMESTAMP
```

Authorization Code Record 4.22 Authorization Code recording is optional and is set using overlay 24. A record is generated when the code is entered and one of the following occurs:

- a trunk is seized
 - | a local set answers Direct Inward System Access (DISA) calls
 - | a DISA call cannot be connected to a local set
 - | Ring Again is activated, in which case both the ORIGID and TERID fields are the DN associated with the station entering the authorization code.

4.23 When authorization codes are stored as auto dial or speed call entries, the number stored must contain the access code followed by the authorization code only. All digits after the access code are interpreted as authorization code digits. This record contains the following information:

```
RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
DIGITS
```

Charge Account Record 4.24 The charge account record is designed to allow direct billing of calls to specific charge account numbers. Charge account number lengths (2 to 23 digits) are defined individually for each customer, by using overlay 15. A charge account number, when entered either before dialing or during an established incoming or outgoing call, causes a Charge Account record to be generated which contains the following information:

```
RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
DIGITS
```

4.25 Numbers of Fixed Length. The system assumes that a charge account number is valid when the number of digits entered corresponds to the account number length as defined in the customer data block.

- (a) When a charge account number is entered before establishing a call and too few digits are entered, the system waits 30 s (15 s for **2500-type** sets) for further input. When no further digits are entered, overflow tone is provided for 15 s, then the set is locked out by the system. A Charge record is generated showing the partially entered charge account number.
- (b) When a charge account number is entered during an established call and too few digits are entered, no response is given until the interdigit timeout occurs. Then overflow tone is provided for 15 s. After this time, the previously established call will be reconnected. On **500/2500-type** sets, if the user does not wait for a response and has dialed too few digits, then each switchhook flash is interpreted as a digit 1 until the charge account length is reached. Dial tone is then returned and the next switchhook flash will reconnect the call. On SL-1 sets without a charge key, if the user does not wait for a response and has dialed too few digits, then the call is reestablished when the DN key is depressed. However, no charge record will be produced.

4.26 Numbers of Variable Length. SL-1 sets and attendant consoles equipped with a charge (**CHG**) key are permitted to enter charge account numbers containing less digits than defined in the customer data block. The Charge account number entered is accepted by the system when the CHG key is operated. Charge account number entry is also accepted by the system by operating a DN, call transfer or conference key that was active before the CHG key was operated. The CHG key may also be used to terminate an entry to correct an error or to enter multiple charge account numbers.

4.27 Deletion of Number. A charge account record is not output by the system unless the call involves a trunk and meets the criteria set for CDR in the route data block.

4.28 Call Transfer. A call transferred from one set (A) to another set (B) generates a Start record for set A and an End record for set B when the call is terminated. However, if set B, instead of terminating the call, enters charge account number and subsequently transfers the call to another set (C), a Charge record is generated for set B. The resulting Start and End records do not indicate set B or any other intermediate set as being involved in the call. The Start, Charge and End records can only be associated on a timestamp and trunk basis, the trunk remaining consistent on all three records.

Charge Conference
Record

4.29 Charge Conference records allow the assignment of one or more charge account numbers to one, several or all members of a conference. Individual Start, Charge Conference and End records are generated for each individual trunk participating in a conference call. Individual End records are generated for each trunk as it disconnects from the conference. The Charge Conference record contains the following information:

RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
DIGITS

4.30 Different Account Numbers. To assign portions of a conference call to different charge account numbers, the account numbers must be entered when adding a trunk to a conference and before conferencing is completed. For **500/2500-type** sets, the account number is entered after the switchhook flash and before the trunk is dialed. For **SL-1** sets, the account number is entered after the conference key is operated, either before or after the trunk is dialed and before the conference key is operated a second time. The Charge Conference record generated shows the set performing the entry, the trunk added and the charge account number. A charge account number is entered for each trunk added to the conference.

4.31 Single Account Numbers. *When* all participants in a conference call are assigned the same charge account number, only one entry is required. Once all the trunks are included in the conference, the charge account number is entered in the usual manner. Separate Charge Conference records are generated for each trunk in the conference call.

Calling Party Number
(CPN) Record

4.32 The CPN record is useful in matching telephone company billing records of collect calls against call detail records. By operating a CPN key, which may be assigned to attendant consoles and SL-1 sets, and entering the calling party number (1 to 23 digits), a CPN record is generated. A CPN record is generated each time the CPN key is depressed, permitting the creation of multiple CPN records. The CPN record contains the following information:

RECTYPE
RECNO
CUSTNO
ORIGID
TERID
AUXID (optional)
TIMESTAMP
DIGITS

5. EXAMPLES OF CALL RECORDS

5.01 This part provides examples of **call** records produced by various call sequences (Table 5-A). For each example, the expected TTY output is given. Note that the DURATION will be expressed as **hh:mm:ss.x** if optional CDR software package CDR5 is equipped and selected in overlay program 15.

Table 5-A
LIST OF SAMPLE CALLS

SAMPLE CALL	FEATURES
1	Charge Account, AUXID
2	CPN
3	Authorization Code
4	Authorization Code, Call Transfer
5	Charge Conference
6	Charge Conference
7	Charge Conference .
8	Charge Account, Call Transfer
9	PPM, Call Charge

Table 5-B
OUTPUT OF. SAMPLE CALL #1

→	REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID s.cc.u	DATE mm/- dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS
	C	008	00	DN723456	To4016	2.04.3	09/07	10:07	123456	Note 2
	N	017	00	DN723456	To4016	2.04.3	09/07	10:07	00:02:10	99361212

Note 1: When DURATION is not recorded, the digits dialed are output in that column.

Note 2: Other unrelated call records may be inserted between these records.

Sample Call 1

5.02 Table 5-B gives an example involving the Charge Account and AUXID features.

- (1) DN 723456, a **6-digit** multiple appearance prime DN, dials 9-936-1212 and enters a charge account (1234561). The call is completed on trunk group 4, member 16.
- (2) The Charge Account record is generated as soon as the account number is fully entered. The Normal record with AUXID (loop/shelf/card/unit) is generated at the termination of the call.

Table 5-C
OUTPUT OF SAMPLE CALL #2

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/- dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS ←
P	025	00	T00005	DN9876		09/07	11:10	2019493000	Note 1
N	027	00	T00005	DN9876		09/07	11:09	00:12:05	

Note 1: When DURATION is not recorded, the calling party number digits are output in that column.

Note 2: Other unrelated call records may be inserted between these records.

Sample Call 2

5.03 Table S-C gives an example involving the Calling Party Number (CPN) feature.

- (1) DN 9876 receives a collect call on trunk group 0, member 5 and enters CPN 201-949-3000.
- (2) The CPN record is generated after the calling party number is entered and the caller has returned to the call. The Normal record is generated at the termination of the call.

Table 5-D
OUTPUT OF SAMPLE CALL #3

→ REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/- dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS
A	039	00	DN3456	T00045		09/07	11:49	12345678	Note 1
N	051	00	DN3456	T00045		09/07	11:51	00:07:15	92126823333

Note 1: When DURATION is not recorded, the authorization code is output in that column.

Note 2: Other unrelated call records may be inserted between these records.

Sample Call 3

5.04 Table 5-D gives an example involving the Authorization Code feature.

- (1) DN 3456 enters an authorization code number 12345678 and dials 9-212-682-3333. The call is completed on trunk group 0, member 45.
- (2) The Authorization Code record is generated after the code is entered and accepted. The Normal call record is generated at the end of the call.

Table 5-E
OUTPUT OF SAMPLE CALL #4

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/- dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS ←
A	057	00	DN7865	T00019		09/07	13:07	87654321	Note 3
S	059	00	DN7865	T00019		09/07	13:10	00:07:45	9*71455592- 92
E	079	00	T00019	DN3131		09/07	13:18	00:08:20	

Note 1: Other call records may be inserted between these records.

Note 2: Total call DURATION is not supplied but can be approximately calculated from the TIME or DURATION fields of S and E records.

Note 3: Authorization digits are output in the DURATION column.

Sample Call 4

5.05 Table 5-E gives an example involving the Authorization Code and Call Transfer features.

- (1) DN 7865 dials 9*-714-555-9292# and the call completes through trunk 19. The call is transferred to DN 3131 but an authorization code (87654321) is entered prior to the transfer. The call terminates shortly after the transfer is made.
- (2) An Authorization Code record is generated after the entry is complete. When the call is transferred, a Start record is generated. When the call is terminated an End record is generated.

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Table 5-F
OUTPUT OF SAMPLE CALL #5

→ REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/- dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS
C	061	00	DN6543	T00016		09/07	15:10	123456	Note 1
C	063	00	DN6543	T00045		09/07	15:11	123457	Note 1
S	071	00	DN6543	T00045		09/07	15:12	00:02:10	99699170
S	072	00	DN6543	T00016		09/07	15:10	00:00:42	9*212626170
C	073	00	DN6543	T00067		09/07	15:12	123458	Note 1
S	079	00	DN6543	T00067		09/07	15:13	00:01:12	92015425747
E	087	00	CF1980	T00067		09/07	15:25	00:12:06	
E	088	00	CF1980	T00016		09/07	15:31	00:18:34	
E	091	00	CF1980	T00045		09/07	15:31	00:18:48	

Note 1: Digits are contained in the DURATION field.

Note 2: Other call records may be inserted between these records.

Note 3: Total call DURATION is not supplied but may be approximately **calculated** from S and E records using the TIME and DURATION fields.

Sample Call 5

5.06 Table 5-F gives an example involving the Charge Conference feature.

- (1) DN 6543 establishes a conference call with 3 other parties, entering an account code prior to connecting each party. The parties were connected sequentially on trunks 16, 45 and 67. The parties disconnected in the order 67, 16 and 45. Conference bridge 1980 was used.
- (2) **The account** codes entered were 123456, 123457 and 123458 in that **order**.
- (3) The digits dialed were g-969-9170 for the first party, **9--212-262-6170** for the second and g-201-542-5747 for the third.
- (4) A Charge Account record is generated after each entry is completed. A Start record is generated after each party is added; however, the first two Start records are generated together after the system recognizes the conference situation exists. An End record is generated as each trunk disconnects.

Table 5-G
OUTPUT OF SAMPLE CALL #6

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/-dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS ←
S	103	00	DN6543	T00016		09/07	11:17	00:00:44	99699170
S	104	00	DN6543	T00045		09/07	11:17	00:00:08	9*21226261-70
s	107	00	DN6543	T00067		09/07	11:18	00:01:12	92015425749
M	112	00	T00045	DN6543		09/07	11:19	123456	Note 1
M	113	00	T00016	DN6543		09/07	11:19	123456	Note 1
M	115	00	T00067	DN6543		09/07	11:19	123456	Note 1
E	121	00	CF1122	T00067		09/07	11:40	00:22:06	
E	126	00	CF1122	T00016		09/07	11:45	00:27:10	
E	127	00	CF1122	T00045		09/07	11:47	00:29:12	

Note 1: Digits appear in DURATION field.

Note 2: Other call records may be inserted between these records.

Note 3: Total call DURATION is not supplied but must be approximately calculated using the TIME and DURATION fields of S and E records.

Sample Call 6

5.07 Table 5-G gives an example involving the Charge Conference feature.

- (1) DN 6543 places the same conference call as in sample call 4, except this time the account code is entered after the conference has been established and the same account code (123456) is intended to apply to all 3 conferees. Conference bridge 1122 is used.
- (2) A Start record is generated as each party is connected. The first two Start records are generated at the same time as the system recognizes a conference situation.
- (3) After the account code is entered, a separate Charge Conference record is generated for each trunk involved.

Table 5-H
OUTPUT OF SAMPLE CALL #7

REC → TYP	REC N O	CUS N O	ORIGID Note 1	TERID	AUXID	DATE mm/- d d	TIME hh:mm	DURATION hh:mm:ss	DIGITS
S	017	00	DN8765	T00027		09/08	10:10	00:02:10	99291123
s	018	00	DN8765	T00037		09/08	10:11	00:01:06	99461130
M	021	00	T00037	DN8765		09/08	10:12	123456	Note 1
M	023	00	T00027	DN8765		09/08	10:12	123456	Note 1
C	037	00	DN8765	T00047		09/08	10:27	123457	Note 1
S	039	00	DN8765	T00047		09/08	10:29	00:18:04	9*9299170
E	051	00	CF01122	T00037		09/08	11:10	00:41:24	
E	053	00	CF01122	T00047		09/08	11:10	00:41:30	
E	055	00	CF01122	T00027		09/08	11:11	00	42:04

Note 1: Digits appear in DURATION field.

Note 2: Other call records may be inserted between these records.

Note 3: Total call DURATION is not given but must be approximately calculated from the TIME and DURATION fields of S and E records.

Sample Call 7

5.08 Table 5-H gives an example involving the Charge Conference feature.

- (1) DN 8765 places a conference call with 2 other parties on trunks 27 and 37. A charge account (123456) is entered after the conference is established. Conference bridge 1122 is used.
- (2) Later during the conference, a fourth party on trunk 47 is added, but a new charge number (123457) is entered prior to adding the fourth party.
- (3) Two Start records are generated after the conference is established
- (4) Two Charge Conference records are generated after the charge account entry is made.
- (5) A **Charge** record is generated after the new charge number is entered.
- (6) A Start record is generated after the fourth party is added.
- (7) End records are generated as the trunks **disconnect** from the conference.

Table 5-1
OUTPUT OF SAMPLE CALL #8

REC TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/- dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS ←
C	076	00	DN6789	T00006		09/08	11:15	123451	Note 3
S	081	00	DN6789	T00006		09/08	11:16	00:01:44	9*23291691- 66
E	097	00	T00006	DN6789		09/08	11:31	00:02:32	

Note 1: Other call records may be inserted between these records.

Note 2: Total call DURATION is not given but must be approximately calculated from the TIME and DURATION fields of S and E records.

Note 3: Digits appear in the DURATION field.

Sample Call 6

5.09 Table 5-I gives an example involving the Call Transfer and Charge Account features.

- (1) DN 6789 enters account code 123451 and dials **9*232-916-9166**. The call is completed on trunk 6 and later transferred to DN 5600.
- (2) **Later** the call is transferred back to DN6789 and terminated shortly **after**.
- (3) An Account Code record is generated when the entry of the code is complete.
- (4) When the call is transferred, a Start record is generated.
- (5) When the second transfer takes place, no record is generated as a Start record already exists for the trunk in use.
- (6) When the call terminates: an End record is generated.
- (7) Note that the identity of the intermediate party (**DN 5600**) is lost because an additional account code was not entered during or prior to the call transfer.

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Table 5-J
OUTPUT OF SAMPLE CALL #9

REC → TYP	REC NO	CUS NO	ORIGID	TERID	AUXID	DATE mm/- dd	TIME hh:mm	DURATION hh:mm:ss	DIGITS
N	025 01234	00 03702	DN1234	T00000		09/09	11:11	00:10:12	92345678
N	050 34567	00 OVF	DN1234	T00000		09/09	13:14	03:10:10	92348765
N	075 00045	00 OVF *1	DN2345	T00003		09/09	16:16	05:12:12	92349876

Note 1: Other call records may be inserted between these records.

Note 2: Call Charge is set at 3 units per pulse

Sample Call 9

5.10 Table 5-J gives an example involving the Periodic Pulse Metering (PPM) and Call Charge features. It shows the TN outputs when the PPM and Call Charge options are specified. Three different Normal records are shown.

- (1) The first record shows the contents of the PPM and Call Charge meters without any overflow. Note that the resulting TN output consists of two lines and the second line contains the PPM and Call Charge meter contents.
- (2) The second record shows the TTY output when the Call Charge meter overflows. Note that only the PPM count and OVF *are* printed.
- (3) The last record shows the TN output when the PPM meter overflows once. In this case the second line consists of the residual PPM count and OVF ***X**, where X is the number of overflows. Note that the Call Charge meter contents are not printed.

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

SINGLE/MULTI-PORT CDR STORAGE SYSTEMS DESCRIPTION

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* SL-1 is a trademark of Northern
Telecom Limited

1. GENERAL

1.01 This practice contains a general description of the features and equipment available with the Single/Multi-Port Call Detail recording (CDR) systems. These systems are used to store Call Detail information generated by SL-1 machines equipped with the CDR software package.

1.02 A more detailed description of the SL-1 CDR software package and the Single/Multi-Port CDR storage systems involves the following documents:

SL-1 Call Detail Recording Feature:

553-2631-100 Description

Single/Multi-Port CDR Systems:

553-2631-210 — Installation

553-2631-310 — Operation

553-2631-510 — Maintenance and Fault Clearing

Equipment Manuals:

Hewlett-Packard — 7970B/E Magnetic Tape Units

Hewlett-Packard — Hand-Held Deguasser

Deltec — Static Inverter : Model D1-702

2. DESCRIPTION

CDR EQUIPMENT	2.01 The Single or Multi-Port CDR storage systems consist of a single QCA11 equipment cabinet. This cabinet contains a Central Processing Unit (CPU), a 9-track Magnetic Tape Unit (MTU) and associated tape control circuitry for the storage of call records, plus various power and cooling equipment.
Input/Output Interface	<p>2.02 Call detail information from the SL-1 machine is passed to the Single/Multi-Port CDR system in one of two ways (Fig. 2-1).</p> <ul style="list-style-type: none"> ● Local CDR — A single 25-wire cable connects the SL-1 to a CDR system located within a distance of 50 ft (15.2 m). This cable connects the Serial Data Interface (SDI) circuit packs installed in both the SL-1 and CDR systems. ● Remote CDR — This connection requires an SDI-modem pair in both the SL-1 and CDR systems. <p><i>Note:</i> CDR records may also be output by an SDI pack to a local or remote teletypewriter (TTY) or RS232-C compatible device.</p>
Central Processing Unit (CPU)	<p>2.03 Each CDR cabinet has a CPU to control the operation of the CDR system and to provide fault monitoring and interrupt supervision. The CPU consists of three circuit packs located in the Common Equipment (CE) shelf. Associated with the CPU is a Read Only Memory (ROM) for storing the system firmware and a Random Access Memory (RAM) for buffering CDR records.</p> <p>2.04 The call records received from the SL-1 systems are stored in the RAM and then transferred to the MTU for storage in block format. Each block contains data for one SL-1 installation.</p>
Magnetic Tape Unit (MTU)	<p>2.05 The MTU is a read/write device used to store the call records received from the SL-1. The unit is controlled by two circuit packs located in the CE shelf (Fig. 2-2).</p> <p>2.06 Data is recorded on the 9-track magnetic tape at 1600 cpi in industry standard phase encoded format. Tapes are written at 25 ips.</p>

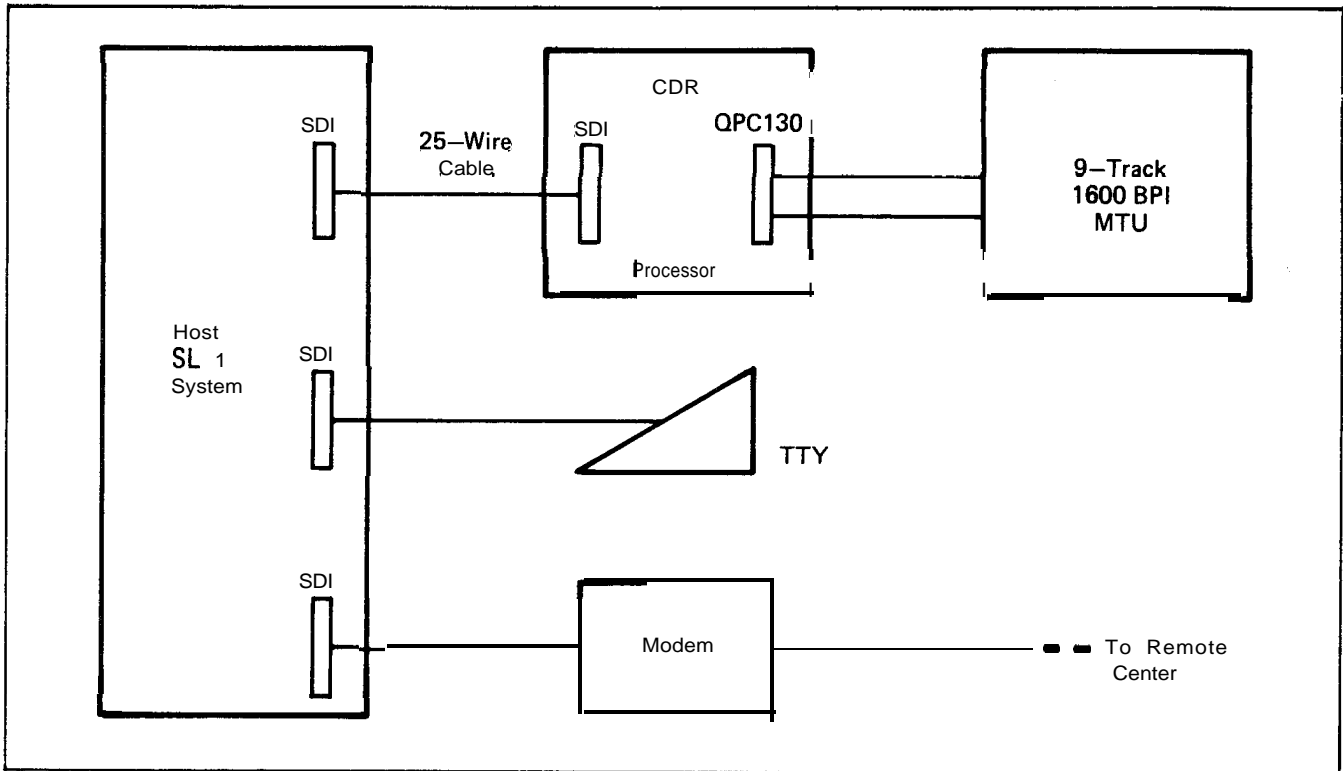


Fig. 2-1
Block Diagram of the CDR System

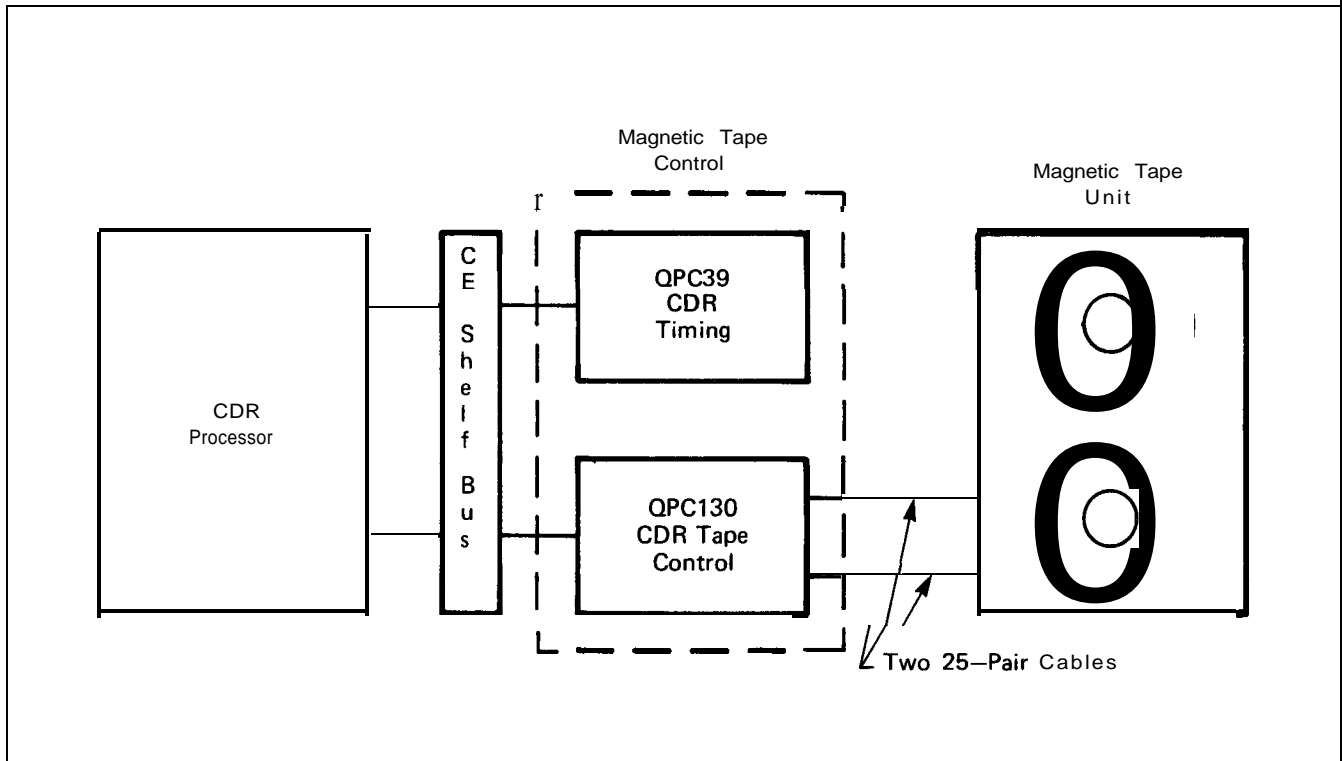


Fig. 2-2
Block Diagram of Magnetic Tape System

TAPE FORMAT

2.07 Data recorded on magnetic tape is organized into blocks (Fig. 2-3), each containing data from a single SL-1 system. Tape blocks from various SL-1 systems are randomly interleaved on the tape. Each tape block is 2048 S-bit bytes long.

2.08 A tape block consists of a block number (2 bytes), a system identification (SYSID, 2 bytes), and a series of call records. All blocks other than tape marks contain valid CDR data; any data following the tape mark should be ignored.

2.09 The second word in a block contains the port number and the SL-1 system identification (SYSID). The port number (upper four bits) depends entirely on the address switches on the SD1 card in the CDR cabinet and may have any value from 0 to 15. The SYSID (lower 12 bits) identifies the SL-1 system that originated the data in the tape block. Each SL-1 system is only guaranteed to send the CDR its SYSID once per midnight, so it is possible to have a SYSID of 0 for large numbers of tape blocks after a CDR initialization. A downstream processor can, however, determine each SL-1's SYSID, because the SYSID is always sent eventually and the port numbers remain unchanged.

2.10 The 5th byte in the block is always the first byte in a call record. A zero in the first byte of a call record indicates a null record; which may be used to pad a tape block if another complete call record doesn't fit into the block. Call records do not span tape blocks. See Fig. 2-3.

2.11 Refer to 553-2631-100 for a complete description of the types of call records which may constitute a tape block. A CDR Tape Information package is available for those who prefer to write their own software, rather than send the CDR tapes to a downstream processor. By Quoting order number PO572490 a customer receives a sample tape with actual call data, plus documentation describing tape format and how the call records were generated.

POWER REQUIREMENTS

2.12 The -48 V dc system power is supplied from a rectifier assembly located in the CDR cabinet, or by direct cabling from a host SL-1 system. A Deltec Static Inverter is available to convert the -48 V dc of the system rectifier or reserve battery into 115 V ac to power the MTU and cooling unit. The power requirements of the CDR system with inverter is 16 A, without inverter is 5 A.

2.13 The type of power equipment required depends on the configuration of the CDR system as outlined in Part 4 of this practice.

CDR ALARMS

2.14 Low Tape Alarm. A contact closure is provided for a remote low tape alarm. The contact closes when 11,000 blocks of data have been written onto the tape. This is equivalent to filling about 90 percent of a 2400 foot reel. It is up to the operating company to provide a suitable type of alarm. Circuit limitations are 30 V A; with a maximum 1 A dc current or 100 V dc maximum, or any value within these limits. When using a smaller sized reel than the 2400 foot one, the low tape alarm will not be activated.

2.15 Service Alarm. A contact closure is provided for using a remote service request alarm. The contact closes when CDR is unable to write onto the tape. This can occur when an end-of-tape is encountered, after a power failure in the CDR tape controller or tape unit, or when a tape restore operation cannot be completed. Again, it is up to the operating company to provide the alarm. Circuit limitations are 30 V A, with 1 A maximum dc current or 100 V maximum.

Note: The CDR machine must be powered and the Central Processing Unit (CPU) operating, before the alarms can function.

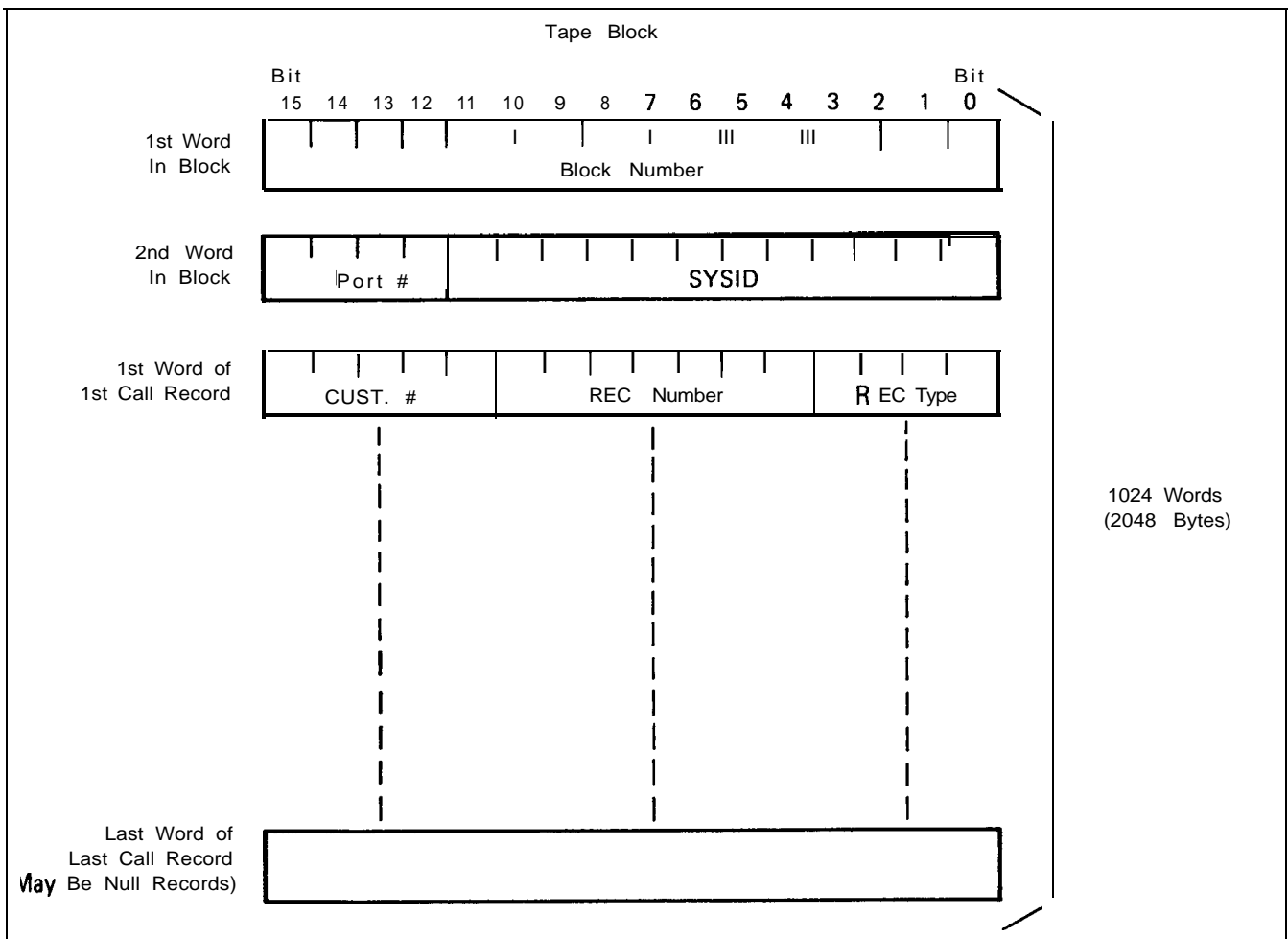


Fig. 2-3
Format of a Tape Block

CDR CAPABILITIES AND REQUIREMENTS

2.16 The number of customers and SL-1 machines that can be handled by a Single or Multiport CDR is subject to the following constraints:

- (a) A Single-Port CDR system can only record data from one SL-1 machine; the Multi-Port CDR system can record data from a maximum of 12 different SL-1 machines.
- (b) Up to 32 customers within a single SL-1 system may share a port to a given CDR.
- (c) An SL-1 connected to a CDR system must have the appropriate software packages. The CDR base package and CDR data-link (CLNK) option are both required, the CDR-TTY (CTY) option is also available.
- (d) An SL-1 L or VL system may have a maximum total of 8 serial data ports for CDR and other purposes (e.g., TTY). All other SL-1's can have a maximum of 16 ports.

LIMITATIONS

2.17 The capacity of the CDR system to record call detail information is limited by two factors:

2.18 **Transmission Rates.** The maximum rate at which information may be transmitted to the serial data devices or to the CDR system limits the number of calls/hour about which the system can record information. Tables 2-A and 2-B show a translation between the serial data rates supported by the SL-1 QPC45 and QPC139 SDI pack and the rate at which recorded calls may be terminated. The calling rates shown in Tables 2-A and 2-B may be compared to the SL-1 line size in Table 2-C. In this way, the CDR call recording capacity may be determined for any particular system under 7000 lines.

2.19 The calling rates for various system sizes shown in Table 2-C are calculated as follows:

- daily rate = hourly rate x 8.1 x 1.2
- monthly rate = daily rate x 23

Note: The figures 8.1, 1.2 and 23 have been determined by traffic studies.

2.20 **Storage Capacity.** The capacity of the storage device limits the call recording capacity of the system. For example:

- The capacity of hard-copy terminal is limited only by paper supply. A 300 baud terminal can support a random calling rate of 1150 calls/hour, and each 11 -inch page can accommodate 66 call records.
- The capacity of the MTU is limited by tape length. Thus, with a tape length of 2400 feet (732 m) the tape capacity is 1.2×10^6 call records. A tape 1200 feet (366 m) long can accommodate 6×10^5 call records.

Loss of Call Records

2.21 Power failures and certain kinds of equipment failures which cause the CDR machine to "initialize" can cause data buffered in the CDR memory to be lost. Under normal circumstances, the loss for each SL-1 system connected to the CDR will be, on average, 1022 bytes (approximately 50 call records), or at worst, 2044 bytes (approximately 100 call records). The loss probability distribution is nearly uniform in the range 120 to 1924 bytes.

Table 2-A
ASCII DATA RATES — FOR HARD COPY

DATA RATE (baud)	UNIFORM CALLING RATE (calls/hour)	RANDOM CALLING RATE (calls/hour)
110	550	385
300	1650	1150
1200	6600	4620
2400	13200	9240
4800	26500	18480
<p><i>Note:</i> For a given baud rate, the RANDOM calling rate represents a 'real world' estimate of calling rates that the CDR system can support. The UNIFORM rate on the other hand represents a theoretical upper limit to the CDR recording rate. Uniform rates are used for comparison purposes, while Random rates are used when a realistic figure is needed.</p>		

Table 2-B
BINARY DATA RATES — FOR MTU

DATA RATE (baud)	UNIFORM CALLING RATE (calls/hour)	RANDOM CALLING RATE (calls/hour)
110	1900	1330
300	5700	3990
1200	23100	15960
2400	46200	31900
4800	92400	63800

Table 2-C
 AVERAGE BUSY SEASON BUSY HOUR (ABS BH) CALLING RATES

SL-1 LINE SIZE	IN CALLS (hour)	OUT CALLS (hour)	TOTAL CALLS (hour)	IN CALLS (day)	OUT CALLS (day)	TOTAL CALLS (day)	IN CALLS (mon.)	OUT CALLS (mon.)	TOTAL CALLS (mon.)
13	38	41	109	369	399	1059	8495	9166	24067
26	50	54	145	486	525	1409	11178	12072	32016
39	62	67	181	603	651	1759	13806	14979	39964
52	74	80	217	719	778	2109	16543	17885	45913
65	86	92	253	836	894	2495	19226	20568	55862
78	97	104	289	943	1011	2809	21685	23250	63811
91	108	117	325	1050	1137	3159	24144	26157	71760
104	120	129	362	1166	1254	3519	26827	28839	79929
117	131	141	398	1273	1371	3869	29286	31522	87878
130	142	153	435	1380	1487	4228	31746	34205	96048
195	194	212	619	1886	2061	6017	43395	47395	136675
260	244	268	804	2372	2605	7815	54549	59914	177523
325	290	322	989	2819	3130	9613	64832	71986	218371
390	334	375	1174	3264	3645	11411	74669	83835	259219
520	416	477	1541	4044	4636	14979	93001	106638	340252
650	491	577	1901	4773	5608	18768	109768	128994	419740
780	566	684	2250	5502	6648	21870	126535	151027	496800
1040	707	913	2922	6872	8874	28402	158057	201590	645177
1300	854	1189	3551	8301	11557	34516	190920	262531	784060
1950	1070	1854	5237	10400	18020	50904	239209	409363	1156329
2600	1300	2454	6700	12636	23853	65124	290628	541843	1479360
3250	1479	3084	8058	14376	29967	78324	330645	680947	1779206
3900	1640	3698	9310	15941	35945	90493	366638	816518	2055648
5000	1877	4862	11381	18244	46909	110623	419622	1078900	2512924
5850	2008	5735	12877	19518	55744	125164	448908	1266288	2843241
6500	2099	6517	14062	20402	63345	136682	469246	1456935	3143686

3. EQUIPMENT IDENTIFICATION

3.01 This part describes the various components used in a Single or Multi-Port CDR storage system.

COMMON EQUIPMENT

QCA11 Cabinet

3.02 Purpose. To hold CDR equipment, cable harness, and Terminal Connecting Block (TCB).

3.03 Features.

- Welded steel construction, with all equipment accessible from the front
- Interconnecting cables are routed through lower sides of cabinet or around the back
- International rack mounting standards (width of 19 in [483 mm]).

3.04 Dimensions.

- Depth 20 in (500 mm)
- Width 31 in (785 mm)
- Height 72 in (1830 mm)

3.05 Approximate Weight. 850 lb (385 kg) when fully equipped.

3.06 Quantity. One cabinet per CDR system.

3.07 Components.

- Cabinet and framework
- Removable front, side, top, and back panels
- Cable harness
- Metal box at base of cabinet to support Deltec inverter.

Magnetic Tape Unit

3.08 Purpose. This unit records call data on 9-track magnetic tape at 1600 BPI under Common Equipment control. Consists of a Hewlett-Packard 7970B/E Digital Magnetic Tape Unit.

3.09 Quantity. One per CDR cabinet.

3.10 Location. Top of QCA11 cabinet.

3.11 Features.

- Standard 110 V ac input
- Read/write capability
- Two tape spools
- One empty reel located on lower spool.

3.12 Approximate Weight. 150 lb (68 kg).

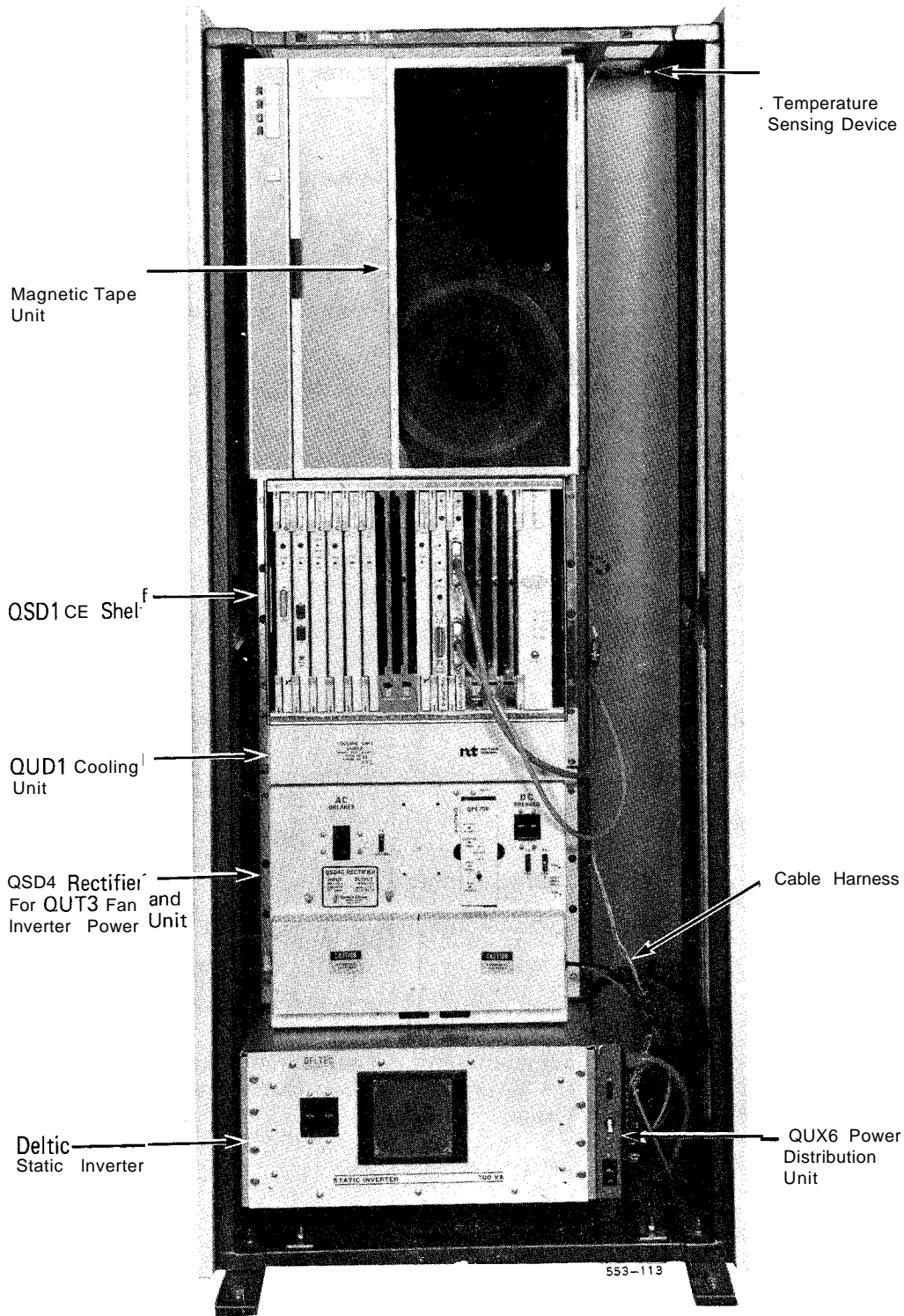


Fig. 3-1
QCA11 CDR Cabinet

QSD1 Common Equipment Shelf	3.13 Purpose. To accommodate CDR, CPU circuit packs, MTU control packs, memory packs, SDI packs and 5/12 V converter. The 5/12 volt converter circuit pack is located in the double slot on the right-hand side of the shelf. All other circuit pack positions are flexible, although they are usually arranged to simplify the faceplate cable attachments.
	3.14 Quantity. One per CDR system.
	3.15 Location. Below MTU in QCA11 Cabinet.
	3.16 Features. <ul style="list-style-type: none"> ● Steel and aluminum construction ● Printed circuit backpanel ● Fully connectorized power and signal connections ● International rack mounting standards (width of 19 in [48.3 cm]).
	3.17 Approximate Weight. 35 lb (16 kg) fully equipped.
	3.18 Components. Shelf and backpanel.
Central Processing Unit	3.19 Purpose. To control the CDR system; provides interrupt supervision and fault monitoring.
	3.20 Components. Three circuit packs: <ul style="list-style-type: none"> ● QPC40 — Arithmetic Logic Unit ┆ QPC41 — Miscellaneous circuit pack ┆ QPC42 — Sequencer.
	3.21 Location. QSD1 CE shelf.
	3.22 Quantity. One CPU is required for each CDR system.
Read Only Memory (ROM)	3.23 Purpose. To provide storage for CDR system firmware.
	3.24 Components. See Table 3-A.
	3.25 Location. QSD1 CE shelf.
Random Access Memory (RAM)	3.26 Purpose. Provides memory for buffering CDR call records. The QPC31 provides 8k of RAM, the QPA62 provides 32k. A CDR with one port requires either a QPC31 or a QPA62; a system with more than one port requires a QPA62.
	3.27 Location. QSD1 CE shelf.
	3.28 Quantity. One RAM circuit pack per system.
	3.29 Features. 8k or 32k words by 17 bits (including parity).

Table 3-A
ROM CIRCUIT PACKS

CDR SYSTEM	ROM PACKS
Single-Port	QPC131 CDR ROM1 QPC132 CDR ROM2 (both required)
Multi-Port	QPC234 CDR ROM1 QPC235 CDR ROM2 (both required) or QPC301 CDR ROM (replaces QPC234/235 pair)

- CDR Tape Control Circuit Pack
- 3.30 Purpose. To translate data, status, and commands between the CPU and MTU.
 - 3.31 Components. A single QPC130 circuit pack.
 - 3.32 Location. QSD1 CE shelf.
 - 3.33 Quantity. One circuit pack per CDR system.
- CDR Timing Circuit Pack
- 3.34 Purpose. To provide timing signals to CDR tape control circuit pack. Provides a real time clock for CDR system.
 - 3.35 Components. A single QPC39 circuit pack.
 - 3.36 Location. QSD1 CE shelf.
 - 3.37 Quantity. One circuit pack per CDR system.
- QPC45 SDI Pack
- 3.38 Purpose. This pack converts the parallel data in the SL-1 or CDR CPU into serial format for transmission. The SDI pack is capable of two way transmission along a 25 pair cable to another SDI, or through a modem pair.
 - 3.39 Components. A single QPC45 circuit pack equipped with a 25-pin faceplate EIA connector to attach the transmitting cable.
 - 3.40 Location. Any position on the QSD1 CE shelf.
 - 3.41 Quantity. See QPC139.
 - 3.42 Features. Data rates of 110, 300, 1200, 2400, and 4800 baud may be selected by the switches located on the circuit board.

QPC139 SDI Pack

3.43 Purpose. Same as QPC45, with two port operation.

3.44 Components. A single QPC139 circuit pack, with two 25-pin EIA connectors to attach the transmitting cables.

3.45 Location. Any position on the QSD1 CE shelf.

3.46 Quantity. Any combination of circuit packs QPC45 and QPC139 may be used, as long as the total number of SDI circuit packs is six or less. Multiport CDR therefore supports up to 6 ports when QPC45 SDI's are used, or up to 12 ports when QPC139 SDI's are used. Single-Port systems require only one SDI pack.

3.47 Features. Two-port operation with data rates of 110, 150, 300, 600, 1200, 2400, 4800 and 9600 baud.

QPC164 Bus Termination Units

3.48 Purpose. To terminate CE bus signals correctly.

3.49 Components. A single plug-in circuit pack QCP164*3, where * indicates the vintage. Use QPC164D3 or later vintage.

3.50 Location. On the backplane of the CE shelf.

3.51 Quantity. One Bus Termination Unit per CDR system.

POWER EQUIPMENT**Deltec Static Inverter**

3.52 Purpose. To convert -48 V dc of the rectifier assembly or reserve battery into 115 V ac to power the MTU and QUD1 cooling unit.

3.53 Location. In a metal box at the base of the CDR cabinet.

3.54 Quantity. One per CDR system, if required by the operating company.

QPC75 Rectifier Control Circuit Pack

3.55 Purpose. To provide phase control, monitoring and regulation of the QSD4 rectifier.

3.56 Location. QSD4 rectifier.

3.57 Features. Achieves a regulation of one percent from no load to full load with a 10 percent variation of input.

3.58 Quantity. One QPC75 is required in each QSD4 rectifier.

QPC85/190 5/12 V Converter

3.59 Purpose. Converts -48 V dc to ± 12 V and +5 V dc supplies for the CE shelf.

3.60 Components. A single QPC85 or QPC190 circuit pack per CDR system.

3.61 Location. Right-hand side of CE shelf.

3.62 Quantity. One per CDR system.

3.63 Features. Magnetic overcurrent and short circuit protection.

QSD4, QRF6, QRF8 48 V Rectifier Assembly

3.64 Purpose. Accommodates transformer and rectifier assembly to convert 115- or 220 Vac, 50- to 60-Hz single phase commercial power supplies to -52.08 V dc (nominal).

3.65 Components. Power shelf assembly ready for mounting in cabinet, except for the QPC75 control circuit pack required in the QSD4 rectifier.

3.66 Location. Below QUD1 fan unit in CDR cabinet.

3.67 Quantity. One assembly required per CDR cabinet, if included in operating company option.

3.68 Features. Strap conversion from 115 through 220 V working. Maintains delivery of power at 42 V during momentary fluctuations in commercial input supply voltage of less than 100 ms.

3.69 Approximate Weight.

QSD4 — 200 lb (91 kg)
QRF6/8 — 85 lb (40 kg)

QUT3 Fan Inverter and Power Unit

3.70 Purpose. To distribute 115 V ac commercial power to fans in cooling unit. Includes inverter which supplies 115 V ac from -48 V dc. The unit is used when the CDR is powered from the host SL-1 -48 V power supply without battery reserve, and when there is a possibility that the host SL-1 -48 V power supply could still be on when the CDR commercial supply fails.

3.71 Location. In CDR cabinet below QUD1 cooling unit.

3.72 Quantity. One per CDR system when required.

QUX6 Power Distribution Unit

3.73 Purpose. To provide interconnection for signaling and power leads to and from various units within the CDR cabinet.

3.74 Location. Right-hand side of the metal box that holds the Deltec Static Inverter (bottom of the cabinet).

3.75 Components.

- One alarm terminal block
- One power terminal block
- One commercial power outlet
- One 0.5 fuse
- One input switch (-48 V and 35 A)
- One CB2 10-A breaker.

48 V Reserve Power Supply (J-87122)

3.76 Purpose. To maintain service in the event of a commercial power failure.

3.77 Location. Within the host SL-1 system and connected, if required, by wires to the QUX6 unit at the base of the CDR cabinet.

CABLE AND MISCELLANEOUS
EQUIPMENT

- COO52743 Hand-Held De-gausser** 3.78 Purpose. To degauss tape heads, thus reducing random data errors caused by residual magnetism.
- 3.79 Location. Kept separate from the cabinet.
- 3.80 Quantity. One per CDR system.
- 3.81 Size. Approximately 1 in (24.5 mm) by 3 in (77 mm).
- NE A25MQ SDI Cables** 3.82 Purpose. Carries serial data between SDI circuit packs in the CDR system and host SL-1 systems.
- 3.83 Location. Plugged into faceplate of QPC45 and QPC139 SDI circuit packs.
- 3.84 Quantity. One per input port.
- 3.85 Length. One of two choices determined by system configuration:
- NE-25MQ2 16 ft (4.9 m)
 - NE-25MQ3 50 ft (15.2 m)
- 3.86 Features. The 25 wire AWG; standard Q1A teletype cable connectorized at both ends.
- QCB12 and QCB13 Cables** 3.87 Purpose. To connect the tape control circuitry to the MTU
- 3.88 Location. Runs from the back of the MTU to the QPC130 tape control faceplate.
- 3.89 Quantity. One QCB12 and one QCB13 per CDR system.
- 3.90 Length. 8 ft (2.6 m).
- QUD1 Cooling Unit** 3.91 Purpose. To dissipate heat generated by the CE shelf.
- 3.92 Components. Shelf, fans and Power connection cord.
- 3.93 Location. Immediately below the CE shelf.
- 3.94 Features. Two removable fan units, fused and operating from the commercial power supply, the Deltec static Inverter, or from the QUT3 power unit.
- 3.95 The fans will operate during a commercial power failure under the following conditions:
- (a) if power is supplied via the Deltec inverter from a reserve battery, or
 - (b) if power is supplied via a QUT3 unit from an SL-1 -48 V.
- 3.96 These power connections ensure that the fans operate whenever a -48 V power source is connected to the CE cabinet.
- 3.97 Approximate Weight. 10 lb (5 kg).
- 3.98 Quantity. One unit required per CDR cabinet.

4. ORDERING INFORMATION

4.01 The CDR system can have several configurations, depending on how it is to be powered. The CDR cabinet can be powered by its own rectifier or by the host SL-1 system. A battery reserve may also be connected to the CDR system.

4.02 The equipment that should be ordered with each of the available options are outlined in Table 4-A and 4-B. Table 4-C lists the additional equipment that is required.

Table 4-A
 OPTION A — CDR WITH RESERVE BATTERY

DESCRIPTION	EQUIPMENT
<p>Option A1 — CDR powered by -48 V from its own rectifier</p> <p>Option A2 — CDR powered from the host SL-1 system. (Note 2)</p>	<p>MTU CE Shelf (Note 1) QUD1 Cooling Unit Rectifier Deltec Inverter</p> <p>MTU CE Shelf (Note 1) QUD1 Cooling Unit Deltec Inverter</p>
<p><i>Note 1:</i> The CE shelf is equipped with the CPU, tape control and power circuit packs. Refer to Table 4-C for the additional equipment required.</p> <p><i>Note 2:</i> It may be necessary to have an additional rectifier in the host SL-1 system, depending on the system load already there.</p>	

Table 4-B
OPTION B — CDR WITHOUT BATTERY RESERVE

DESCRIPTION	EQUIPMENT
<p>Option B1 — CDR powered from its own rectifier; the MTU, cooling unit and rectifier all powered from commercial 115 V ac outlets.</p>	<p>MTU CE Shelf (Note) QUD1 Cooling Unit Rectifier</p>
<p>Option B2 — CDR powered from host SL-1 system; MTU powered from commercial 115 V ac outlet.</p>	<p>MTU CE Shelf (Note) QUD1 Cooling Unit QUT3 Fan Inverter Power Unit</p>
<p><i>Note:</i> The CE shelf is equipped with CPU, tape control and power circuit packs. Refer to Table 4-C for additional equipment required.</p>	

Table 4-C
ADDITIONAL EQUIPMENT REQUIRED

EQUIPMENT	QUANTITY
QPC45 or QPC139 SDI circuit packs	6 max.
QPC31 or QPA62 RAM circuit pack	1
NE-A25MQ SDI cable 16, 25 or 50 ft (4.9, 8.2 or 16.4 m) long	1 per SDI port

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

CDR EQUIPMENT DESCRIPTION
(COMPLIANCE **WITH** FCC REGULATIONS)

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Battery Distribution Box	2-2

* SL-I is a trademark of Northern Telecom Limited

1. GENERAL

1.01 This appendix gives descriptive information about CDR cabinets and equipment that comply with FCC regulations concerning electromagnetic interference (EMI).

1.02 The information given in this appendix takes precedence over the information given in 553-2631-110 for similar items of equipment.

2. IDENTIFICATION

CABINETS

- Cabinet Door
- 2.01 The cabinet front panel is a hinged metal panel equipped with a key operated locking mechanism. A key for operating the lock is supplied with each cabinet.
- 2.02 The door panel is equipped with metal gasketing strips rivited to the inside edges to ensure metal to metal contact between the cabinet frame and the door.

Note: Plexiglass or glass-type doors should not be used with these cabinets.

- Top Panel
- 2.03 The top panel (I/O panel) of the cabinet serves as the primary connecting point to the equipment within the cabinet. The panel contains filter connectors required to interface with Serial Data Interface ports (RS-232 connections). Filter connectors can be added to the top panel as required and are connected to the equipment within the cabinet with flat interface cables.

INTERFACE CABLES

- Alarm and Miscellaneous Equipment
- 2.04 Alarm and miscellaneous equipment cables use the same type of interface cables as PE shelves. They terminate on filter connectors on the I/O panel where they are extended to the cross-connect terminal as required.
- Serial Data Interface Packs
- 2.05 The QPC45 and QPC139 Serial Data Interface (SDI) packs are connected with a QCAD42 interface cable to a filter connector located on the I/O panel of the cabinet. All links, modems and terminals connect to the filter connector on the outside top of the I/O panel.

INTER-CABINET CABLING

- Adjacent Cabinets
- 2.06 Four cable access holes are provided in each cabinet side and rear panel which are used for inter-cabinet cabling when cabinets are installed side-by-side or back-to-back. The cabinets are secured to each other by a bulkhead assembly installed in each adjacent access hole.

POWER AND GROUND WIRING

- Cabinet Equipped With a Rectifier
- 2.07 The power supply cable from the rectifier is run in a flexible conduit terminated at the cabinet top panel. The commercial power supply (as described in 553-2yy1-200) must be installed within 3 ft (915 mm) of the cabinet top panel.
- Reserve Battery
- 2.08 All -48V battery distribution and charge wires are run in metallic electrical conduit, maximum 3/4 in=(20 mm) in diameter.
- 2.09 Charge and -48V power leads are routed through access hole marked DC PWR on the cabinet top panel. Isolated ground bus (systems without reserve battery) and i-48V leads are routed through the access hole marked GND.

Battery Distribution
Box

2.10 If possible, the Battery Distribution **Box** should be located in such a manner that no more than 10 ft (3050 mm) of wiring is required between the connection in the box and the connection within the cabinet. This permits up to three 6 AWG wires to be run in the 3/4 in conduit.

Note: Approximately 6 ft (1830 mm) of wiring is required within the cabinet, leaving 4 ft (1220 mm) of wiring available between the I/O panel and the connection in the Battery Distribution Box.

2.11 When the Battery Distribution **Box** is located further from the cabinets, the wire size must be increased to 4 AWG and run in a minimum 1 in conduit. The 4 AWG wire is then terminated on a Curtis Type H or equivalent terminal block located in a Hoffman type **pullbox** within 10 cable ft (3050 mm) from the cabinets being served. A 3/4 in conduit must then be provided between the **pullbox** and the cabinet being served.

BUSINESS COMMUNICATIONS SYSTEM**SL-1*****MINI CALL DETAIL RECORDING DESCRIPTION**

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* SL-1 is a trademark of Northern Telecom Limited

1. GENERAL

1.01 This practice provides a general description of the Mini Call Detail Recording (CDR) feature. A complete description of the CDR system involves the following additional documents:

SL-1 Call Detail Recording Feature:

553-2631-100 — Description

Mini CDR:

553-263 1-2 11 — Installation

553-263 1-3 11 Operation

Note: The Mini CDR feature is only available on SL-1 M systems.

2. DESCRIPTION

CDR Storage	<p>2.01 The Mini CDR is used to temporarily store SL-1 CDR call records. This feature is only available on SL-1M systems and consists of a second tape unit in the main SL-1M cabinet.</p> <p>2.02 Mini CDR has all the features and options available with the basic SL-1 CDR feature, plus the following additional capabilities.</p> <p>2.03 The CDR information is assembled by the SL-1 CDR software into various call records (refer to 553-2631-100). The records are accumulated in the CDR buffer in system memory, and then transferred as a block to the Mini CDR tape unit in packed binary format. The call records may also be sent to a teletypewriter (TTY) or any RS232-C compatible device in ASCII format, when the call record is generated.</p>
Tape Format	<p>2.04 The Mini CDR tape unit is identical to the system tape unit and is housed in the same tape shelf. Each tape unit is connected to the system by its own connector cable that runs from the front of each tape unit to a dedicated tape unit interface circuit pack in the CE shelf.</p> <p>2.05 The Mini CDR magnetic tape is formatted into four tracks of adjacent blocks, each containing 518 words.</p> <p>2.06 The tape block consists of different fields as shown in Fig. 2-1. The first and last words are used for synchronization. The second and third words contain the block number and type respectively. The block type is followed by 512 words for the storage of call records. The Checksum word is used to check for errors within the block.</p> <p>2.07 The format of call records on the tape is described in 553-2631-100. A null call record is used to fill up a block so that the individual call records do not span tape blocks.</p>
Polling	<p>2.08 The call records stored on the CDR tape can be printed on a TTY by using the Mini CDR overlay. To accomplish this polling procedure the CPU reads the CDR tape, converts the call records into ASCII format, and outputs the records to the active TTY. The CDR tape can also be polled by a computer, or a remote device through an SDI link and modem.</p>
CDR Buffer	<p>2.09 New call records generated while the CDR polling is in progress are temporarily stored in the CDR buffer. When the buffer is full, and the tape is not in a writable position, a limited number of call registers are used to store the new call records.</p>
Mini CDR Password	<p>2.10 A password is provided for those personnel who are to perform CDR polling functions. This password only allows access to the Mini CDR overlay and does not permit the user to perform changes to the SL-1 data base. This overlay may be accessed in the normal manner using the level 1 or level 2 passwords. The Mini CDR password is defined in the Configuration Record (overlay 17).</p>

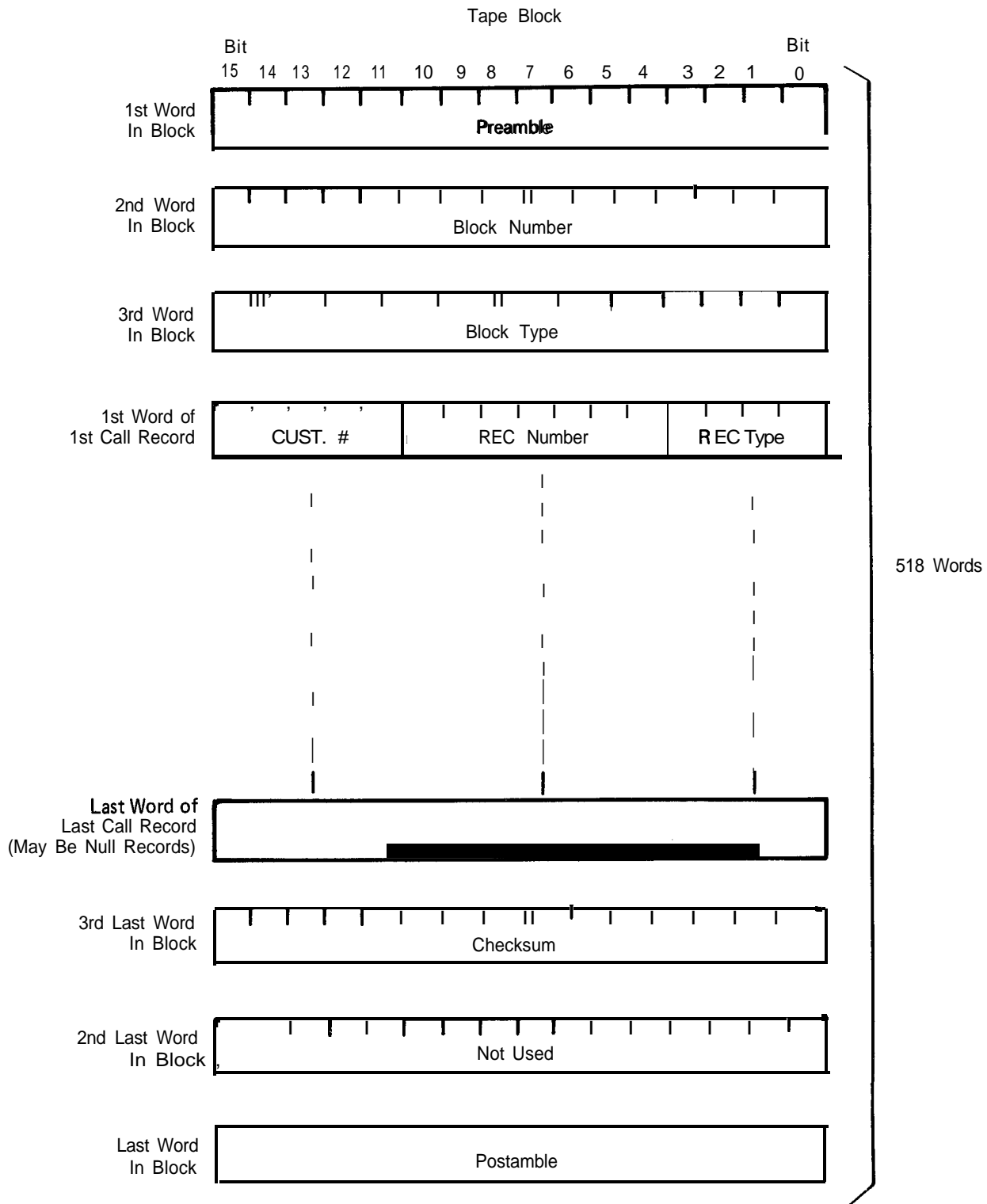


Fig. 2-1
Format of a Tape Block Record

Low Tape Alarm

2.11 This allows a flexible key/lamp on the attendant console to indicate when the Mini CDR tape is 75% full. The low tape alarm lamp becomes steadily lit when the tape is 75% full, and flashes when 100% full or when the system is reloaded. The lamp goes dark when the CDR writing on tape is started at the load point. The key associated with the low tape alarm lamp has no function.

2.12 When the tape reaches 75% of the maximum capacity, a message is printed on all maintenance TTY and an entry is placed in the history file (if set for maintenance messages).

3. EQUIPMENT IDENTIFICATION

Tape Unit (QUW9)

3.01 The tape system is a random read access and sequential write device that uses a 4 track tape cartridge. The tape drive may be either a 3M or Tandberg type, which is powered and controlled by the following circuit packs:

- QPC90 — 18 V converter
- QPC91 — 18 V regulator
- | QPC332 — tape phase encoder
- | QPC333 — tape motion control
- | QPC334B — tape write encoder

3.02 These circuit packs are located below the tape drive. The electrical interconnections between the circuit packs, the tape drive and the SL-1 M CPU are shown in Fig. 3-1.

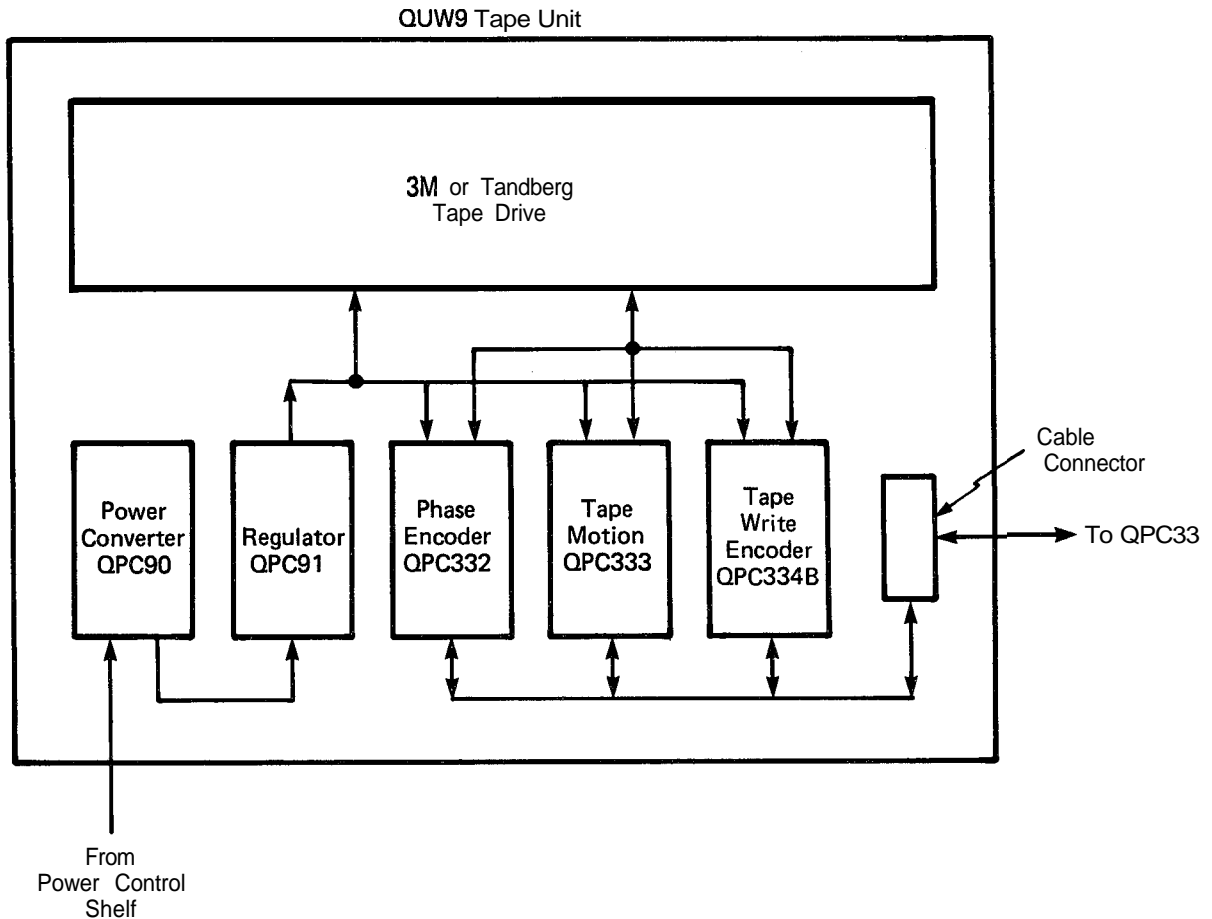


Fig. 3-1
Tape System Block Diagram

3.03 Each tape unit is separately powered by a connector cable running from the back of the tape unit to the QUX14 power harness. The voltage converter pack receives the system -48 V dc from the tape unit backplane and provides the dc voltage levels required by the remaining circuit packs and the tape drive.

3.04 The phase encoder converts the data from the CPU into the correct form for storage on the tape and vice versa. The two control circuit packs interpret signals from the CPU to control the tape drive.

3.05 The system and Mini CDR tapes are identical except for the position of a programming plug in one of the tape unit circuit packs, and the software definition of the QPC33 tape interface packs.

Caution: Do not insert the system tape cartridge into the CDR tape unit as this may cause the loss of protected system data.

3.06 If the system tape unit fails and must be replaced, then it is possible to use the CDR tape unit as the system tape unit. Although this eliminates the CDR storage capability, it allows the system to continue operating until a replacement unit is available. Refer to 553-2051-570 for this system tape unit replacement procedure.

Tape Unit Interface
(QPC33)

3.07 This circuit pack provides the interface between the SL-1 system and the tape unit. A connector cable (QCB6) runs from the front of each tape unit to an interface circuit pack. The CDR interface pack is located in slot position number 4 in the front of the CE shelf.

Tape Cartridges

3.08 The tape unit uses a 3M magnetic tape cartridge to store the CDR data. The tape records are duplicated on two tracks to allow recovery from a read error.

3.09 The tape records are written on tracks 1 and 2 until the Early Warning (EW) marker is encountered. The writing then continues on tracks 3 and 4 until the next EW is encountered. Thus the full capacity of the tape is used.

3.10 Since there is no pre-formatting requirement of the CDR tape, any physically good 3M tape cartridge may be used. This includes previously used CDR tapes or old system tapes. The old contents of these tapes are overwritten.

4. CAPABILITIES

Input/Output Ports

4.01 The number of SDI ports is limited to 16 per system. This includes all TTY or equivalent, a port to a Single/Multi-Port CDR storage system and the history file (treated as a pseudo input/output port). This does not include the tape interface to the Mini CDR tape unit.

CDR Buffer Capacity

4.02 New call records generated while the CDR polling is in progress, are stored in the CDR buffer and call registers. However, the buffer space is limited and prolonged polling activity in a high traffic period may cause the loss of new call records.

4.03 Table 4-A gives the Unprotected Data Store required to buffer CDR records generated during polling for a given polling time and traffic level. The values are calculated for a 400 line system.

Table 4-A
CDR BUFFER REQUIREMENTS

CCS	POLLING TIME (Hours)	NUMBER OF CALLS PER POLLING TIME	BUFFER SIZE (K Words)
1		142	3.83
	2	284	7.66
	10	1420	38.3
5	1	709	19.1
	2	1418	38.2
	10	7090	191.0

4.04 For example, if polling takes 2 hours and the traffic is 1 ccs, then the number of calls generated is 284 and it would take 7.66 k words to buffer the CDR records during polling.

4.05 These estimates are not necessarily additional storage required for CDR buffering. However, this buffering affects the engineering of the call registers (refer to Engineering and Assignment 553-2001-151 Appendix 7). The total required call registers is the greater of:

- (a) Call registers for maximum traffic (i.e., present engineering), or
- (b) Call registers for traffic at time of polling plus buffering required for CDR as per Table 4-A.

Polling Time

4.06 The time required to poll a full tape depends on the tape length and the baud rate of the SDI port. Table 4-B gives the polling times to output the ASCII data to a TTY or equivalent. A tape length of 3600 in or 5400 in may be used.

Table 4-B
POLLING TIMES

BAUD RATE	POLLING TIME (Hours)	
	3600 in TAPE	5400 in TAPE
110	76.4	114.5
300	28.0	42.0
1200	7.0	10.5
2400	3.5	5.25
4800	1.75	2.6
9600	0.88	1.32

4.07 From the polling times and CDR buffer requirements given in the preceding tables it can be seen that polling activity should be done at periods of low traffic. It is recommended that the baud rate of the SDI port be as high as possible and the tape length be 3600 in.

Tape Capacity

4.08 An alternative to low traffic polling would be to poll the CDR tape more frequently with a less than full tape. The time required to fill a tape depends on the tape length and the number of call records generated.

4.09 The tape capacity is 368k words (3600 in tape) or 533k words (5400 in tape) assuming the recording density is 1067 bits/in and a tape block size of 517 words.

4.10 Table 4-C estimates how long it takes to fill the CDR tape with call records for the given calling rates and the following assumptions:

- tape length — 3600 in
- line size — 400
- intercom ratio — 0.5
- call record size — 10 words (12 digits dialed)
- featured calls — 10% of total calls

Table 4-C
MINI CDR TAPE CAPACITY IN UNITS OF DAYS

CCS	INCOMING AND OUTGOING CALLS* (Calls/Day)	TAPE CAPACITY† (Days)
3	3849	8.45
4	5132	6.34
5	6415	5.08
6	7698	4.22
7	8981	3.63

* When only outgoing calls are recorded, tape capacity is doubled.

†When a 5400 in tape is used, the capacity is increased to 1.5 times the values indicated.

Tape Life

4.11 The life of the CDR tape cartridge is approximately 5000 passes. Typically, fourteen passes occur per polling operation. Assuming a polling period of five days, the life of the CDR tape is 1785 days (5 years). For other polling periods, refer to Table 4-D.

Table 4-D
CDR TAPE LIFE

POLLING PERIOD (Days)	LIFE (Days)
5	1785
4	1750
3	1315
2	714
1	357

System
Initialization

4.12 The effects of a system initialization during normal CDR recording or in the middle of physically writing to the tape, are as follows:

- The call records buffered in the system are lost. No data on the tape is lost.
- The last CDR tape block number and the track number are recovered from the protected data store.
- Normal recording continues after the system recovers.
- All Mini CDR overlay commands may be used as explained in 553-2631-311.

4.13 System initialization during CDR polling causes the termination of polling without warning or any special message to the accessor. The accessor must re-initiate the polling after system recovery.

System Reload

4.14 If the system reloads during normal CDR recording, the tape block and track numbers and CDR records buffered in the system are lost. When the system recovers the low tape alarm flashes and CDR writing to tape will have to be re-initiated by using the Mini CDR overlay. (refer to 553-2631-311).

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

SINGLE/MULTI-PORT CDR STORAGE SYSTEMS INSTALLATION

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* SL-1 is a trademark of Northern
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1. GENERAL

1.01 This practice describes the installation of the Single/Multi-port Call Detail Recording (CDR) storage system, its connection to the host SL-1 systems, commercial power requirements, and system start-up.

2. PRE-INSTALLATION CONSIDERATIONS

POWER SOURCES

2.01 The CDR system may be powered by the host SL-1 system, or by its own QSD4, QRF6 or QRF8 rectifier assembly. The rectifiers can operate on 115, 208 or 220 V ac, 50/60 Hz. The wiring options and input/output power specifications for the different rectifiers are given in Fig. 4-5(a), 4-5(b), 4-6 and 4-7. Refer to Part 4 of 553-2631-110 for a description of the various CDR configurations. ←

CDR Equipment Environment

2.02 The cabinet room should be dry, clean and well ventilated. The environment specifications in 553-2yy1-200 should be referred to before deciding on the location of the CDR system.

Final Location

2.03 If the CDR cabinet is powered with -48 V dc from a host SL-1 system, the CDR must be local to the SL-1, i.e., within 50 cable ft (15.2 m).

2.04 If the CDR cabinet is to be more than 50 cable ft (15.2 m) away from the host SL-1 system(s) the SDI cable carrying information between the two systems must be plugged into a modem pair, as shown in Fig. 2-1. Refer to Chart 3-1 for the ac outlets required for each CDR configuration.

2.05 If the CDR system is equipped with a rectifier, then the cabinet must be within 10 ft (3.05 m) of a grounded 110- or 220 V, 50 or 60 Hz ac power outlet.

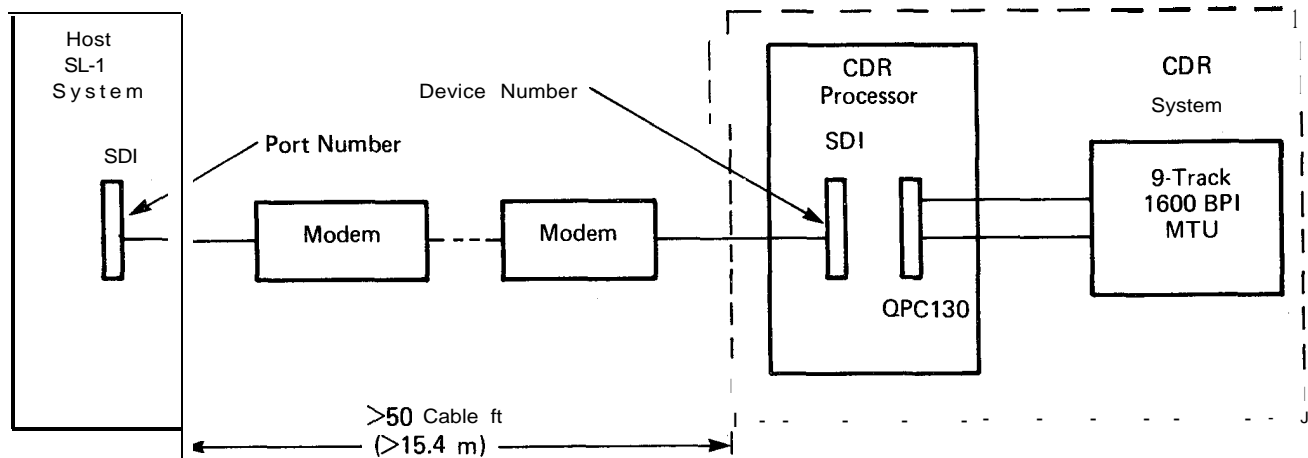


Fig. 2-1
Host SL-1 Connection to Remote CDR Cabinet

3. UNPACKING AND INSPECTION

3.01 For convenience of shipping and installation, customers receive their CDR systems in several packages.

- CDR cabinet with cables and common equipment
- Magnetic Tape Unit
- Deltec static inverter
- QUT3 fan and inverter power unit
- Rectifier Assembly
- Hand-held degausser
- Documentation

3.02 To assemble a CDR system, follow the instructions in Charts 4-1 through 4-8. The final assembled product should resemble the typical QCA11 CDR cabinet illustrated in Fig. 3-1.

3.03 The cabinet itself arrives on the installation site wrapped in protective plastic and bolted onto a wooden pallet. The customer also receives, taped to the cabinet, a plastic package containing a total of 15 self-tapping bolts and four Phillip screws. Also in the cabinet, but not yet installed in it, are a QCB12 and a QCB13 cable and a black ac line cord. Already installed in the cabinet are the CE shelf, the cooling unit, the power distribution unit, and the following circuit packs:

- QPC85 or QPC190 — 5/12 V Converter
- QPA62 or QPC31 — RAM
- QPC39 — CDR Timing
- QPC40 — Arithmetic Logic Unit
- QPC41 — CPU Misc. Registers
- QPC42 — Sequencer
- QPC45 — Serial Data Interface (Single Port) (Note 1)
- QPC139 — Serial Data Interface (2-Port) (Note 1)
- QPC130 — CDR Tape Interface
- QPC131/QPC132 — CDR Single Port ROM pair
- QPC234/QPC235 — CDR Multiport ROM pair (Note 2)
- QPC301 — CDR Multiport ROM pack (Note 2)

Note 1: The number of SDI packs required depends on CDR system configuration.

Note 2: The QPC301 ROM circuit pack may be used in place of the QPC234/235 ROM pair.

PRACTICE 553-2631-210

3.04 The remaining CDR units arrive in packaging similar to that used for SL-1 system units, (refer to 553-2001-205). The precautions mentioned there for unpacking units also apply to the CDR system.

3.05 Upon receiving the CDR equipment, customers should check that all required units have arrived. In its original packing case there should be a Hewlett-Packard MTU. Shipped with most units, but considered as options to be ordered by the customer, are the rectifier assembly, the Deltec static inverter, and the QUT3 fan and inverter power unit. Each of these units are shipped in a separate carton.

Chart 3-1
RECEIVING CDR SYSTEM

STEP	PROCEDURE
1	Unpack CDR System from packaging (refer to 553-2001-205).
2	Inspect all units for broken, disfigured, or bent parts. If there are any, contact the manufacturer.
3	Ensure that all equipment pertaining to the option ordered has been shipped (refer to 553-2631-110).
4	Check that all circuit packs are there, and secure them with the locking devices at the top and bottom. <i>Note:</i> The QPC85 or QPV190 5/12 V converter must be located in the right-hand side of the CE shelf. The position of the other packs is flexible. They are, however, positioned so that the wiring connections from the faceplates are simplified.
5	Move cabinet to proper location: within 50 cable ft (15.2 m) of host SL-1 system, and within 10 cable ft (3.05 m) of commercial 115 or 220 V, 50 or 60 Hz ac power outlets. The ac outlets required for the various CDR configurations are outlined below.
	Option A1 --- One 2320 Hubble or equivalent ac power outlet needed --- 115 or 220 V ac
	Option A2 --- No ac power outlet needed
	Option B1 --- Three ac power outlets needed --- Two standard U-ground 115 V ac; one Hubble-type 115/220 V 50/60 Hz
	Option B2 --- One standard U-ground ac power outlet needed --- 115 V ac.

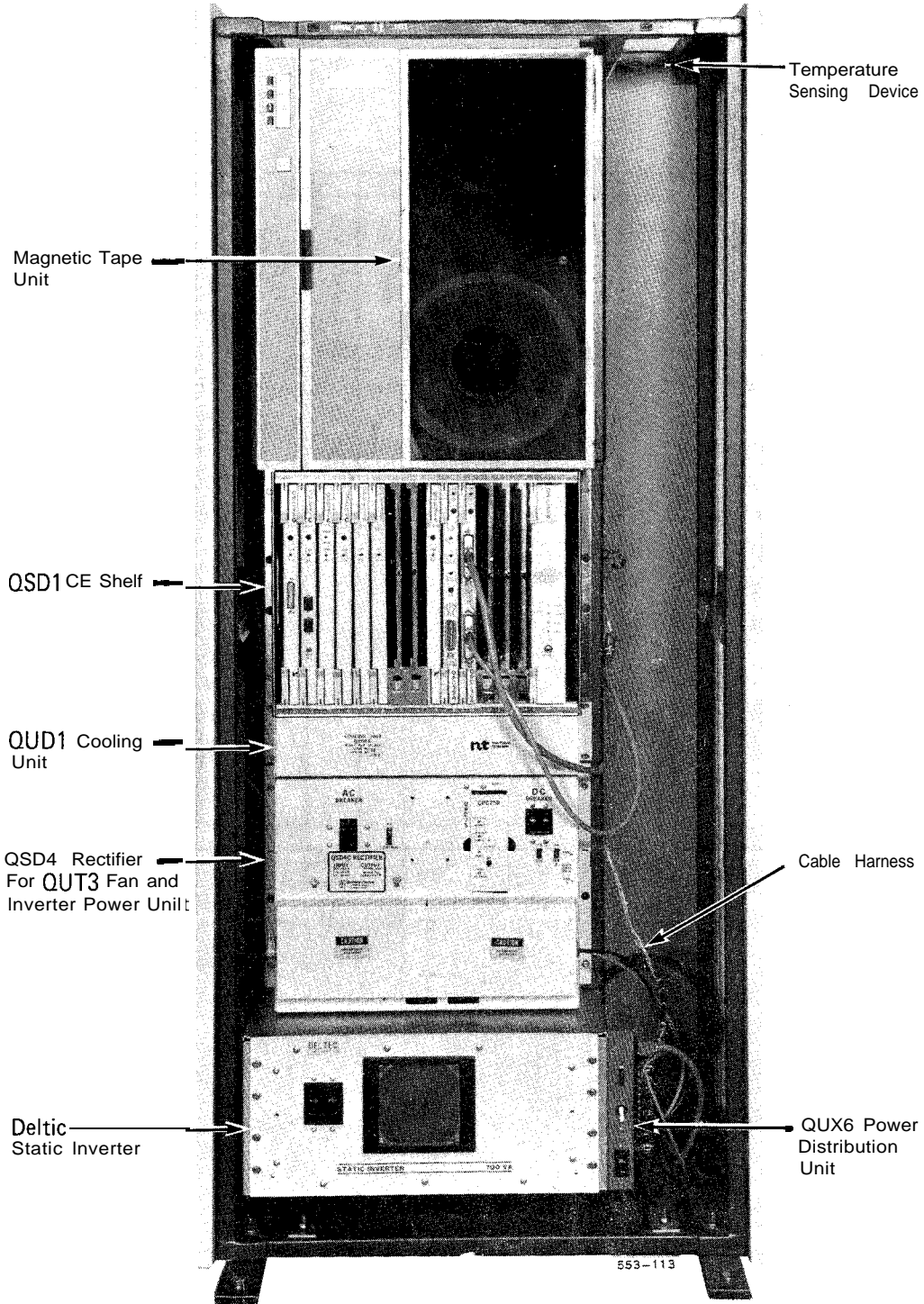


Fig. 3-1
Typical QCA11 CDR Cabinet

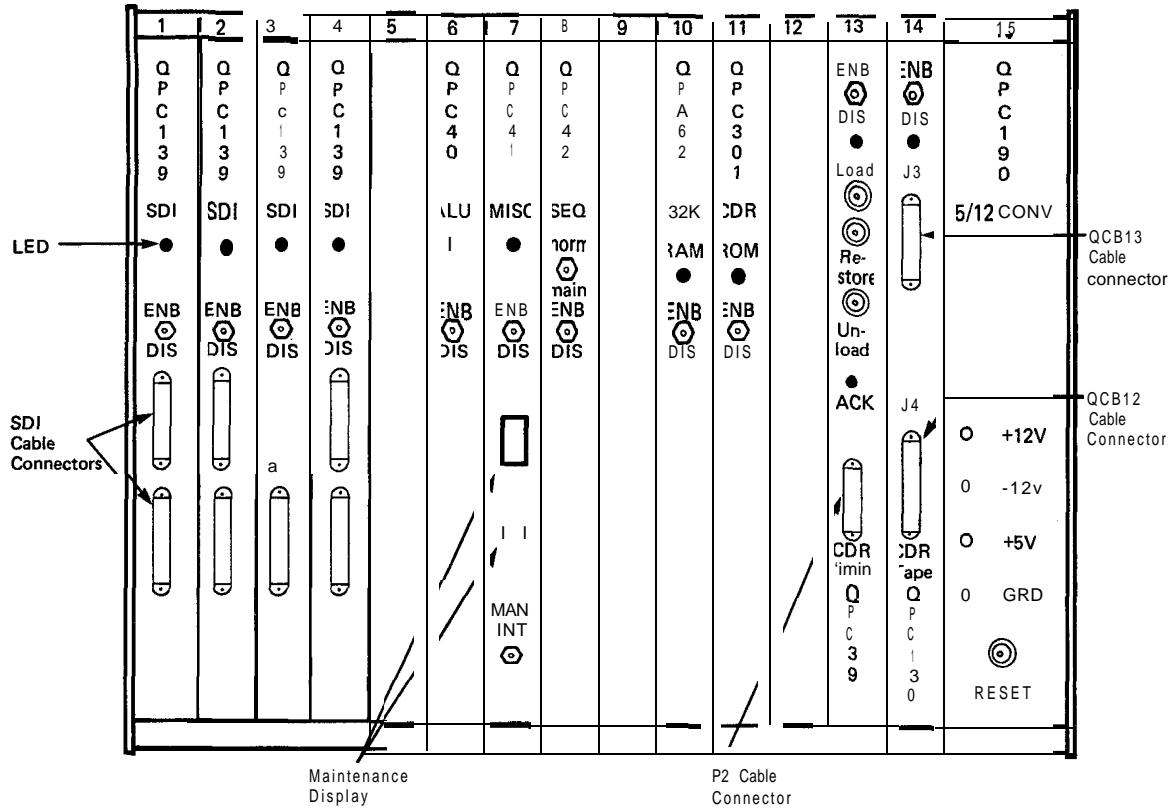


Fig. 3-2
Typical **QSD1** Shelf Configuration

4. ASSEMBLY AND START-UP

4.01 Customers are advised to assemble the cabinet, because of its weight, at the location of its intended use. Follow recommendations in 553-2yy1-200 for proper placement of the unit.

4.02 Units in the CDR cabinet are best installed just off-site with both front and backpanels removed. Charts 4-1 to 4-8 describe this type of installation and connection. After installation, the backpanel can be replaced, and the fully equipped cabinet pushed into place.

4.03 With all the units installed, the cabinet is top-heavy and has to be kept upright with stabilizing feet. If these are to be removed, then the CDR cabinet frame must be bolted either to the floor or to the adjacent cabinet frame, and a slightly different installation procedure must be followed as discussed below.

4.04 The feet must be removed off-site, since it involves tilting the cabinet backwards. This is best done while the cabinet is still unequipped. The sidepanels must be removed (see 553-2yy1-210). Then, with the backpanel left on, the cabinet is moved to its final position and bolted to the floor or adjacent cabinets. Charts 4-1 through 4-8 should still be followed step-by-step, but units with connections at the back must be connected up before being mounted into the CDR cabinet.

Chart 4-1
PREPARATION

STEP	PROCEDURE
1	Free the cabinet from its pallet by removing the two bolts on each of the two shipping brackets in the base of the cabinet.
2	Remove front and back panels of the cabinet (refer to 553-2yy1-210).

PRACTICE 553-2631-210

Chart 4-2
INSTALLING THE MAGNETIC TAPE UNIT

Caution: At least two people are required to insert this unit: it weighs 150 lb (68 kg).

STEP	PROCEDURE
1	Hold the MTU securely at top and bottom. Ease it into the upper-most position in the CDR cabinet, and rest it on the brackets provided.
2	Open the MTU dust cover door, and loosen the screw on the right side of the inner door of this unit (Fig. 4-1). Open the inner door by holding it at the top and bottom; the hinges are on the left side (Fig. 4-2). <i>Caution: Open MTU inner door carefully. The door can hit the cabinet frame.</i>
3	While one person holds this door open, insert screws through the holes in the MTU side to the holes in the CDR cabinet frame. Three self-tapping hex screws go into the right side of the MTU, and four Philips screws go into the left, as shown in Fig. 4-2. Alternate sides when tightening screws.
4	Feed the QCB12 cable marked CONNECTOR END ASSEMBLY into the tape unit through the centre opening in the back of this unit. Mate this end to the contacts of connector 'G' in the top right-hand corner of the MTU (Fig 4-2).
5	The other end of the QCB12 cable is then connected to J4 on the face of the QPC130 pack in the CE shelf. Then secure the two screws on J4 to hold the cable in position.
6	Take the double-ended QCB13 cable assembly, and feed the 2-edge connectors E and F through the opening at the back of the MTU in the same manner as for the QCB12. Connect as shown in Fig. 4-2. The other end of the QCB13 is connected to J3 of the QPC130 pack on the CE shelf. Check the fuses in the MTU. See Hewlett-Packard manual for location.
→ 7	Make sure that the three toggles switches on the MTU internal circuit board on the inside of the MTU door are all in the DOWN position. Close the inner MTU door; tighten the latch screws. Close the door of the dust cover.
8	Take the P2 gray line cord from the cable harness, and plug this into the jack on the face of the QPC39 pack (Fig 4-13).

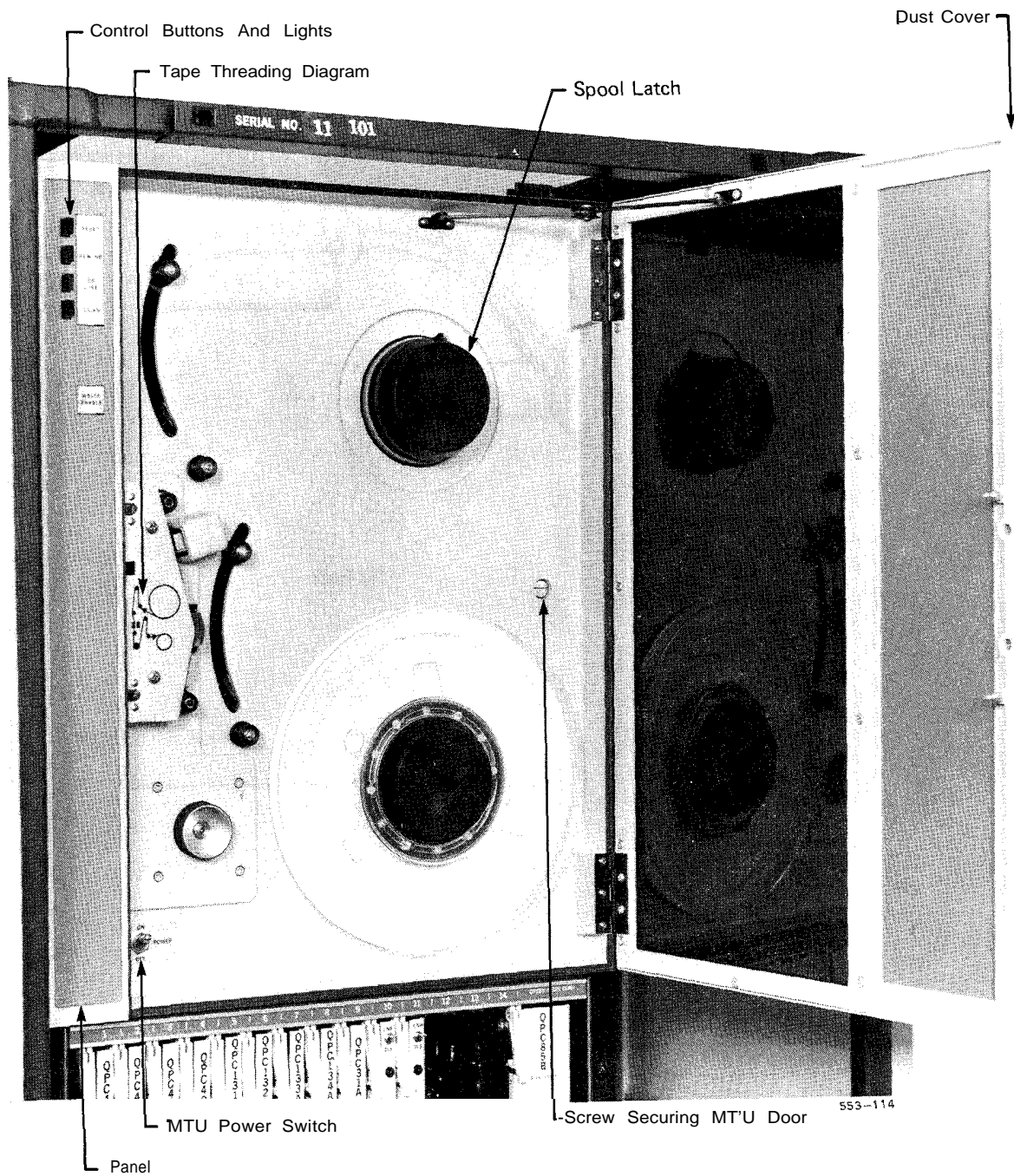


Fig. 4-1
Front View of Magnetic Tape Unit (Dust Cover Open)

PRACTICE 553-2631-210

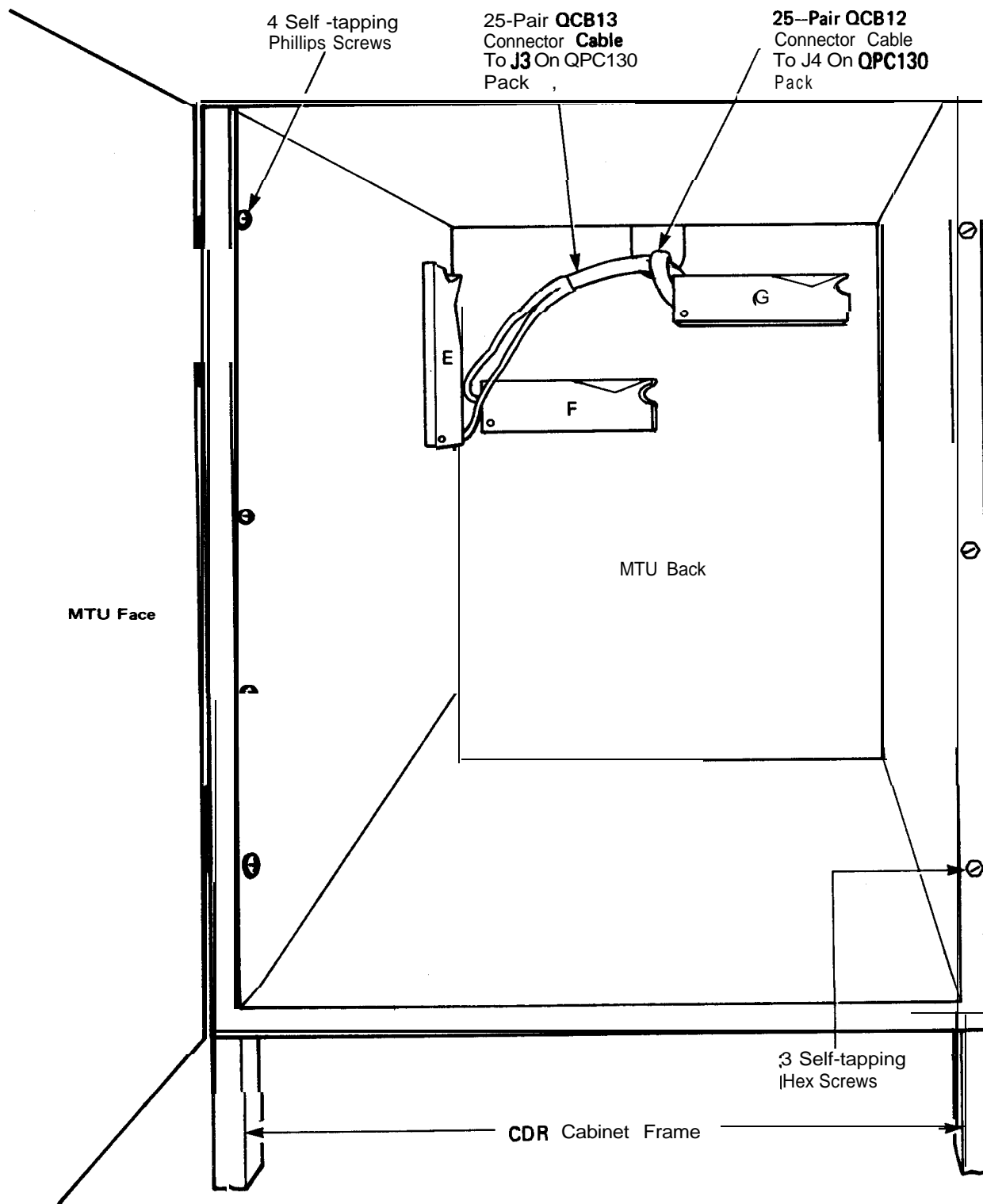


Fig. 4-2
Front View of Magnetic Tape Unit (Inner Door Open)

Chart 4-3
INSTALLING THE DELTEC STATIC INVERTER
(OPTION A1 OR A2)

STEP	PROCEDURE
1	Lift the inverter, and slide it into metal box provided for it at the base of the CDR cabinet.
2	Secure with eight hex screws as shown in Fig. 4-3.
3	Connect the inverter, as shown in Fig. 4-4, terminating the top black lead at the bottom right hand corner of the static inverter to the power terminal 3 on the power terminal block of the QUX6 unit (Fig. 4-13). The middle red wire goes to terminal 1, and the bottom black wire to the ground lug on the cabinet (Fig. 4-14).
4	Plug the gray ac line cord from the QUX6 power distribution unit into the bottom outlet on the duplex receptacle located on the back of the inverter (Fig. 4-4).
5	Take the black ac power cord and plug it into the bottom right-hand plug on the back of the MTU and into the top duplex receptacle on the inverter.
6	Plug the QUD1 ac line cord into the duplex receptacle on the QUX6 unit.

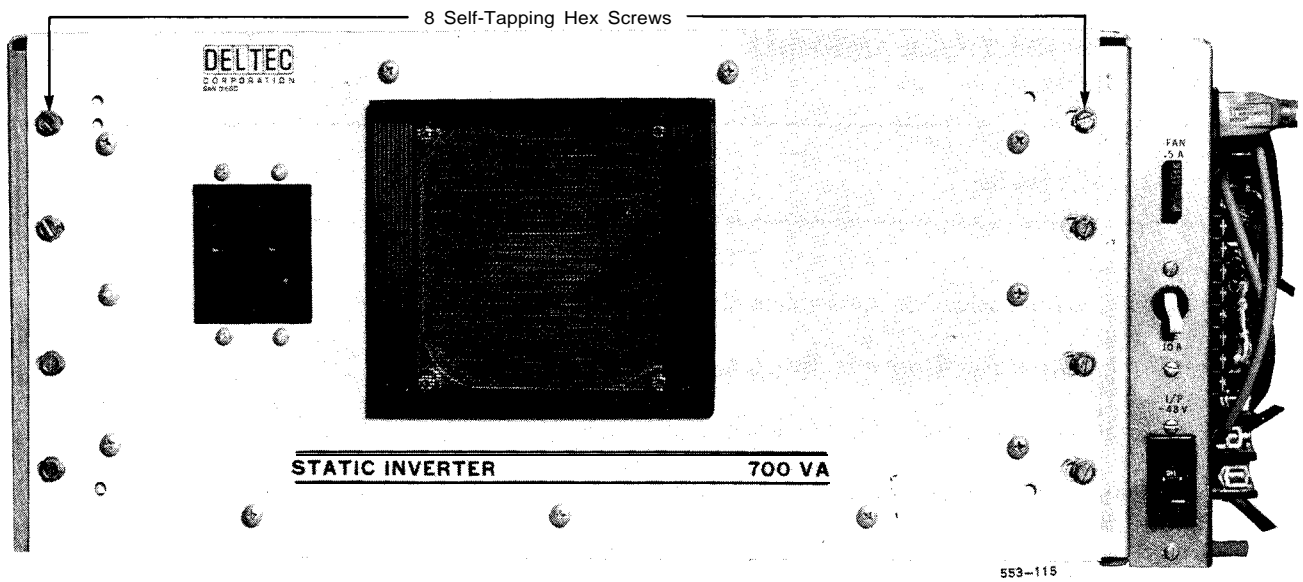


Fig. 4-3
Mounting the Deltec Static Inverter

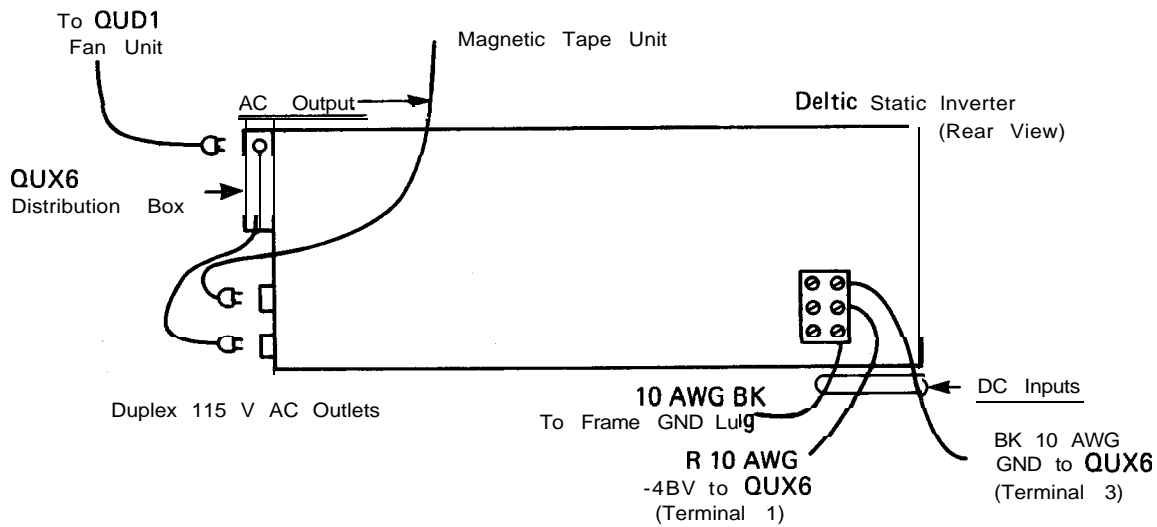
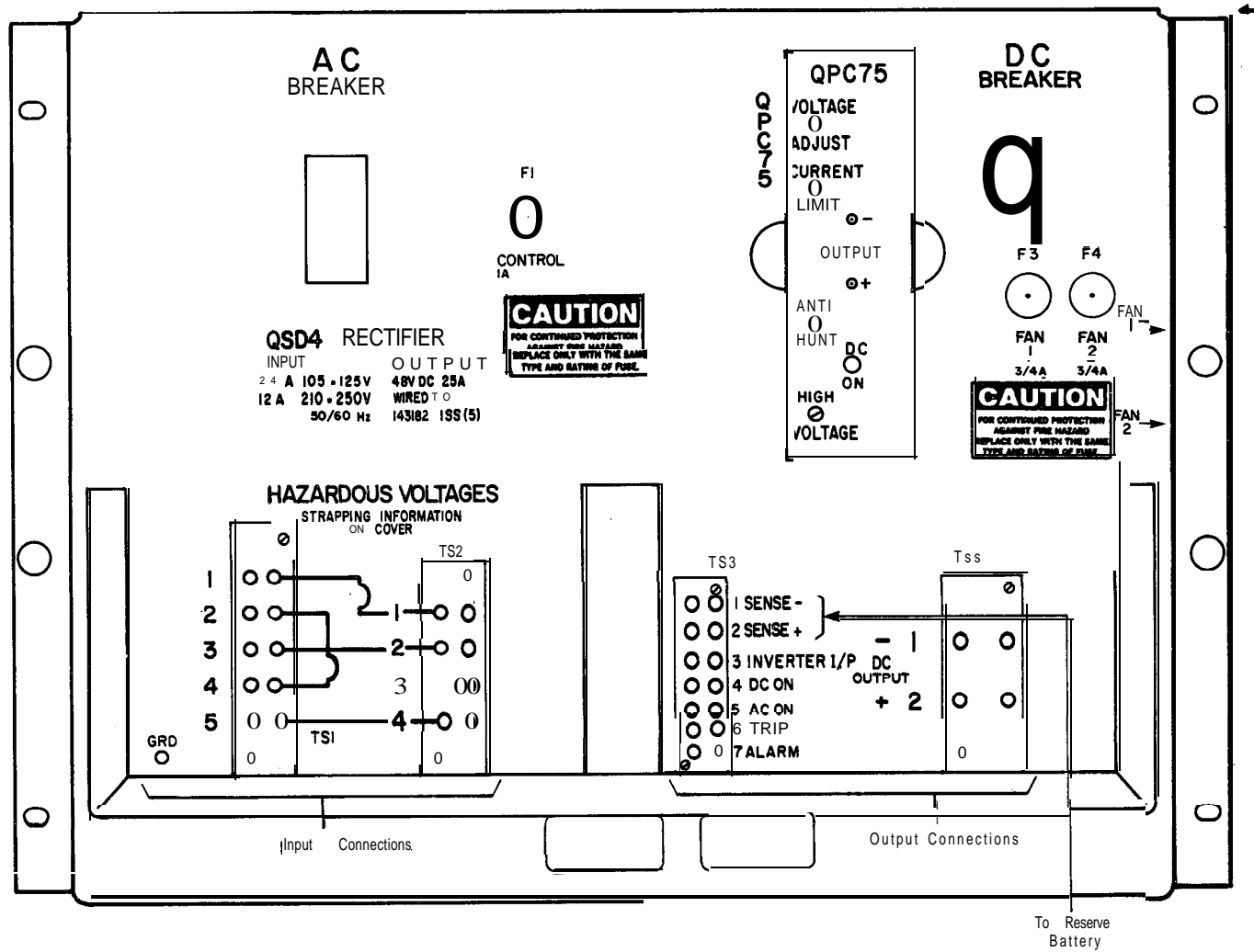


Fig. 4-4
Connections to the Deltec Static Inverter

Chart 4-4
INSTALLING THE RECTIFIER ASSEMBLY
(OPTION A1 OR B1)

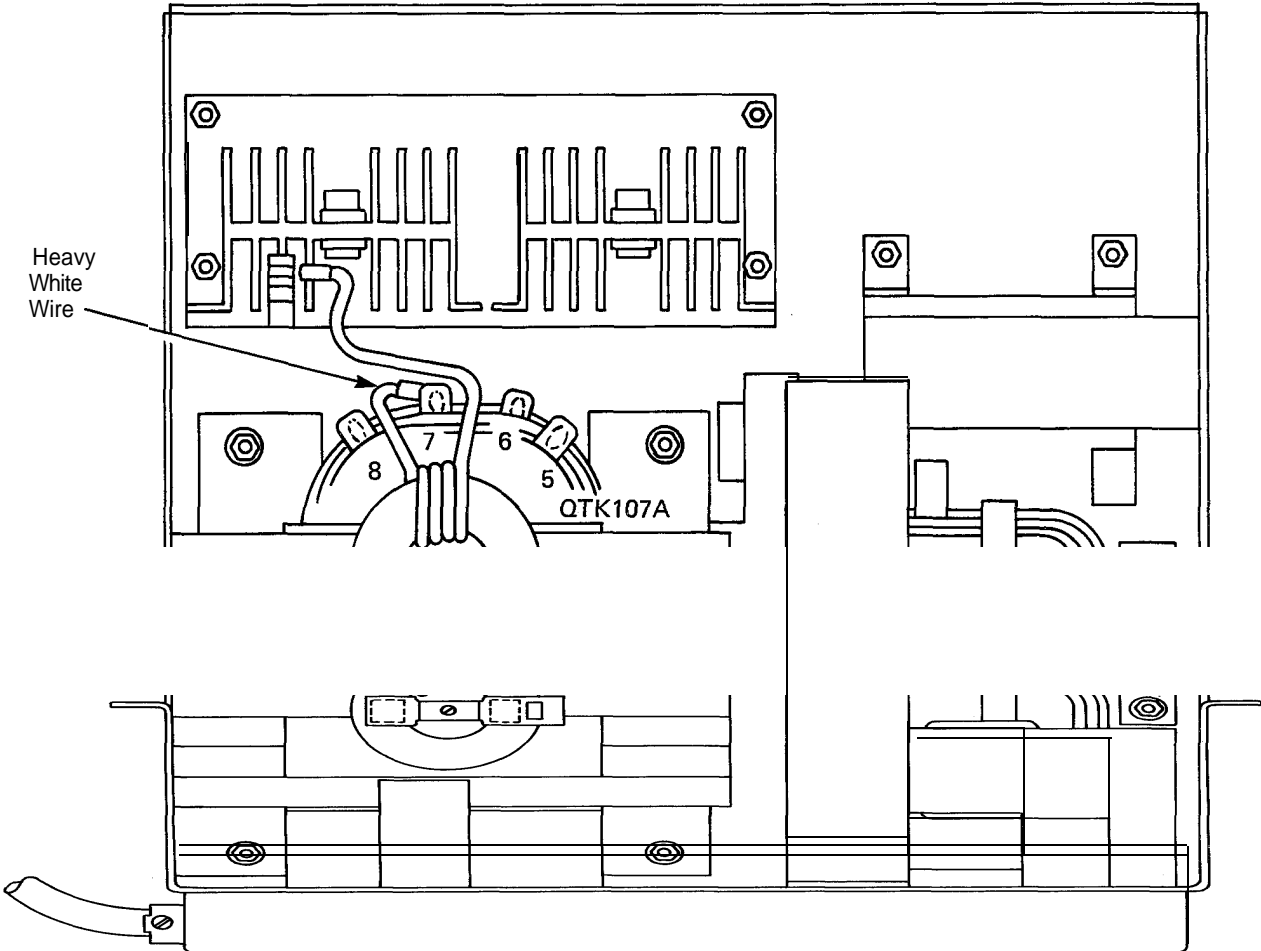
STEP	PROCEDURE
1	Lift rectifier and slide into cabinet above metal box provided for Deltec static inverter. <i>Caution: Rectifier weighs up to 200 lb (91 kg).</i>
2	Secure to frame with eight hex self-tapping screws provided.
3	Strap rectifier as shown in Fig. 4-5, 4-6 or 4-7 as required by the type of rectifier and ac input used.
4	The power and ground connections to the rectifier are performed in Chart 4-7.



Input Voltage	Input Connections			Straps	
	Phase	Phase	Neutral	TS1	TS2
220V Phase-to-Phase	TS1-1	TS1-2	-	4 to 5	2 to 3
220V Phase-to-Neutral	TS1-1	-	TS1-5	-	2 to 3
115V Phase-to-Neutral	TS1-1	-	TS1-5	1 to 2	3 to 4

TS3 Terminals	Lead	
	Color	Gauge
3	BL	18
4	BL	24
5	B L-W	24
6	BR	24
7	W	24

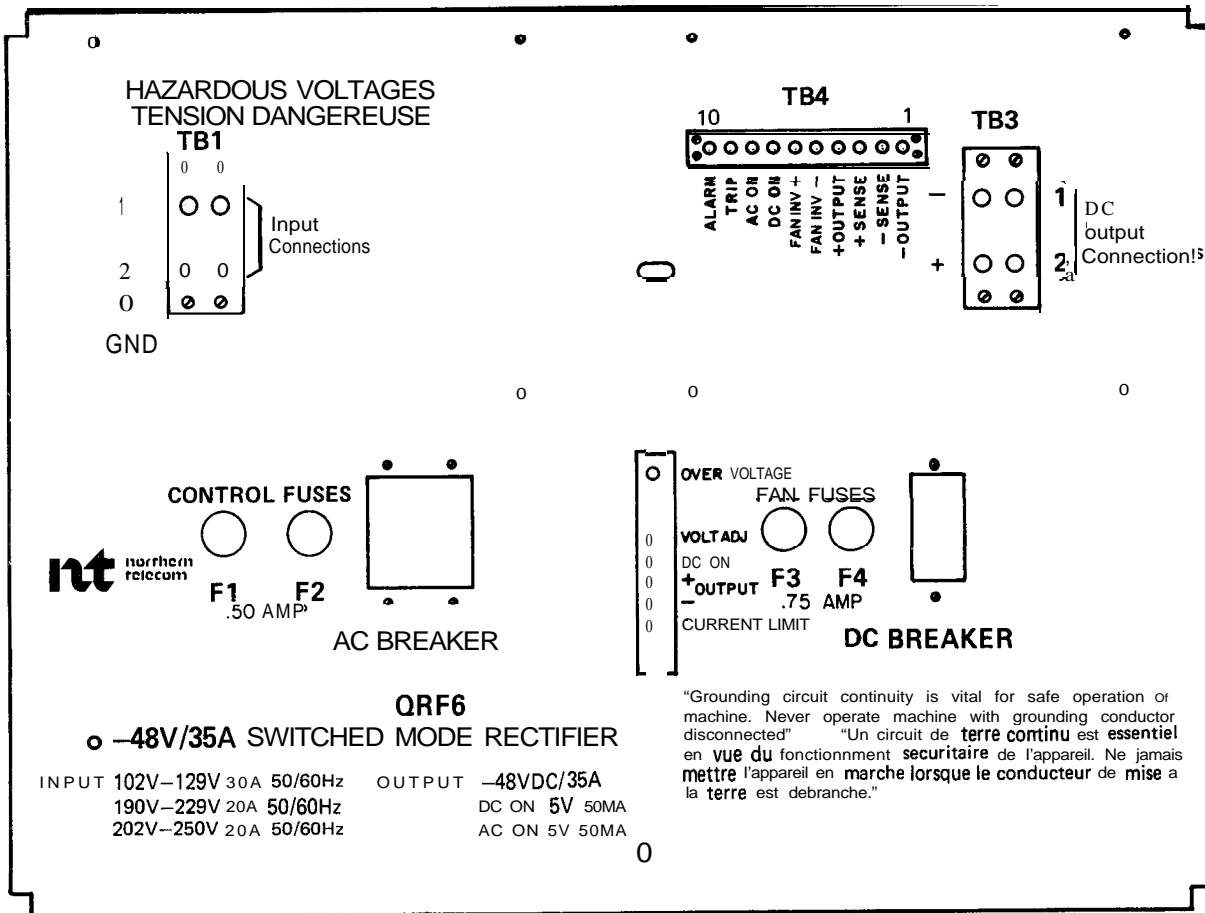
Fig. 4-5(a)
QSD4 Rectifier Wiring Options



NOTE: Shown Wired For 115/230 V ac

Nominal Input Voltage	Connect White Lead to Pin
105V/208V	8
115V/230V	7
125V/240V	6

Fig. 4-5(b)
QSD4 Rectifier - Output Tap Connections

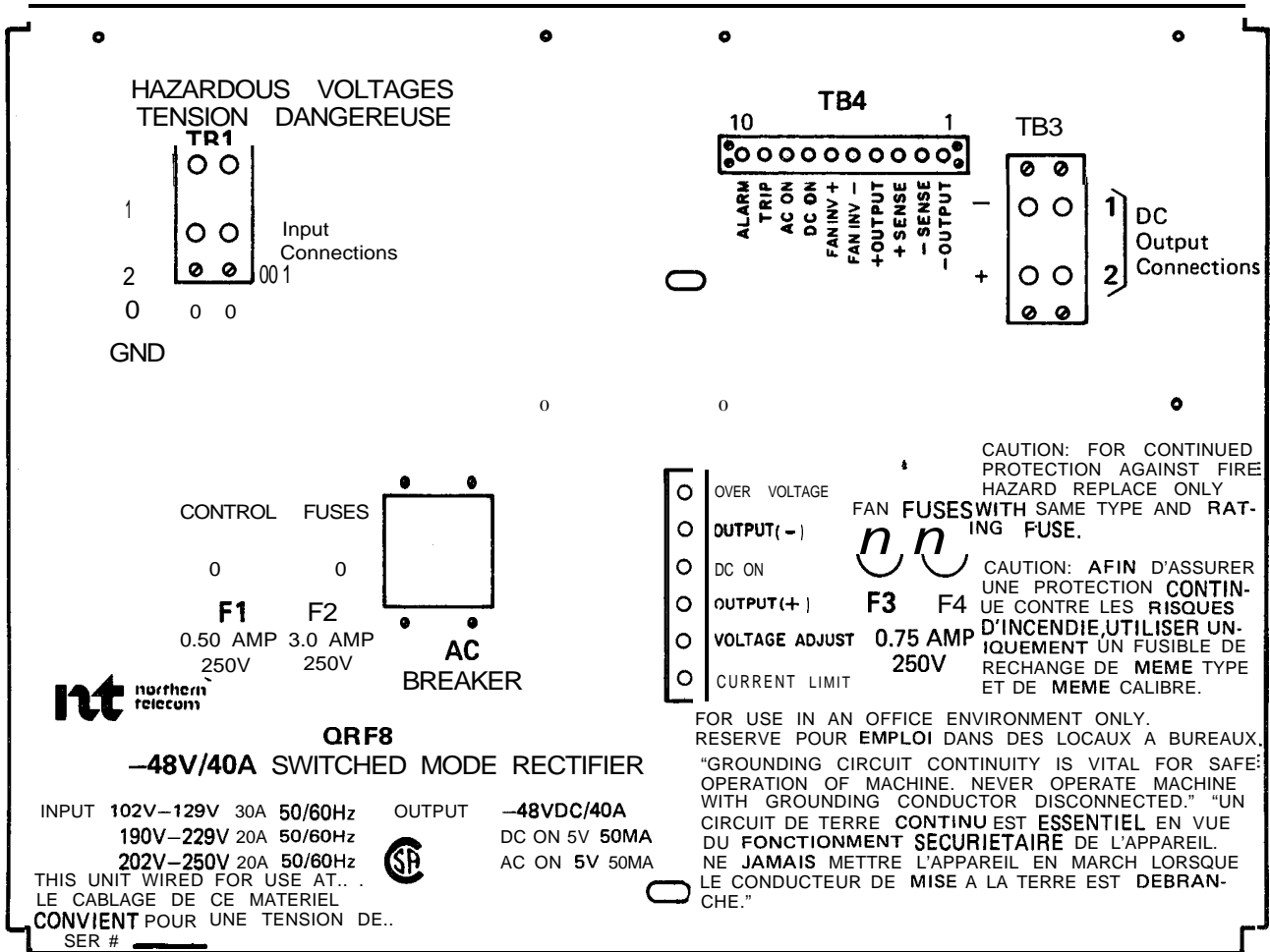


Input Voltage	Straps on TB2 (See Note)
102V-129V 60 Hz	F1 1 ● e n n n n 12
102V-129V 50 Hz	F1 1 ● e e e e e e e e e e e e 12
190V-229V 50/60 Hz	F1 1 ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● 12
202V-250V 50/60 Hz	F1 1 ● ● ● ● ● - i e e - e e 12

TB4 Terminals	Color	
	Color	Gauge
5	BL	18
7	BL	24
a	B L-W	24
9	BR	24
10	W	24

NOTE: TB2 is located inside the rectifier.

Fig. 4-6
QRF6 Rectifier Wiring Options



Input Voltage	Straps on TB2 (See Note)
102V-129V 60 Hz	
102V-129V 50 Hz	
190V-229V 50/60 Hz	
202V-250V 50/60 Hz	

TB4 Terminals	Color Gauge	
	Color	Gauge
5	BL	18
7	BL	24
8	B L-W	24
9	BR	24
10	W	24

NOTE: TB2 is located inside the rectifier.

Fig. 4-7
QRF8 Rectifier Wiring Options

Chart 4-5
 INSTALLING THE QUT3 FAN AND INVERTER POWER UNIT
 (OPTION B2)

STEP	PROCEDURE
1	Attach the QUT3 unit (Fig. 4-8) to the CDR cabinet frame below the CE shelf with four screws.
2	Connect terminal 1 on the QUX6 unit (Fig. 4-13) to terminal 3 on the TS3 connector block (the front of the QUT3).
3	Connect ground lead from terminal 3 on the QUX6 unit to terminal 8 on TS3 (QUT3).
4	If a power alarm is to be used, connect the alarm lead to terminal 5 (AC-ON) on TS3 which provides a 3 A, 30V dc ground contact closure to indicate ac power failure.
5	Plug the ac line cord for the QUD1 cooling unit into one of the duplex receptacles on the QUT3 unit.

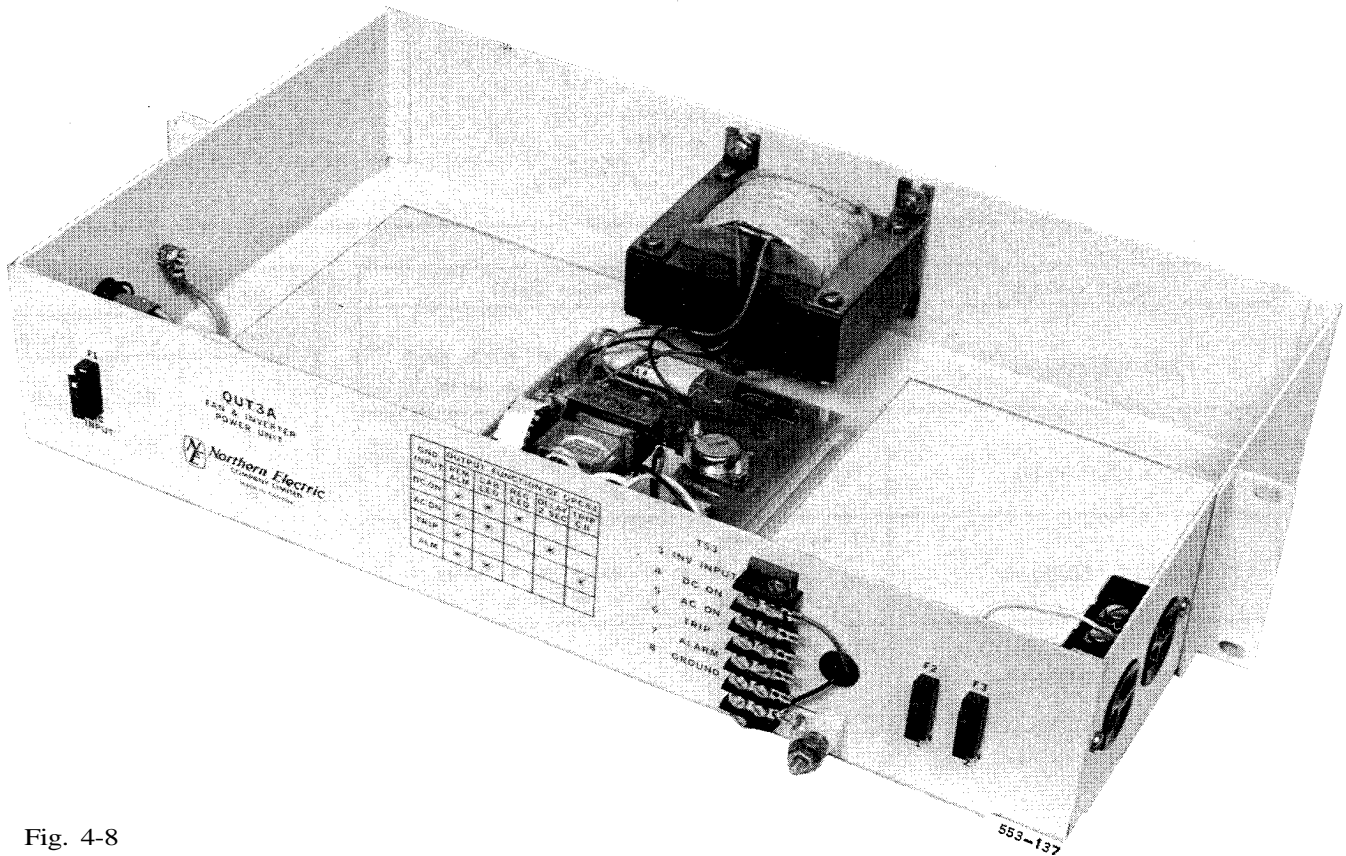


Fig. 4-8
 The QUT3 Fan Inverter Unit

Chart 4-6
CONNECTION OF THE SERIAL DATA INTERFACE

STEP	PROCEDURE
------	-----------

1 Set up QPC45 and QPC139 SDI packs which connect the SL-1 system(s) to the CDR as follows (see also Fig. 4-9, 4-10):

SL-1 SDI

CDR SDI(S)

(a) Shorting plug in modem position.

When the cable length to the SL-1 is less than 50 feet, set the shorting plug in the EIA position. For longer distances using a modem, set the plug in the MODEM position,

(b) Speed switch settings must be set for the same speed in both SDI packs.

(c) SDI address switch settings for port number must set to correspond to the link number in the SL-1 software (see 553-2yy1-220).

Select appropriate device numbers (see "Device Number Selection" immediately following).

2 **Device Number Selection:** The CDR machine requires that each SDI port be assigned a "device number". The device number is selected by address switches inside the SDI circuit pack. The QPC45 (single port) SDI requires one device number and the QPC139 (2 port) SDI requires two consecutive device numbers.

The following rules apply in the selection of device numbers:

(1) If the CDR machine has an 8k RAM (QPC31), there is only one port and its device number must be 0.

(2) On a 32k machine (QPA62), which can have up to a maximum of 12 SL-1 ports, the choice of device numbers is subject to the following constraints:

(a) No two device numbers may be the same. Note that the port number of the SL-1 SDI and the device number of the CDR SDI need not be the same.

(b) For a QPC45 with device number n, where n is even, there must be no QPC139 with the device number pair (n+8, n+9). If n is odd, there must be no QPC139 set to respond as device numbers (n+7, n+8).

(3) In choosing SDI device numbers in the CDR machine, it is recommended that:

(a) Low device numbers be preferred over high.

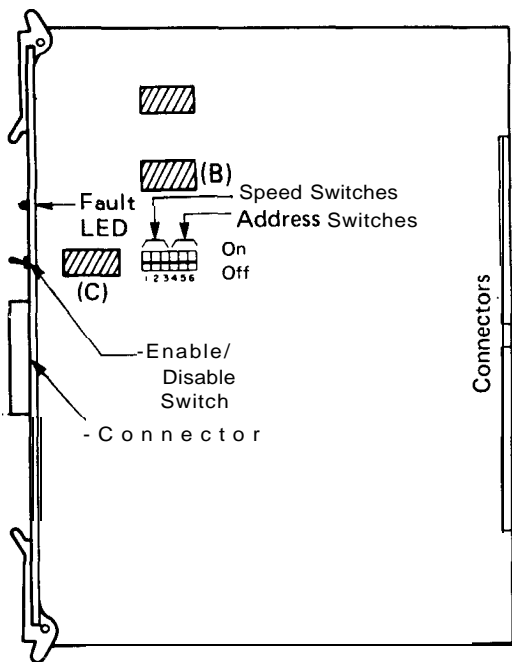
(b) High baud rates be associated with the lower device numbers.

Note: The software of all SL-1 machines connected to the CDR must be modified to establish the SDI ports and customer options. Refer to 553-2yy1-220 for this information.

Chart Continued

Chart 4-6 Continued
CONNECTION OF THE SERIAL DATA INTERFACE

STEP	PROCEDURE
3	<p>Plug the double-ended 25-wire NE-25MQ SDI cable into the 25-pin connector(s) on the faceplate of the SDI circuit packs: one in the CDR system, and the other into the corresponding SDI pack of the host SL-1 system. Ensure that the baud rate is the same in the CDR port as in the corresponding SL-1 port.</p> <p><i>Note:</i> For connection to a modem, use the same 25-wire cable from SL-1 SDI to modem, and from modem to CDR SDI.</p>
4	<p>If a QPA62C (32k RAM) is used, set the switches as shown in Fig. 4-11. If a QPA62D, or later vintage is used, set the switches as shown in Fig. 4-12.</p>



Each Input/Output device has an address (Device Number) which is set on the selection switch on the QPC45 pack. See Table Below.

The Operational speed of each device is also set on the selection switch. An Option plug is inserted in one of the plug positions (A), (B), (C), depending upon type of device connected to the QPC45 pack as follows:

- A) When QPC45 pack connects to a Modem (TTY in remote location)
- B) For a standard EIA data terminal, e.g., Datacom 300.
- C) For a TTY that connects directly to QPC45 pack (N Modem).

Address Selection	
Device Number	Switches 4 5 6
0	● ●
1	0 ●
2	0 ●
3	0 0
4	● ● 0
5	0 ● 0
6	0 0
7	0 0 0

Device Speed (Baud)	Switches		
	1	2	3
110	●	○	○
300	○	○	○
1200	○	○	●
2400	○	●	○
4800	○	●	●

○ Switch Off
 | Switch On

Fig. 4-9
 QPC45 SDI Pack Internal Plug Positions and Address Switch Settings

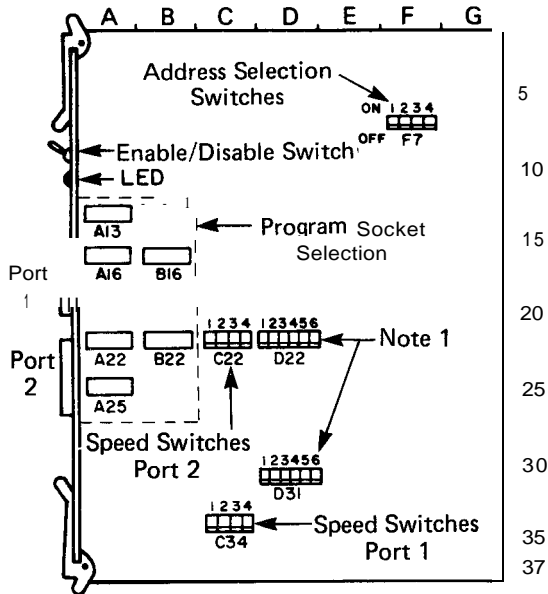


Table A

Address Selection				
Device Number	Switches			
	2	3	4	
0-1	●	●	●	
2-3	●	●	○	
4-5	●	○	●	
6-7	●	○	○	
8-9	○	●	●	
10-11	○	●	○	
12-13	○	○	●	
14-15	○	○	○	

Table B

Speed Switches				
Device Speed (Baud)	Switches			
	2	3	4	
110	●	●	●	
150	○	●	●	
300	●	○	●	
600	○	○	●	
1200	●	●	○	
2400	○	●	○	
4800	●	○	○	
9600	○	○	○	

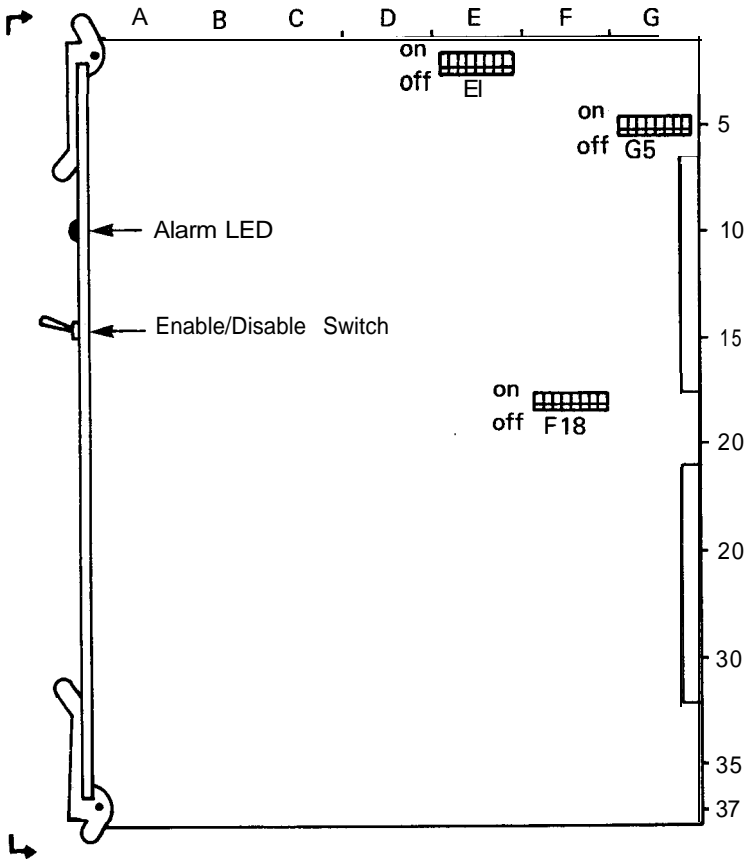
● = ON Note: Switch 1 Not Connected
○ = OFF

Note: 1.
These switches are not used.
Set to OFF position.

Table C

Program Socket Selection		
Socket Location	Port	Connecting Device
A13	1	When QPC139 Connects to a Modem (Remote TTY)
A25	2	
A16	1	When QPC139 Connects to Standard EIA RS 232 Data Term.
A22	2	
B16	1	When QPC139 Connects directly to a TTY
B22	2	

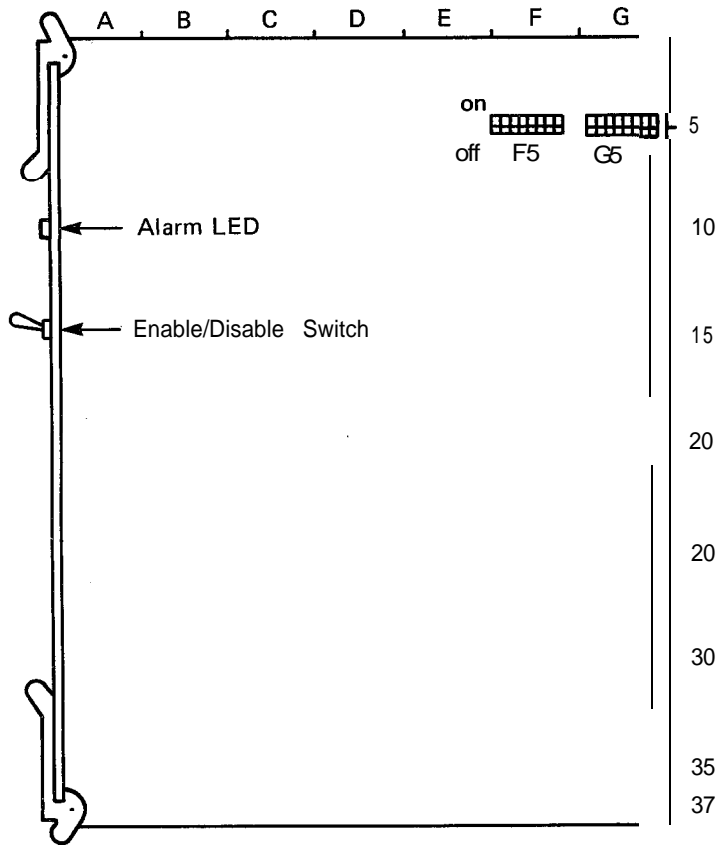
Fig. 4-10
QPC139 SDI Pack Internal Plug Positions and Address Switch Settings



The switches on the QPA62C and earlier vintage and QPA62E/QPA62G (32K memory) must be set to respond as the first 32K words of data store 0. There are three sets of switches inside the pack that must be set as follows:

○ Switch OFF ● Switch ON	
Switch	Setting
	1 2 3 4 5 6 7 8
EI	○ ○ ● ○ ○ ○ ○ ○
G5	○ ○ ● ○ ○ ○ ○ ○
F18	○ ○ ● ○ ● ○ ● ○

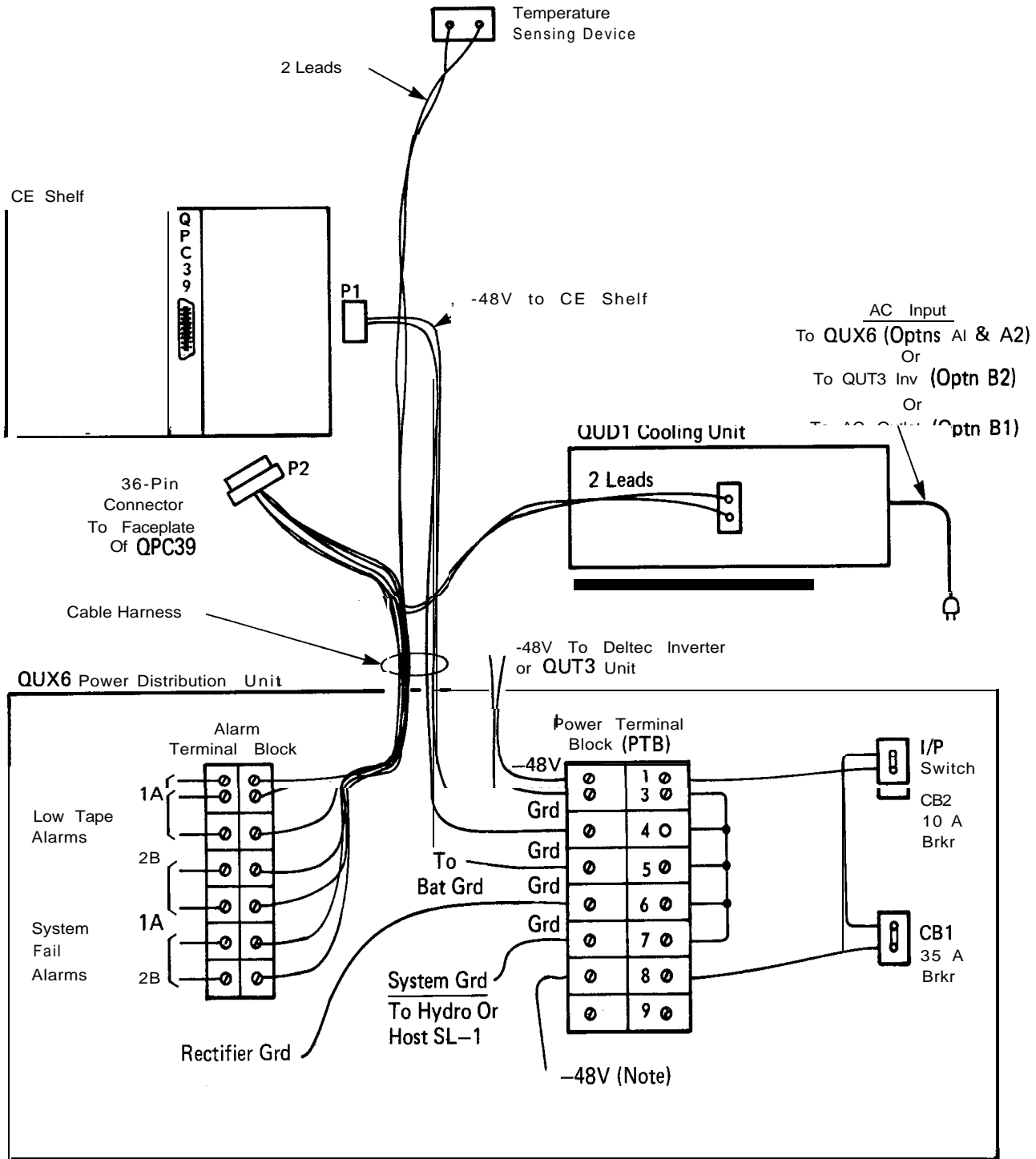
Fig. 4-11
QPA62C/QPA62E/QPA62G 32k RAM Switch Settings



The switches on the **QPA62F/QPA62H** (32K memory) must be set to respond as the first 32K words of data store 0. There are two sets of switches inside the pack that must be set as follows:

0 Switch OFF		● Switch ON						
Switch	Setting							
	1	2	3	4	5	6	7	8
G5	○	○	○	●	○	○	○	○
F5	0	0	0	0	0	0	0	0

Fig. 4-12
QPA62F/QPA62H or **QPA62D** 32k RAM Switch Settings



Note: The -48V goes to the battery, if available. Otherwise connect to rectifier. (Refer to Chart 4.7).

Fig. 4-13
CDR Cabinet Internal Wiring

Chart 4-7
POWER AND ALARM CONNECTIONS

STEP	PROCEDURE																								
1	Ensure that the ENB/DIS switch at the top of each circuit pack is switched to ENB.																								
2	Ensure that all circuit breaker switches are in the OFF position. <ol style="list-style-type: none"> ac and dc breakers on the rectifier, CB2 10 A breaker on the QUX6 unit, I/P 48 V switch on QUX6, circuit breaker on Deltec static inverter (if present). 																								
3	Check that all fuses have the correct rating as indicated below: <p>QSD4 — F1 (1.3 A) and F3 (0.75 A) QRF6 — F1, F2 (0.5 A) and F3 (0.75 A) QRF8 — F1 (0.5 A), F2 (3.0 A) and F3 (0.75 A) QUT3 — F1 (1.3 A) and F2, F3 (0.5 A)</p> <p>The color codes for fuses are as follows:</p> <table border="1"> <thead> <tr> <th>PART NUMBER</th> <th>RATING (A)</th> <th>COLOR</th> </tr> </thead> <tbody> <tr> <td>QFF1A</td> <td>1.3</td> <td>white</td> </tr> <tr> <td>QFF1B</td> <td>2.0</td> <td>orange</td> </tr> <tr> <td>QFF1C</td> <td>3.0</td> <td>blue</td> </tr> <tr> <td>QFF1F</td> <td>0.25</td> <td>violet</td> </tr> <tr> <td>QFF1G</td> <td>0.5</td> <td>red</td> </tr> <tr> <td>QFF1H</td> <td>0.75</td> <td>brown</td> </tr> <tr> <td>QFF3A</td> <td>dummy</td> <td>black</td> </tr> </tbody> </table>	PART NUMBER	RATING (A)	COLOR	QFF1A	1.3	white	QFF1B	2.0	orange	QFF1C	3.0	blue	QFF1F	0.25	violet	QFF1G	0.5	red	QFF1H	0.75	brown	QFF3A	dummy	black
PART NUMBER	RATING (A)	COLOR																							
QFF1A	1.3	white																							
QFF1B	2.0	orange																							
QFF1C	3.0	blue																							
QFF1F	0.25	violet																							
QFF1G	0.5	red																							
QFF1H	0.75	brown																							
QFF3A	dummy	black																							
4	Connect the low tape and system failure alarms leads to the alarm terminal block on the QUX6 unit (Fig. 4-13). <p><i>Note:</i> CDR systems can accommodate four alarm devices (two for each type). The alarm device, if required, should be supplied by the operating company. Alarms are rated at 30 VA; 1 A max.</p>																								
5	Connect CDR grounds by performing step (a) or (b) as required. Refer to Fig. 4-13 and 4-14. <ol style="list-style-type: none"> If CDR is within 50 cable ft (15.2 m) of host SL-1 and if -48 V dc comes from host SL-1 then make the following connections: <ol style="list-style-type: none"> CDR frame ground lug must be connected to SL-1 frame ground via an insulated 6 AWG wire. 																								

Chart Continued

Chart 4-7 Continued
POWER AND ALARM CONNECTIONS

STEP PROCEDURE

(2) CDR system ground must be connected to SL-1 system ground via an insulated 6 AWG wire.

(b) If CDR is more than 50 ft (15.2 m) from host SL-1; i.e., modems are present, then:

(1) CDR system ground must be connected to street side of water meter (or equivalent) via an insulated 6 AWG wire.

(2) CDR frame ground must be connected to CDR system ground via an insulated 6 AWG wire

Note: Electronic PBX are extremely sensitive to transient voltage or noise in the ground system. Resistance of the ground wire must be as close to zero as possible, never exceeding 0.5 Ω .

6 Connect power to the CDR system according to the CDR options listed below.

To connect the rectifiers (option A1 and B1) one of the front covers must be removed and the connecting wires threaded through the snap bushing on the right side of the rectifier. For a QSD4 rectifier the lower right cover is removed; for QRF6/8 the top cover is removed. The connecting terminals are as follows:

QSD4 — TS5-1 (neg. terminal), TS5-2 (pos. terminal)
QRF6/8 — TB3-1 (neg. terminal), TB3-2 (pos. terminal)

Option A1 — CDR with rectifier and reserve battery

(1) Connect grounds from the QUX6 unit: terminal 5 to the positive terminal on the battery distribution box, terminal 6 to the positive terminal on the rectifier. Refer to 553-2yy1-235 and Fig 4-13.

(2) Using 4 AWG wire, connect the negative terminal of the rectifier to the charge terminal on the battery distribution box.

(3) Using 4 AWG wire, connect the discharge terminal on the battery distribution box to terminal 8 on the QUX6.

(4) Connect the sense and monitor leads as discussed in 553-2yy1-235.

(5) Replace rectifier cover.

(6) Plug the rectifier ac cord into the Hubble-type outlet.

Chart Continued

Chart 4-7 Continued
POWER AND ALARM CONNECTIONS

STEP	PROCEDURE
------	-----------

Option B1 — CDR with rectifier

- (1) Connect the positive terminal of the rectifier to terminal 5 on the QUX6 unit (Fig 4-13).
- (2) Connect the negative terminal of the rectifier to terminal 8 on the QUX6.
- (3) Replace the rectifier cover.
- (4) Plug the rectifier ac cord into the Hubble-type outlet.

Option A2 and B2 CDR powered from host SL-1

- (1) Using 4 AWG wire, connect terminal 5 on the QUX6 unit (Fig. 4-13) to the positive bus on the QBL6/15 unit in the SL-1 system. Refer to 553-2yy1-235.
 - (2) Using 4 AWG wire, connect terminal 8 on the QUX6 unit to the discharge terminal on the QBL6/15 unit in the SL-1 system.
 - (3) For option B2 plug the MTU into the standard U ground ac outlet.
- 7 Replace the back cover on the CDR cabinet and slide the cabinet into place.
- 8 Adjust the footscrews so the cabinet is level.

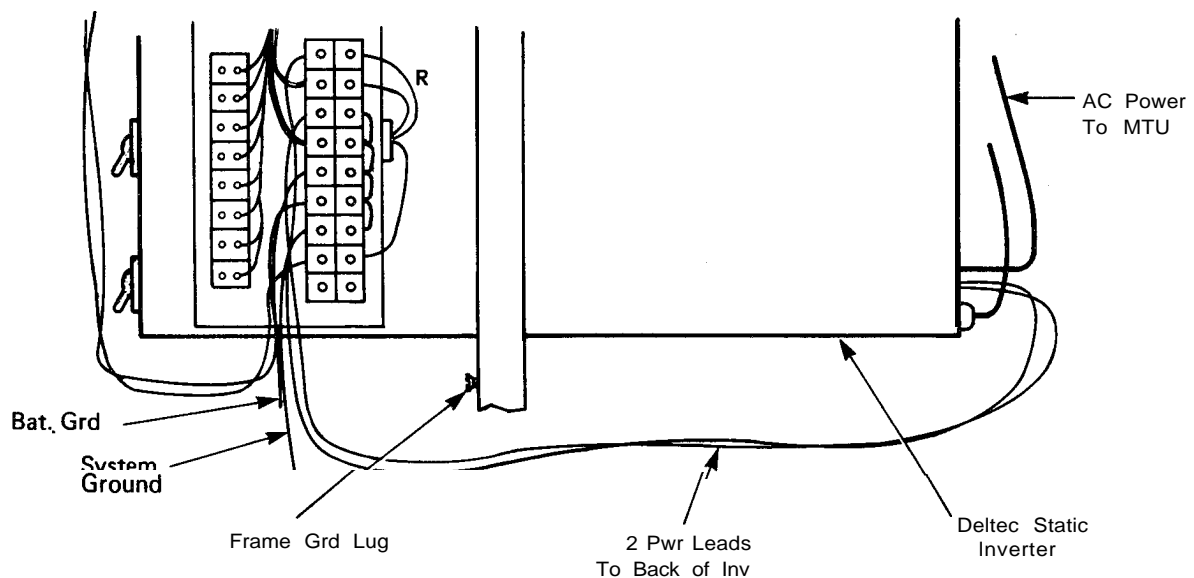


Fig. 4-14
Ground Screw Location at Base of Cabinet

PRACTICE 553-2631-210

Chart 4-8
SYSTEM START-UP

STEP	PROCEDURE
1	Turn on rectifier power breakers in the following order: QSD4 rectifier — ac breaker followed by the dc breaker QRF6 rectifier — dc breaker followed by the ac breaker QRF8 rectifier — ac breaker (no dc breaker provided)
2	Turn on the CB2 and I/P breakers on the TCB (QUX6 unit).
3	Turn on the breaker on the Deltec static inverter.
4	At left rear of MTU check that the input power voltage is set to 115 or 230 V ac as required. Open the dustcover door on the MTU (see Fig. 4-2). Locate the power switch in lower left-hand corner and switch it to 'ON'. The SYSTEM FAIL alarm may operate at this point, but goes go off once the MTU is recording properly.
6	Mount tapes. (Refer to 553-2631-310 CDR System Operation, for method.)

Chart 4-9
ACTIVATING MULTI-PORT CDR

STEP	ACTION	VERIFICATION
1	After power-up and when all the SDI packs have been set up for CDR recording, the connected ports should be automatically enabled for CDR recording from the respective SL-1 machines.	Load overlay 42 from one of the connected SL-1 machines and issue the 'port n' and 'stat sll all' commands (refer to 553-2001-505). This operation prints a summary of the CDR port configuration. Then issue the 'echo n m' commands to test the CDR links for transmission.
2	If a CDR port is being added to an existing CDR multiport configuration, then load overlay 42 from an existing SL-1 and issue an 'enl sll m' command where m is the device number switch setting on the CDR SDI pack to the added.	Same as above.
3	If a QPC139 dual SDI has one unused port that is to be added into the CDR, then use overlay 42 to check status on the ports. Then disable the used unit on the QPC139 by issuing the 'dis sll m' command. Set the switch settings for the new unit and then enable both SL-1 ports.	Same as above.
	<i>Note:</i> CDR records from the SL-1 originally connected to QPC139 are lost during this procedure.	

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

CDR ASSEMBLY AND INSTALLATION
(COMPLIANCE WITH FCC REGULATIONS)

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2-3 QPC45 AND QPC 139 SDI PACK WIRING	2-5
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* SL-I is a trademark of Northern Telecom Limited

1. GENERAL

1.01 This appendix describes the methods used to install CDR systems that conform with FCC regulations concerning electromagnetic interference @MI).

1.02 Installation procedures for equipment not covered in this appendix are given in 553-2631-210.

1.03 Installation requirements are given in 553-2yy1-200 and in Appendix 1 to 553-2yy1-200.

2. INSTALLATION PROCEDURES

Cabinet Door

2.01 Opening Door. The cabinet door is opened by inserting a key in the locking mechanism located on the right side of the door and turning the key counter-clockwise. The door is then swung open by pulling the right side of the door outward.

Note: A key is shipped with the cabinet and is located in a bag secured to the top outside right hand corner of the cabinet. The key is the equivalent to a $5/32$ in Allen wrench.

2.02 Removing Door Panels. Once the door is unlocked, swing the door panel open, and lift upward to unhook from its hinges.

2.03 Installing Door Panel. Line hinge on door panel with hook-on hinge on outer left side edge of front of cabinet and lower panel into position.

2.04 Closing Door. Insert key in locking mechanism and gently push door against cabinet and turn key clockwise until the locking mechanism engages.

Bulkhead Assemblies

2.05 Bulkhead assemblies permit wiring to be run between cabinets installed side-by-side or back-to-back. These assemblies are installed once the cabinets are unpacked and are being placed in their final location. See Chart 2-1 for bulkhead assembly installation procedures and Chart 2-2 for removal procedures.

Chart 2-1
BULKHEAD ASSEMBLY INSTALLATION PROCEDURES

It may be necessary to temporarily remove some equipment from the cabinet to facilitate the installation of bulkhead assemblies.
Bulkhead assemblies should be installed in cabinets placed back-to-back first.

STEP PROCEDURE

- 1 Unlock and remove doors from cabinets.
 - 2 Remove door panels by opening doors and lifting upward until unhooked from hinges.
 - 3 Remove screws securing covers on cable access holes requiring bulkhead assemblies.
 - 4 Place cabinets back-to-back or side-by-side.

Note: Install cabinets back-to-back first.
 - 5 Align cable access holes between cabinets.

Note: It may be necessary to adjust the feet on the cabinets to align the access holes. Unscrewing the feet raises the cabinet and screwing the feet lowers the cabinet.
 - 6 Dismantle bulkhead assemblies.
 - 7 Place one washer plate on each pipe coupler. (See Fig. 2-1.)
 - 8 Insert pipe coupler in cable access hole in first cabinet ensuring that it protrudes through the hole into second cabinet.
 - 9 Place a washer plate on end protruding into second cabinet. (See Fig. 2-1.)
 - 10 Place lock nut (Fig. 2-1) on pipe coupler in second cabinet. Do not tighten nut.
 - 11 Repeat Steps 8 through 10 for remaining bulkhead assemblies.
 - 12 Tighten all lock nuts securely.
 - 13 Place protective bushing (Fig. 2-1) on each pipe coupler.
-

Chart 2-2
BULKHEAD ASSEMBLY REMOVAL PROCEDURES

Bulkhead assemblies should not be removed unless the cabinet is being moved or removed. All wiring between affected cabinets should be removed from bulkhead assemblies before proceeding. It may be necessary to temporarily remove some equipment from the cabinet to facilitate bulkhead assembly removal.

STEP PROCEDURE

- 1 Unlock cabinet doors by inserting key in door locking mechanism and turning key counter-clockwise.

Note: The key is the equivalent, to a 5/32 in Allen wrench.

- 2 Remove door panels by opening doors and lifting upward until unhooked from hinges.
- 3 Remove protective bushing from bulkhead assembly being removed. (See Fig. 2-1.)
- 4 Loosen and remove lock nut from pipe coupler. (See Fig. 2-1.)
- 5 Remove first washer plate from pipe coupler. (See Fig. 2-1.)
- 6 Grasp pipe coupler from adjacent cabinet and pull coupler out of cable access hole.
- 7 Separate cabinets and install metal cover plates over cable access hole.

Note: All bulkhead assemblies between two cabinets must be removed before cabinets can be separated. System should not be operated without bulkhead assemblies unless the cable access holes are covered with cover plates.

Serial Data interface
Pack

2.06 Serial Data Interface packs are installed as described in 553-2631-210 with the addition of a filter connector in the I/O panel and a flat interface cable. (See Chart 2-3 for installation procedures.)

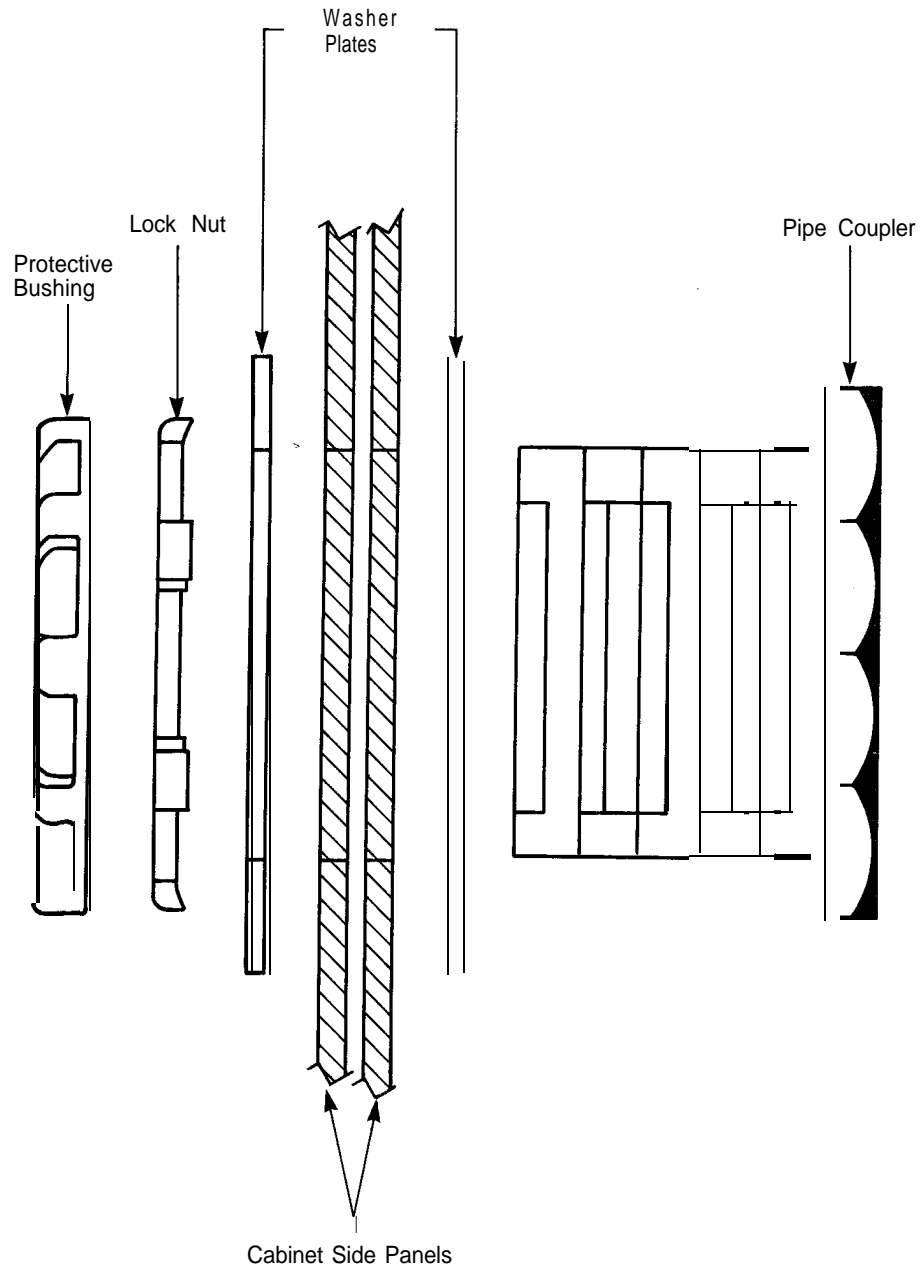


Fig. 2-1
Bulkhead Assembly

Chart 2-3
QPC45 AND QPC139 SDI PACK WIRING

Refer to Fig. 2-2.

STEP PROCEDURE

- 1 Install circuit pack in assigned slot as described in 553-2631-210.
 - 2 Remove the two screws securing the cover plate over a filter connector opening on top of cabinet and remove cover.
 - Caution: Do not drop screws or cover plates into cabinet as equipment may be damaged.
 - 3 Insert the filter connector into opening from inside cabinet.
 - 4 Hold filter connector and secure with two screws inserted from the top of the cabinet.
 - 5 Connect QCAD42 interface cable to assigned filter connector on underside of cabinet top panel. Secure connector with bale locks.
 - 6 Connect other end of interface cable to the SDI pack and secure with connector screws.
 - Note: Interface cables should be run flat (not twisted) in the cabinet to avoid congestion.
 - 7 Secure interface cable to cabinet frame with plastic cable ties.
 - 8 Complete installation as described in 553-2631-210.
-

Ground Wire, Power Wire and Conduit Size	2.07 Information on size of wiring and conduit required is given in 553-2yy1-200 and in 553-2yy1-235.
Power	2.08 All -48 V wiring must be run in conduit bonded to the cabinet and to the equipment at the far end. The wiring enters the cabinet through the conduit designated PWR on the cabinet top panel. Power connections are as described in 553-2631-210.
Grounding	<p>2.09 System Equipped with Rectifier. Chart 2-4 gives grounding procedures for CDR cabinets equipped with rectifiers.</p> <p>2.10 System Not Equipped with a Rectifier. Chart 2-5 gives grounding procedures for CDR cabinets not equipped with rectifiers.</p>

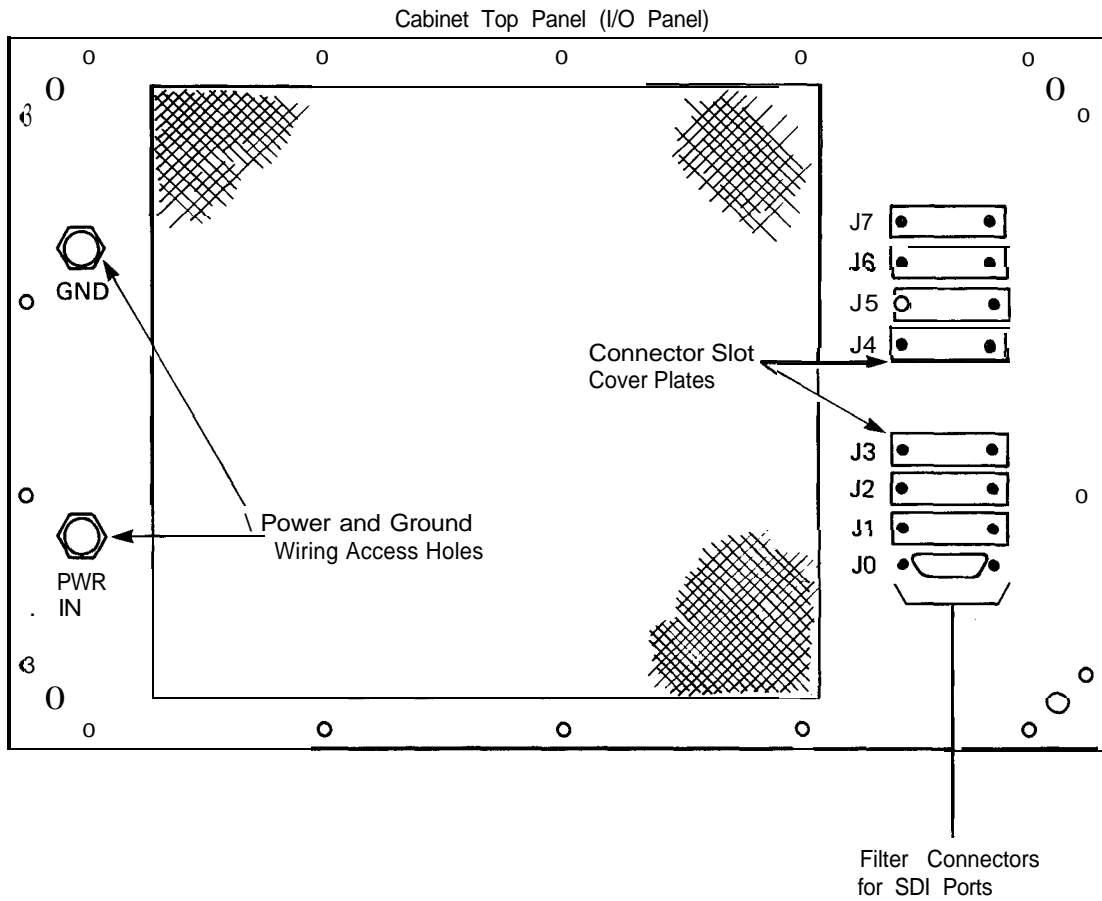


Fig. 2-2
Cabinet Top Panel (I/O Panel)

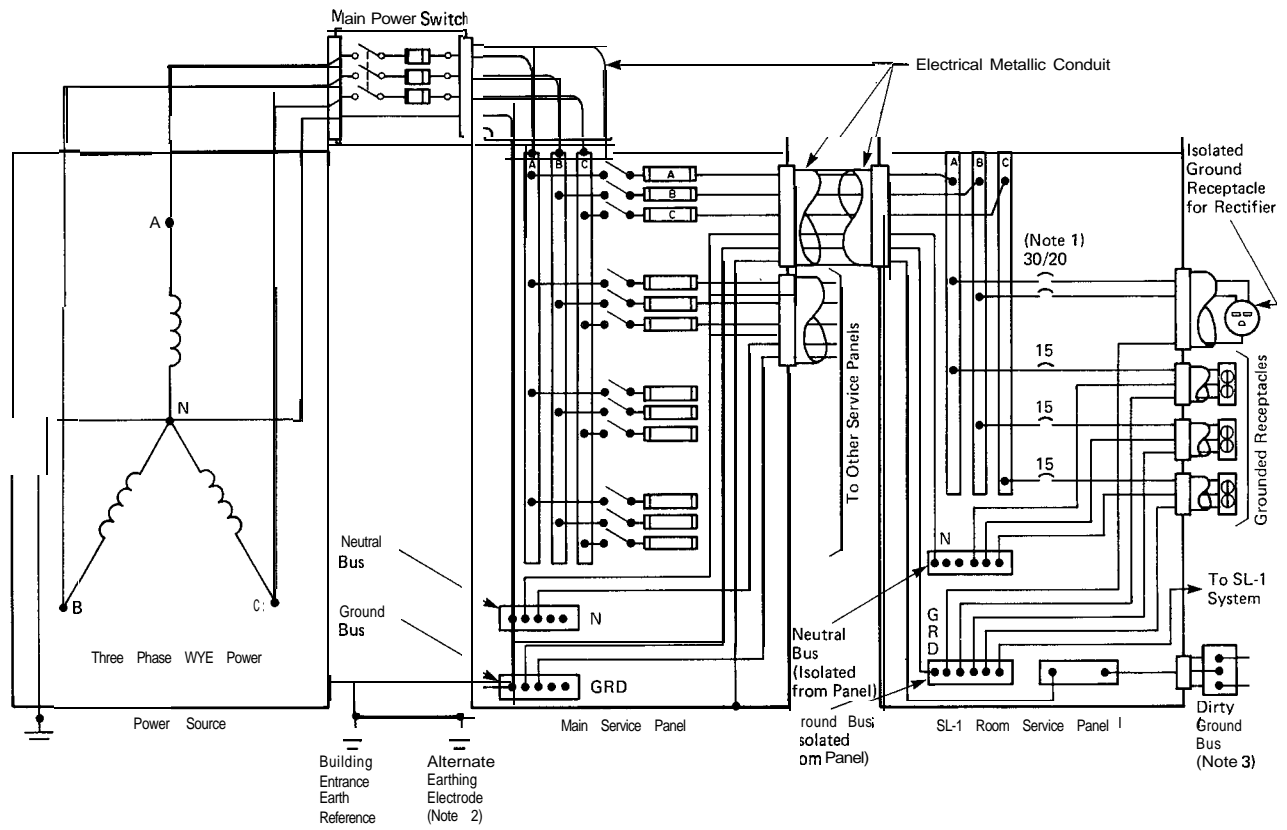
Chart 2-4
GROUNDING CDR CABINET EQUIPPED WITH A RECTIFIER

STEP PROCEDURE

- 1 Install a ground wire in the cabinet conduit designated GND on the cabinet top panel.
 - 2 Connect the wire to the lug on the cabinet upright near the exit of the conduit at the bottom of the cabinet.

Note: No more than 2 in (50 mm) of wire should be exposed between the conduit exit and the terminal lug.
 - 3 Connect the other end of the wire to:
 - 1 the isolated ground bus in the switchroom service panel on systems not equipped with reserve batteries (see Fig. 2-3)
 - the positive bus of the battery distribution box on systems equipped with reserve batteries

Note: This wire is not required to run in conduit. If conduit is used, it must be bonded to the cabinet and to the battery distribution box.
 - 4 Ensure that all ground connections are secure. The personal hazard ground is provided through the green wire safety ground from the switchroom service panel to the rectifier ac power connection. Ensure that this wire is properly connected.
-



- Notes:
1. Fuse at 20 or 30A depending on system requirements.
 2. If required, install an alternate earthing electrode at least 6 ft. (1.8m) from the building entrance earth reference. Connection to building entrance earth reference must be cadwelded or secured with two ALCu rated clamps.
 3. Connections to dirty ground bus include grounds to Main Distributing Frames, cable sheaths and protector blocks.

Fig. 2-3
Typical Service Panel Arrangement

Chart 2-5
GROUNDING CDR **CABINET** NOT EQUIPPED WITH RECTIFIER

STEP PROCEDURE

- 1 Install a ground wire in the cabinet conduit designated GND on the cabinet top panel.
 - 2 Connect the wire to the lug on the cabinet upright near the exit of the conduit at the bottom of the cabinet.

Note: No more than 2 in (50 mm) of wire should be exposed between the conduit exit and the terminal lug.
 - 3 Connect the other end of the wire to the SL-I ground window.

Note: This wire is not required to run in conduit. If conduit is used, it must be isolated from the cabinet and from the ground window.
 - 3 Install a second ground wire from a terminal lug on the cabinet upright, through the cabinet conduit, to a terminal lug on the QCA13 cabinet upright.

Note: This wire is not required to run in conduit. If conduit is used it must be isolated from both cabinets.
 - 5 Ensure all ground connections are secure.
-

Alarm and Trip **Leads**

2.11 Alarm and trip leads are connected as described in 553-2631-210. These leads may be run in the same conduit as the -48 V leads provided that 2 conductor 16 AWG stranded shielded type wire is used. The shield must be connected to the cabinet frame within 2 in (50 mm) of the exit of the DC PWR conduit in the bottom of the cabinet.

BUSINESS COMMUNICATIONS SYSTEM

SL-1 *

MINI CALL DETAIL RECORDING INSTALLATION

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* SL-1 is a trademark of Northern
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1. GENERAL

1.01 This practice describes the installation procedures for the Mini CDR feature used in SL-1 M systems.

1.02 The Mini CDR feature requires that a second tape unit (QUW9) be installed in the tape shelf (QSD33) of the main SL-1 M cabinet (QCA37). The two tape units are completely separate and each unit is connected to a tape unit interface pack located in the Common Equipment (CE) shelf.

2. INSTALLATION

2.01 Chart 2-1 outlines the procedures for installing the Mini CDR tape unit and tape unit interface pack.

Chart 2-1
MINI CDR TAPE UNIT INSTALLATION PROCEDURES

STEP	PROCEDURE
1	Unpack the Mini CDR tape unit (QUW9) and the tape unit interface circuit pack (QPC33) as described in 553-2001-205.
2	Set the enable/disable switch on the QPC84 pack in the QSD23 power control shelf to DIS.
3	Remove the fuse for 'tape unit 1' from the back of the Power Control shelf (QSD23).
4	Set the switch on the front of the tape unit to the 'OFF' position.
5	Feed the CDR tape unit power cable (P1B) through the side of the tape shelf. This cable runs from the QUX14 power harness.
6	Hold the tape unit up to the tape shelf and plug the power cable (P1B) into the connector at the back of the tape unit.
7	Slide the tape unit into the tape shelf beside the system tape unit. Use the plastic guides to ensure proper positioning of the tape unit (Fig. 2-1).
8	Fasten the tape unit to the shelf by inserting screws into the four front corners of the tape unit.
9	Remove the shipping screw from the front of the tape unit (refer to 553-2001-205).
10	Remove the lower faceplate on the tape unit and slide out the QPC334B circuit pack (first pack on the right hand side). Locate the programming plug on the pack. Ensure that this plug connects pin numbers 1 and 2. (Pins 2 and 3 are connected on system tape units.)
11	Replace the QPC334B circuit pack and the faceplate cover.
12	Set the enable/disable switch on the faceplate of the tape unit interface pack (QPC33) to disable.
13	Set the address switches on the QPC33 to tape unit 1 (Fig. 2-2).
14	Insert and lock the QPC33 into slot position 4 in the front of the CE shelf. Refer to 553-2051-210 for circuit pack installation procedures.
	Note: The CE shelf must be equipped with a second power converter circuit pack (QPC190 or QPC355) located in slot position 1 in the front of the CE shelf.
15	Connect the 5 ft (1.5 m) QCB6 tape unit connector cable from the faceplate of the QPC33 pack to the front of the tape unit.

Chart continued

Chart 2-1 Continued
MINI CDR TAPE UNIT INSTALLATION PROCEDURES

STEP	ACTION
16	Replace the tape unit fuse in the rear of the power control shelf.
17	Set the switch on the QPC84 pack in the QSD23 power control shelf to ENB.
18	Set the switch on the front of the tape unit to the 'ON' position.
19	Label each tape unit to ensure that the correct tape cartridge is always inserted in the appropriate tape unit. <i>Caution: Do not insert the system tape cartridge in the CDR tape unit.</i>
20	Set the enable/disable switch on the QPC33 faceplate to enable.
21	Insert the CDR tape cartridge in the CDR tape unit.
22	Load overlay 17 (Configuration Record) and respond 'new tape 1' to the prompt 'ADAN' (refer to 553-2yy1-220, -221).
23	Assign the CDR options as described in 553-2yy1-220.
24	Load overlay 37 (IOD), to enable the CDR tape unit (refer to 553-2001-505).
25	The operation of the Mini CDR tape unit may be tested using the IOD overlay (refer to 553-2001-505). I/O fault clearing procedures are outlined in 553-2001-545.
26	Refer to 553-2631-311 for the Mini CDR operating procedures.

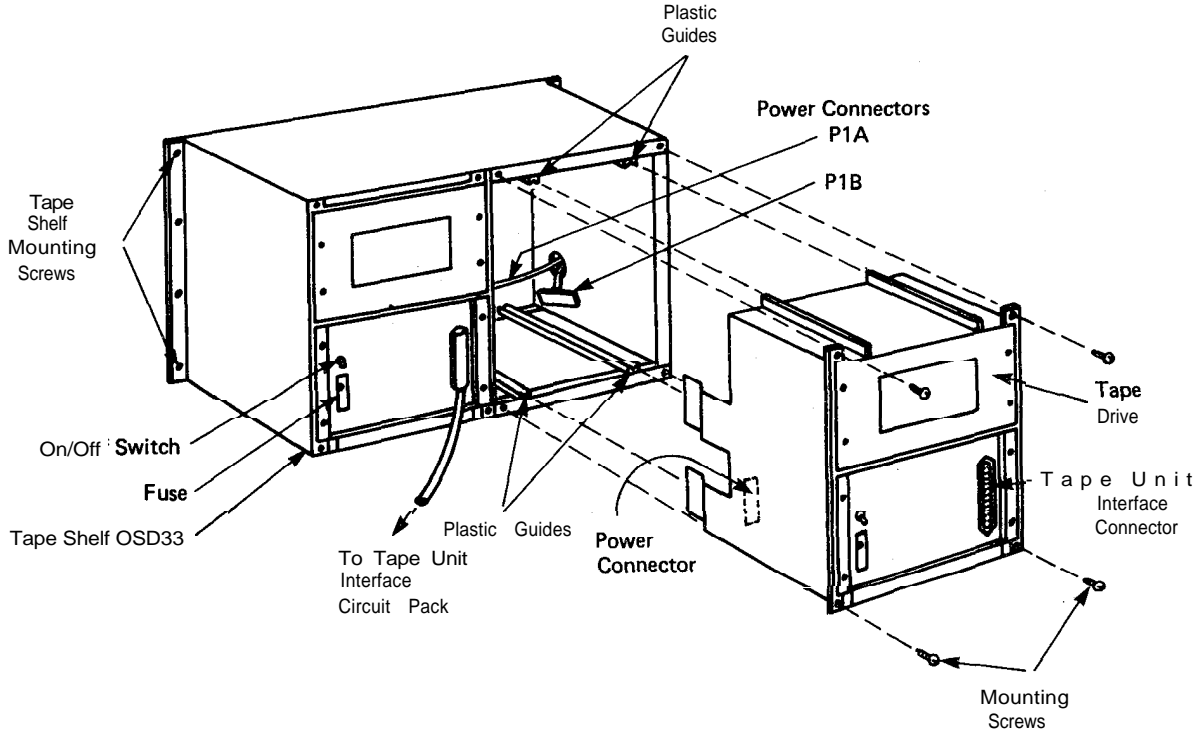
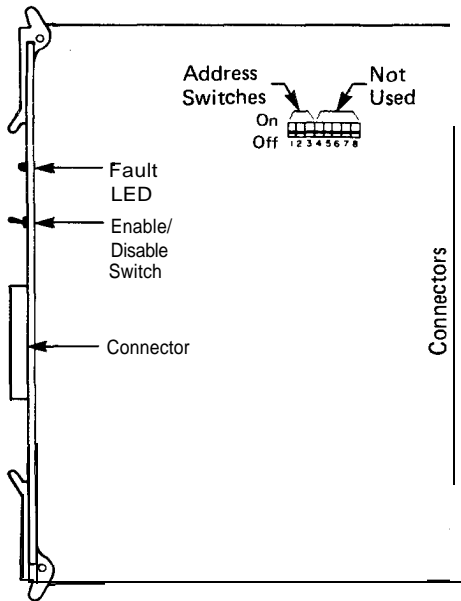


Fig. 2-1
Tape Unit Installation



Each Tape has an address that is set by the address switches on the corresponding QPC33 pack. See Table Below.

In a duplicate CPU System both QPC33 packs which interface with the primary tape unit should have their address switches set for unit 0.

The primary Tape Unit is the tape unit that holds the System Tape. All systems have this tape unit.

The Tape Units 1, 2 and 3 can be added, as required, to perform additional functions such as message recording etc.

Use Tape Unit 1 for the Mini CDR feature.

Tape Unit	Switches		
	1	2	3
0	o	●	o
1	●	o	●
2	●	●	o
3	o	o	o

o Switch Off

● Switch On

Fig. 2-2
QPC33 — Switch Location and Settings

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

SINGLE/MULTI-PORT CDR STORAGE SYSTEMS OPERATION

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* SL-1 is a trademark of Northern
Telecom Limited.

1. GENERAL

1.01 This practice describes the methods of operation for the Single/Multi-Port CDR storage systems. The operations described should be performed as tests on completion of the installation described in 553-2631-210. These tests ensure that the installation meets the applicable operating requirements before being handed over to the customer.

1.02 Satisfactory completion of the operation tests confirms that:

- (a) The apparatus has been properly installed in accordance with the applicable instructions (refer to 553-2631-210).
- (b) All apparatus installed and subjected to testing is functioning correctly.

1.03 If any operation cannot be completed as described.

- (a) Verify that the operation applies to the installation, and that the feature being tested is included in the installation.
- (b) Ensure that the apparatus, which provide the feature or facility under tests, are correctly installed (see 553-2631-210).
- (c) Refer to 553-2631-510 for information concerning fault-finding and repair.

Pretest Requirements

1.04 Before proceeding with operation tests, ensure that the CDR cabinet installation and connections have been made as described in 553-2631-210. Open the inner door of the Magnetic Tape Unit (MTU), check the fuses and red toggle switches on the circuit board on the back of the door; all three toggles must be in the DOWN position. Then, proceed with the operation tests as described in Part 4.

2. SYSTEM MONITORING AND CONTROL

2.01 The CDR system is equipped with a number of buttons and lights which offer limited manual control and visual assessment of the CDR operation. These indicators and controls are outlined in Tables 2-A to 2-D.

Table 2-A
OPERATION CONTROLS AND INDICATORS ON MTU

LABEL	PURPOSE	FUNCTION
RESET	Reset button	Clears any command which is in progress and permits manual commands from other MTU buttons.
	Reset light	Lights up whenever RESET button is pressed.
REWIND	Rewind button	Rewinds tape back to LOAD POINT.
	Rewind light	Lights up while tape is being rewound; goes out once the LOAD POINT has been reached.
ON-LINE	On-Line button	Puts tape ON-LINE, if tape has been loaded.
	On-Line light	Lights up once tape is on-line, and remains lit while tape is on-line.
LOAD	Load button	Tensions tape and moves it forward until LOAD POINT (reflective marker) has been detected.
	Load light	Lights up once the LOAD POINT has been reached. Goes out when LOAD or RESTORE button is pressed on the QPC39.
WRITE ENABLE	Write enable light	Lights up when a tape reel has been put onto the top spool of MTU and the write enable ring attached to the tape reel.
POWER	Power switch	Powers MTU.

Table 2-B
OPERATION CONTROLS AND INDICATORS ON CE SHELF

LABEL	PURPOSE/LOCATION	FUNCTION
ENB/DIS	LED on 6 circuit packs: SDI, MISC, ALU, MEM, CDR TAPE, CDR TIMING.	Lights up whenever the corresponding circuit packs have been disabled.
ENB/DIS	Enable/disable switch on all circuit packs.	Switch enables or disables corresponding circuit pack.
NORM/MAIN	Switch on SEQ pack.	Used in conjunction with maintenance programs to debug the system. Unless specified, this switch should always be in NORM position.
—	Digit display on MISC pack.	Displays fault information in 2-digit hex code. Code can be interpreted by referring to Table 4-A in 553-2631-510.
MAN INT	Manual interrupt.	Clears maintenance display.
LOAD	Button on CDR timing pack.	Initializes tape and commences storing data. Tape should be at LOAD POINT and the MTU on-line before activating LOAD switch. <i>Caution: Pressing the LOAD button causes any data already on tape to be lost.</i>
RESTORE	Button on CDR timing pack.	Commands the CDR to locate end-of-data and to resume writing of call data. New call data is written following any data that is already on tape. Hence, the old data is not lost.
UNLOAD	Button on CDR timing pack.	Outputs any accumulated data onto tape and rewinds tape off button reel. Pressing UNLOAD twice quickly in a multiport CDR will cause accumulated data to remain in the CDR Machine rather than being written onto tape.
ACK	LED on CDR timing pack.	Lights up whenever a function initiated by pressing one of the three faceplate buttons, is in progress. Light goes out once function has been completed.
12/+5 V GRD	Four test points on 5/12-V CONV pack.	Test points are used to determine whether correct voltages are being passed through the converter.
RESET	Reset button on 5/12-V CONV pack.	Resets power into CE shelf.

Table 2-C
OPERATION CONTROLS AND INDICATORS ON
DELTEC INVERTER, QUT3 AND QUX6 UNITS

LABEL	PURPOSE/LOCATION	FUNCTION
	Breaker on Deltec inverter.	Provides input protection for the Deltec inverter.
—	LED on QUT3 inverter.	Indicates ac power failure.
10A	CB2 10-A breaker on QUX6 unit.	Protects common equipment shelf.
-48 v I/P	-48 V breaker on the QUX6 unit.	Provides output protection, -48 V dc to the QUX6 unit.

Table 2-D
OPERATION CONTROLS AND INDICATORS
ON SYSTEM RECTIFIERS

LABEL	PURPOSE/LOCATION	FUNCTION
AC BREAKER	Breaker	Provides input protection.
DC BREAKER	Breaker	Provides output protection (not on QRF8).
DC ON	LED	Lights up when rectifier is outputting dc power.
OUTPUT	Test points	Serve as test points for dc output voltage.
VOLTAGE ADJUST CURRENT LIMIT OVER VOLTAGE ANTI HUNT (QSD4)	Adjustment Screws	Adjust the voltage/current output, set by the manufacturer.

3. OPERATING PROCEDURES

3.01 This part provides the procedures for operating the CDR system.

3.02 Note that the system fail alarm indicates that the CDR data is not being recorded onto tape. The alarm, therefore, might be already activated, but stops once the MTU tape system is recording properly.

3.03 When the multiport CDR tape drive sits idle for 30 minutes, the software causes a quick backspace followed by a forward space over the most recently written block. This is to counteract slippage that is characteristic of the HP tape drive. Note that this operation is performed only after a write operation. It is NOT performed after a RESTORE or LOAD.

Chart 3-1 MOUNTING AN EMPTY TAPE REEL

An empty reel should already be on the bottom spool. If not, put an empty reel onto that spool in the manner shown below.

STEP	PROCEDURE
------	-----------

- | | |
|---|--|
| 1 | Open MTU dust cover door. Use handle and pull door towards you. |
| 2 | Pull spool latch towards you. It should snap open. |
| 3 | Place reel onto spool. Press the reel gently against face of MTU cupboard. |
| 4 | Snap spool latch shut. |
-

PRACTICE 553-2631-310

Chart 3-2
MOUNTING A NEW TAPE REEL

A new reel of tape can be an new unwritten tape, or tape that has been processed and has old data which can be overwritten. Go to Chart 3-5 to remount a tape reel that has data to be preserved.

Note: Keep the tape as clean as possible.

STEP	ACTION	VERIFICATION
1	Clean tape heads (see 553-2631-510).	
2	Check to see whether tape reel is write enabled.	Write enable status is indicated by a plastic ring in the reel.
3	Mount tape reel as described in Chart 3-1.	If tape has the plastic ring, the WRITE ENABLE light located on the panel to the left of the tapes lights up the moment the tape is mounted. This light must be on before the CDR system can begin to record information onto tape.
4	Thread tape through various spindles in the manner shown by the diagram etched into the metal beside those spindles (see Fig. 4-1 of 553-2631-210).	
5	Wind tape around empty reel several times.	
6	To activate a new tape reel go to Chart 3-3; to activate a remounted tape reel (i.e., preserve the data on the tape) go to Chart 3-5.	

Chart 3-3
TAPE UNIT ACTIVATION (New Reel)

STEP	ACTION	VERIFICATION
1	Mount tape and tape reels as shown in Chart 3-2.	
2	Press LOAD button (on MTU) located on panel to left of tapes.	Tape starts to move. When LOAD POINT is reached, the LOAD light comes on, and tape stops moving.
3	Press ON-LINE button located just above LOAD button.	ON-LINE light comes on and stays on as long as the tape is in use.
4	If there is a select strip in the lower left-hand corner of the MTU, make sure that the button beside the 0 light is depressed.	
5	Press LOAD BUTTON (slight touch is sufficient) on CDR timing circuit pack. This initiates the tape.	ACK light below button on QPC39 pack comes on. Tape moves a few inches as it is initialized.
6	When the ACK light goes off, the CDR is recording information from the host SL-1 System.	System fail alarm, if on, also goes off.
7	Test tape unit (see Chart 3-4).	

PRACTICE 553-2631-310

Chart 3-4
TAPE UNIT TESTING (New Reel)

STEP	ACTION	VERIFICATION
1	Load overlay 42 into a host SL-1 machine via host TTY.	CDM000
2	Put program in maintenance mode (see 553-2001-505).	
3	Enter the command TEST at host TTY.	When test is finished, TTY outputs CDM035 N X X X X where: N is the link number. Test fails if X fields are not equal to 0. (Refer to 553-2631-510 for fault clearing procedures.)
4	Retry TEST command until test passes consistently.	
5	Press RESET, REWIND, ON-LINE buttons on MTU to get to load point, and press RESTORE on QPC39 to initialize tape to start recording data.	

Chart 3-5
 REMOUNTING A TAPE

Note: This procedure is required to mount a partially written tape so that the stored data is not lost.

STEP	ACTION	VERIFICATION
1	Mount tape and tape reels as shown in Chart 3-2.	
2	Press LOAD button (on MTU) located on panel to left of tapes.	Tape starts to move. When LOAD POINT is reached, the LOAD light comes on, and tape stops moving.
3	Press ON-LINE button located above LOAD button.	ON-LINE light comes on and stays on as long as the tape is in use.
4	If there is a select strip in the lower left-hand corner of the MTU, make sure that the button beside the 0 light is depressed.	
5	Press RESTORE button (slight touch is sufficient) on CDR timing circuit pack (QPC39). This initiates the tape.	ACK light located below button on QPC39 pack comes on. Tape is read (moves) until last data entry is found. Then tape stops.
6	When the ACK light goes off, the CDR system is ready to receive CDR information from the host SL-1 system.	System fail alarm, if on, goes off.

PRACTICE 553-2631-310

Chart 3-6
REMOVING A TAPE FROM THE MTU

When the tape is full, it should be removed and replaced with a new tape. Tape heads should be cleaned at this time. To remove a tape, follow the procedure outlined below.

STEP	ACTION	VERIFICATION
1	Press UNLOAD button on the CDR timing pack (QPC39).	ACK light comes on. When ACK light goes out, tape should have been rewound off end of bottom reel.
2	Turn top reel several times counterclockwise.	
3	Pull spool latch towards you.	Spool latch snaps open.
4	Remove top reel with tape.	
5	Snap latch shut.	
6	Clean tape heads (see 553-2631-510).	
7	If tape is not full and was removed for maintenance reasons and is to be remounted to collect more data, refer to Chart 3-5. Otherwise, refer to Chart 3-3.	

Chart 3-7

BRINGING UP A CDR CONTROLLER AFTER A POWER FAIL

If a fault, power interruption, or tape operation causes the MTU to go off-line, it does not go on-line automatically, so a manual tape **restoral** is required. If the tape alone is affected during this interval, the CDR data is stored within the controller until all buffers are filled, whereupon subsequent records are lost. This storage capacity is approximately 600 call records in a 8k CDR machine; 2500 in a 32k machine.

STEP	ACTION	VERIFICATION
1	Press LOAD button on MTU.	This tensions tape and starts it moving forward, looking for the LOAD POINT .
2	Quickly hit RESET button on MTU.	This stops tape motion. Tape motion must be stopped because with no LOAD POINT , the tape could eventually run off the reel.
3	Press ON-LINE button on MTU.	ON-LINE light lights up. Tape is now on-line.
4	Press RESTORE button on QPC39 circuit pack.	Tape moves backwards approximately 40 blocks (60 inches) and then reads forward until the last block has been found. ACK light which lit up when RESTORE button was pressed, goes out once the last block has been found. Tape motion ceases, and tape is ready for further data.

Note: The multiport CDR “powers up” with all properly-responding ports enabled.

BUSINESS COMMUNICATIONS
SYSTEM
SL-1*
MINI CALL DETAIL RECORDING
OPERATING PROCEDURES

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* SL-1 is a trademark of Northern
Telecom Limited.

1. GENERAL

1.01 This practice describes the operation of the Mini Call Detail Recording (CDR) feature in SL-1 M systems. Refer to 553-2631-100 for a description of the SL-1 CDR feature.

2. OPERATION

2.01 The operation of the Mini CDR feature is controlled using the Mini CDR overlay. This overlay allows the user to input commands to the system to perform the following functions:

- To initiate the writing of call records on magnetic tape.
- To output the call records stored on tape to a teletypewriter (TTY).
- To install and remove CDR tape cartridges.
- To determine the amount of storage space left on the tape.

CDR POLLING

2.02 Polling the CDR tape is accomplished by loading the Mini CDR overlay on a TTY and entering the DCRA command (Chart 2-4). When this instruction is entered, the SL-1 M system performs the following operations:

- (1) If the CDR buffer contains any call records, and the tape is positioned immediately following the last written block of data, the system writes the buffer contents to tape, followed by an end of file record.
- (2) Rewinds the tape to the load point (beginning of the tape).
- (3) Reads each tape block in turn and transfers it to the system buffer.
- (4) When a tape block is stored in the system buffer, the system translates the call records from the compact binary format to ASCII format and transmits the records to the active TTY.
- (5) Transfers the next tape block to the system buffer.

2.03 This procedure continues until the last tape block has been read or one of the following conditions is encountered:

- More than two wrong tape block types were read.
- More than two tape block numbers were missing during a read.
- A nonrecoverable tape error has occurred.

2.04 If the polling stops due to one of the above errors, a system message (IODXXX or MCRXXX) is generated. Refer to Table 2-B to determine the cause of the error.

CALL RECORDING DURING POLLING

2.05 During the time the polling is in progress, the CDR output to the CDR TTY carry on as normal. The call records scheduled to go to the Mini CDR tape unit are stored in the CDR buffer until the buffer is full. When the buffer is full, the call records are held in the CDR queue. Should call processing run out of call registers' it uses registers from the CDR queue and these call records are lost. It is recommended that polling be done during low traffic periods (refer to 553-2631-111).

Table 2-A
 MINI CDR COMMANDS
 (OVERLAY 89)

COMMAND	DESCRIPTION
DCRA	Request CDR ASCII dump. This dumps CDR tape records in ASCII format starting at record 1 and ending at the last tape record successfully written (inclusive). Only ASCII dump is available, i.e., no binary dump can be performed.
SPCR	Spool CDR tape. This command spools the CDR tape to the early warning point and then back to the load point. This command is used to retension the tape. It also allows the system to compute the tape length and determine how full the tape is (see TPCD command). This command is only allowed when the Mini CDR tape is not in a writable position (e.g., after a system reload, UNLD or DCRA commands).
TPCD	How full is the CDR tape? The response is a percentage of the total CDR tape capacity. This tape capacity calculation uses the 3600 inch tape unless the SPCR command is issued while installing the tape or the first track of the tape is full.
LPCD	Restart CDR writing from the load point (beginning of tape).
EFCD	Restart CDR writing from End of File (EOF) (the last successfully written tape record).
UNLD	Prepare the tape for unloading. Before the tape is physically removed from the drive, the CDR buffer and EOF mark are written on the tape. The crafts person should remove the tape only after the system issues the special message "OK" indicating that the tape is ready to be unloaded.
'*'	End the current command without aborting the overlay. The system prompts for a new command once the current command is aborted.
'****'	Abort the overlay.

Chart 2-1
LOADING THE MINI CDR OVERLAY

STEP	ACTION
1	Login to any TTY (local or remote). Password level 1, level 2 or the 'Mini CDR only' password allows access to the Mini CDR overlay.
2	Load overlay 89 by entering 'LD 89'.
3	After successfully loading the overlay, the system issues the prompt requesting a command input (Table 2-A).

Chart 2-2
INSERTING A NEW CDR TAPE

STEP	ACTION
1	Load Mini CDR overlay (Chart 2-1).
2	If there is an old tape in the tape drive, remove the tape by using the proper tape removal procedure in Chart 2-3.
3	Insert the new tape.
4	Issue the spool command (SPCR) to retension the tape and to enable the system to compute the tape length.
5	Issue the command to start CDR writing at the load point (LPCD). This causes the tape to rewind to the load point where normal CDR writing begins.

Chart 2-3
REMOVING A CDR TAPE

STEP	ACTION
1	Load the Mini CDR overlay (Chart 2-1).
2	Enter the unload command (UNLD). This causes the contents (if any) of the CDR buffer to be written on tape followed by an End of File (EOF) record. The tape then rewinds to the load point.
3	Wait for the completion message ('OK') on the TTY.
4	Remove the tape cartridge from the tape unit.
5	A new tape may now be inserted (Chart 2-2).

Chart 2-4
 POLLING THE CDR TAPE

STEP	ACTION
1	Load the Mini CDR overlay (Chart 2-1).
2	Enter the CDR ASCII dump command (DCRA). This causes the system to output the call records to the active TTY in ASCII format. Refer to text for the steps involved in this operation. The polling activity may be aborted at any time by entering '*'.
3	Wait for the 'DONE' message indicating that the polling is complete. <i>Note:</i> If an error condition occurs a system message (IODXXX or MCRXXX) is generated and polling activity is aborted. Refer to Table 2-B to determine the cause of the error. See also 'Additional Aspects for CDR Polling' below.
4	Restart normal CDR writing by entering the LPCD command. The user may install a new tape cartridge first (Chart 2-2). If CDR is not explicitly restarted after polling, the CDR recording will begin from where it left off previously (i.e., EOF) when the overlay is aborted.
5	To exit from the Mini CDR overlay, enter '****'. <i>Note:</i> There is also an automatic exit mode. The overlay is automatically aborted by the system 20 min after the conclusion of the last command.

Additional aspects in CDR polling:

Abort Remote Dump. Any other TTY in the system is able to abort the on-going polling activity by logging in. There will not be an indication to the aborting TTY that CDR polling is in progress other than normal "OVL1 11 TTY X" message.

Removal of CDR Tape During CDR Polling. The removal of the CDR tape during CDR polling will cause the abortion of the on-going CDR polling activity. A special message IOD254 indicating "TAPE EJECT" will be sent to the accessor.

System Initialization During CDR Polling. System initialization during CDR polling will cause the termination of CDR polling without warning or any special message to the accessor. The accessor will have to reinitiate the polling activity after the system recovery.

Remote Access Request Without CDR Tape Mounted. If the CDR polling is requested without a CDR tape mounted, a special message IOD254 indicating "TAPE EJECT" will be sent to the accessor.

Loss of CDR Polling Accessor. If the accessor is disconnected during dump in the ASCII format, the SL-1 can not detect the disconnect and the polling activity will continue to completion unless aborted through the reestablished link. After the link is reestablished the accessor must abort the ongoing polling, if any, with a '*', before entering another command.

2.06 The system messages shown in Table 2-B are generated by either the resident software or the Mini CDR overlay.

Table 2-B
SYSTEM MESSAGES

CODE	DESCRIPTION
OK	CDR tape can be unloaded.
DONE	Command successfully done.
MCROOO	SYS ID code.
MCR001	Invalid command.
MCR002	Invalid command argument.
MCR003	Mini CDR overlay load illegally requested by system.
MCR004	Mini CDR not in configuration, or feature not enabled.
MCR005	Tape cannot be positioned to EOF due to sysload.
MCR006	CDR records missing from tape. Bad records will be skipped.
MCR007	Invalid tape record type found during polling.
MCR008	Spool command not allowed as tape is in writable position.
MCR009	Tape percent unknown due to sysload.
MCR010	During CDR polling more than 2 contiguous bad records (non-CDR records) found, or tape records missing. Polling is aborted.
IOD050 1	CDR tape not ready.
IOD051 1	Timeout or early warning encountered while reading CDR tape.
IOD100 1 N	Head fault or bad record on track N.
IOD101 1 N	Requested data record not found on track N.
IOD110 1	Requested data record not found on either track.
IOD200 1	Tape disabled detected during read/write attempt. Current activity suspended.
IOD201 1	Tape not equipped detected during read/write effort. Current activity suspended.
IOD203 1	Tape not ready detected during read/write effort. Current activity suspended.
IOD204 1	Tape write protected detected during write effort. Write effort suspended.
IOD205 1	Early warning detected during write effort. Write effort suspended if track 3.
IOD206 1	Read error detected during write effort. Hardware fault suspected; write effort suspended.

Table 2-B Continued
SYSTEM MESSAGES

CODE	DESCRIPTION
IOD207 1	Write error detected during write effort. Write effort suspended after retries fail.
IOD208 1	Record not found error detected during read effort. Read effort suspended.
IOD209 1	Timeout detected during read/write effort. Current activity suspended.
IOD210 1	Tape is put in tape full state. Write effort suspended.
IOD211 1	CDR tape has been locked for too long. Write effort suspended.
IOD212 1	Mini CDR tape is now 75% full.
IOD250 1	Tape response idle while erasing. Hardware fault suspected.

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

SINGLE/MULTI-PORT CDR STORAGE SYSTEMS FAULT-CLEARING PROCEDURES

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* SL-1 is a trademark of Northern
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1. GENERAL

1.01 This practice describes how to maintain the Single/Multi-Port CDR systems. This consists of both preventive and corrective maintenance.

(a) Preventive maintenance requires:

- replacement of the filter pad on the Deltec static inverter
- ┆ clearing and replacement of certain parts of the Magnetic Tape Unit (MTU) at regular intervals
- degaussing of tape heads.

(b) Corrective maintenance is necessary when the system fails to function within design capabilities and limits. Faults and deterioration can be traced to apparatus (circuit packs, shelves, etc.), which is then replaced and sent for repair.

2. PREVENTIVE MAINTENANCE

Deltec Static Inverter 2.01 The fan filter inside the inverter (Fig. 2-1) should be cleaned periodically, depending on the environment. To gain access to the filter, use a screwdriver to remove the filter cover. Then, clean the filter with a soft brush in a room other than the one occupied by the SL-1 and CDR equipment. Replace the filter by fitting it into the recessed cover and pressing the cover back over the inverter unit.

Magnetic Tape Unit (HP7970E) 2.02 To maintain the MTU, a mandatory maintenance program? periodic cleaning and regular degaussing of the tape heads are required.

2.03 *Maintenance Program.* To ensure reliable operation of the MTU at its optimum potential, a scheduled preventive maintenance program is MANDATORY. The customer must arrange a detailed periodic maintenance (see tape drive manual) with the tape drive manufacturer on a call or contract basis.

2.04 *Cleaning.* The MTU requires cleaning in the following areas: head and associated guides, roller guides, tape cleaner, and capstan. To clean the head and head guides, use a lint-free cloth or cotton swab moistened in isopropyl alcohol or Du Pont Freon IF or equivalent. ←
Wipe the head carefully by rubbing gently in the direction of tape travel ← to remove all accumulated oxide and dirt.

2.05 It is recommended that the tape head and capstan be cleaned every four to six weeks and whenever a new tape is mounted. ←

2.06 *Degaussing Procedure.* The MTU tape head must be degaussed regularly to minimize residual magnetism. This should be done approximately every 3 months using the degaussing tool (Part No. CO052473).

- (1) Remove magnetic tape from the MTU.
- (2) Remove ac power from the MTU.
- (3) Hold back the cross-talk gate (Fig. 2-2), insert the degausser parallel to the face of magnetic head, and pull the degausser slowly away from the tape transport.

Caution: Do not allow the degausser to come into contact with the face of the head. If it contacts the head, the head may be magnetized.

- (4) Return the MTU to operation.

2.07 For the other work concerned with preventive maintenance, refer to Hewlett-Packard specification HP7970 series, Magnetic Tape Drives operator's manual. This manual is delivered with each MTU.

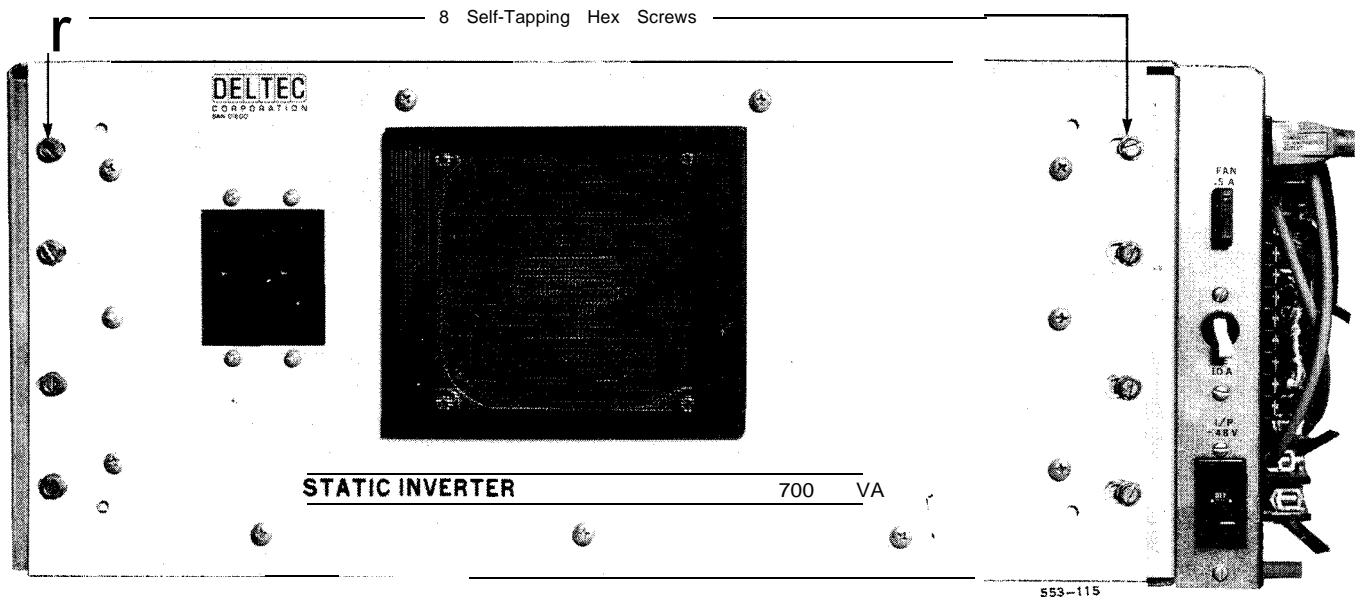


Fig. 2-1
Deltec Static Inverter

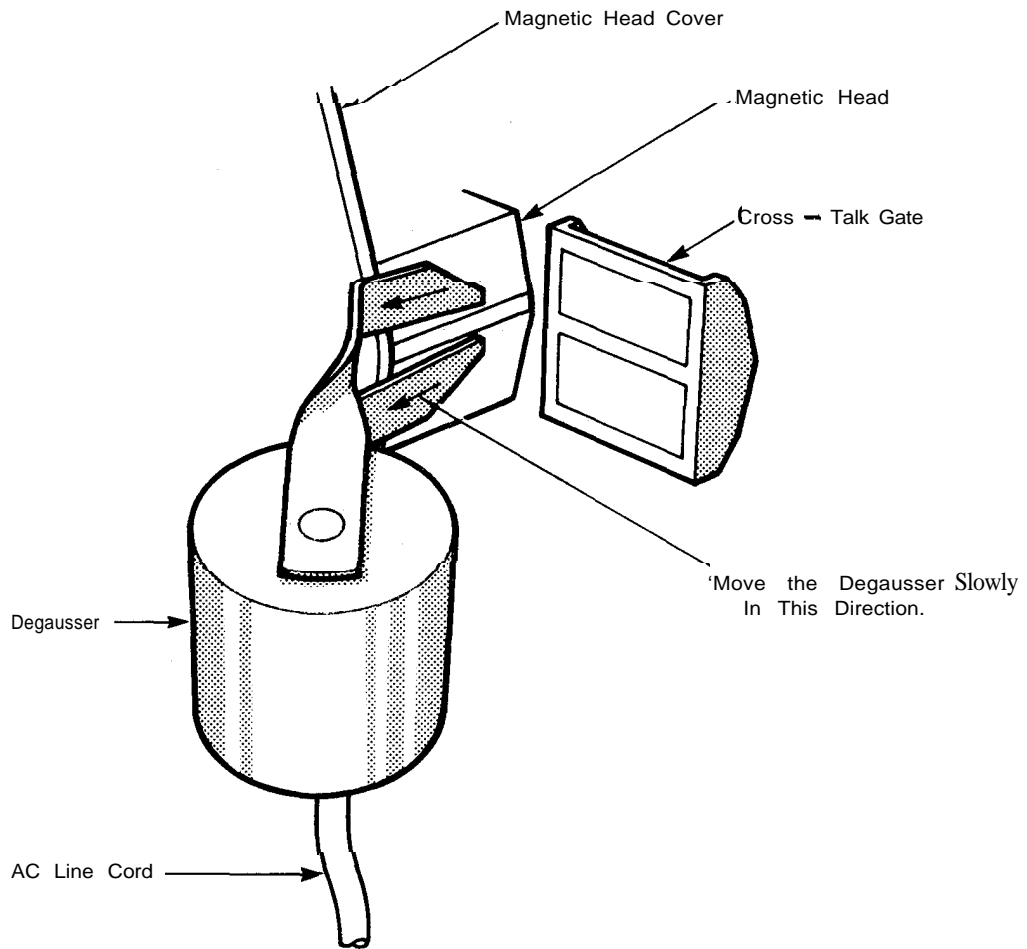


Fig. 2-2
Hand-Held Degausser

3. CORRECTIVE MAINTENANCE

Precautions

3.01 When clearing faults in the CDR system, the following general precautions should be taken.

3.02 *Circuit Packs*

- Common Equipment (GE) packs with ENABLE/DISABLE switches should be disabled before being removed from or inserted into a shelf. When replacing Serial Data Interface (SDI) packs in a multiport CDR machine, overlay 42 may be used to allow the replacement to be done without interruption of service.
- Handle circuit packs by the card stiffeners and edges of packs only. Do not handle circuit packs by the contacts or components.
- Circuit packs that contain static sensitive components have a caution label in the package. These circuits require the following handling precautions:
 - (a) Do not set pack down on any surface other than the protective antistatic bag supplied in the shipping carton.
 - (b) Do not unpack or handle circuit pack near such machinery as electric motors or transformers.

3.03 *Tape Reel.*

- Avoid touching any portion of the magnetic tape that contains electronic information.
- Do not subject a magnetic tape to rapid changes in temperature and humidity or to strong magnetic fields.

3.04 For more detailed precautions refer to 553-2yy1-500.

Fault Indications

3.05 The state of the CDR system may be indicated by any one of the following:

CDM error messages at host SL-1 systems
 MTU
 Tape Movement
 RESET Light
 REWIND Light
 ON LINE Light
 LOAD Light
 CDR CE Shelf Light Emitting Diodes (LEDs)
 Maintenance Display
 Circuit Pack LED
 ACK LED on QPC39
 Rectifier
 DC ON Light

3.06 *Low-Tape Alarm.* The low tape alarm is a contact closure which is completed by the CDR system whenever an existing tape has recorded more than 11 000 blocks of data (2048 bytes each); i.e., has written about 90 percent of a 2400-foot tape.

3.07 System **Fail** Alarm. The system fail. alarm is a contact closure which is completed whenever the CDR is unable to write onto the magnetic tape. This can occur, for example, after a power failure in the CDR tape controller or tape unit, when a tape restore operation cannot be completed, or tape is not write enabled.

3.08 Fan Failure and Overheating. Whenever a fan failure or overheating is detected in the CDR system, each host SL-1 types a message and raises the minor alarm.

Fault Clearing

3.09 Faults are indicated by any combination of the host SL-1 monitoring circuitry, system alarms or customer complaints.

3.10 A system fault can be pinpointed to the CDR by following step-by-step flowchart instructions in 553-2yy1-515. Flowcharts 5-1 through 5-4 of this practice can then be used in conjunction with diagnostic overlay programs to clear the CDR fault. A specific CDR system fault can be indicated by any number of fault indications listed above.

3.11 The CDR diagnostic overlay programs (553-2001-505) are provided on the SL-1 system tapes to aid operators in locating system faults. These diagnostics can be called upon either automatically (overlay program 40) or manually (overlay program 42) to locate and isolate the fault.

3.12 Once the defective apparatus has been isolated and replaced, finish the fault clearing process by following the final maintenance procedures listed in 553-2yy1-500.

Returning Defective Apparatus

3.13 Defective apparatus should be replaced with apparatus known to be good, and defective items tagged, packed (see 553-2001-205), and returned to a repair center. Do not attempt to repair circuit packs.

4. CALL DETAIL RECORDING MAINTENANCE DISPLAY CODES

4.01 On the QPC41 MISC pack of the CDR CPU, there are two LED display panels. In case of faults, a combination of letters and numbers (i.e., a “display code”) appears on the panels, the top being the first and the bottom being the second character in the code. For an explanation of each code, see Table 4-A.

Table 4-A
CDR MAINTENANCE DISPLAY CODES

CODE	INTERPRETATION								
00	Trap caused by system power reset. This code is provided for information only.								
01	<p>The CPU failed the CPU test.</p> <p>If CDR is not running, then suspect one or more of the following packs:</p> <table style="margin-left: 40px;"> <tr> <td>ALU</td> <td>QPC40</td> </tr> <tr> <td>MISC</td> <td>QPC41</td> </tr> <tr> <td>SEQ</td> <td>QPC42</td> </tr> <tr> <td>ROMS</td> <td>QPC131, QPC132, QPC234, QPC235, or QPC301</td> </tr> </table> <p>If none of these packs clear the fault, then check all other packs on the shelf.</p> <p>See flowchart 5-3 for CPU fault clearing procedures.</p> <p>If CDR is running, then an intermittent fault is present. Use overlay 42 to clear the CDR maintenance display or press the MAN INT button on multi-port CDR machines. If the problem recurs then suspect the packs listed above.</p>	ALU	QPC40	MISC	QPC41	SEQ	QPC42	ROMS	QPC131, QPC132, QPC234, QPC235, or QPC301
ALU	QPC40								
MISC	QPC41								
SEQ	QPC42								
ROMS	QPC131, QPC132, QPC234, QPC235, or QPC301								
02, 03	<p>The ROM failed to checksum correctly.</p> <p>If CDR is not running, then suspect the following packs:</p> <table style="margin-left: 40px;"> <tr> <td>02</td> <td>CDR ROM 1</td> <td>QPC301, QPC131, or QPC234</td> </tr> <tr> <td>03</td> <td>CDR ROM 2</td> <td>QPC301, QPC132, or QPC235</td> </tr> </table> <p>If neither of these packs clear the fault, then it may be a processor fault. Respond as if 01 were in the display. One QPC301 pack may be used to replace the QPC234 and QPC235 pair.</p>	02	CDR ROM 1	QPC301, QPC131, or QPC234	03	CDR ROM 2	QPC301, QPC132, or QPC235		
02	CDR ROM 1	QPC301, QPC131, or QPC234							
03	CDR ROM 2	QPC301, QPC132, or QPC235							

Table Continued

Table 4-A Continued
 CDR MAINTENANCE DISPLAY CODES

CODE	INTERPRETATION
	<p>If CDR is running, then an intermittent fault is present. Use overlay 42 to clear the CDR maintenance display.</p> <p>If the problem reappears, then suspect the packs listed above.</p> <p>If the problem reappears and persists, treat it as a code 01.</p>
05	<p>Trap caused by response timeout.</p> <p>A device suddenly failed to respond. This could happen if an SDI card was removed without first using LD 42 to disable the card.</p> <p>If this problem is intermittent, a card may not be properly secured into the CDR machine, or a card may be faulty.</p> <p>If the problem still occurs, respond as for code 01.</p>
06	Trap caused by write protection violation.
07	Trap caused by watchdog timeout.
	<p>These codes probably indicate a CPU problem. Suspect cards in the same order as for code 01.</p>
08	<p>Trap caused by memory parity error.</p> <p>If this recurs after clearing the maintenance display, respond as for code 10.</p>
09	<p>Trap for undeterminable causes.</p> <p>This code probably indicates a CPU problem. Suspect cards in the same order as for code 01.</p>
10	<p>The read/write memory failed the memory test.</p> <p>If CDR is not running, then suspect the following packs:</p> <p>R/W MEM QPC31, QPC51, QPA62</p> <p>If CDR is running, then an intermittent fault exists. Use LD 42 to clear the CDR maintenance display.</p> <p>If the fault recurs, then suspect the packs listed above followed by the packs listed for code 01.</p>
20	CDR timing did not respond.
21	CDR timing had a stuck interrupt.
	<p>Codes 20 and 21 suspect:</p> <p style="padding-left: 40px;">CDR TIMING QPC39</p>

Table Continued

Table 4-A Continued
CDR MAINTENANCE DISPLAY CODES

CODE	INTERPRETATION
	<p>If CDR is not running, then see flowchart 5-3.</p> <p>If CDR is running, then an intermittent fault exists. Use overlay 42 to clear the CDR maintenance display.</p> <p>If the problem recurs, then suspect the QPC39 pack followed by all the packs listed for code 01.</p>
30	<p>The requested sequence is already being performed.</p> <p>Wait for it to complete.</p>
31	<p>The requested function can not be performed because another request is still pending.</p> <p>Wait for the pending function to complete or press the UNLOAD button to terminate it.</p>
32	<p>The requested function (RESTORE or LOAD) can not be performed because CDM has been loaded and has put the CDR machine in MAINTMODE.</p> <p>If it is desired to terminate the MAINTMODE, then UNLOAD may be pressed to unload the tape.</p>
33	<p>RESTORE can not be performed because a LOAD or UNLOAD sequence is in progress; wait for the sequence to complete.</p>
34	<p>LOAD can not be performed because the drive is not disabled.</p> <p>If another function is being performed, then wait for it to complete, and try again</p> <p>If the drive is not disabled, then you must UNLOAD the tape before you can LOAD it. LOAD will destroy any data already written on the tape.</p>
35	<p>LOAD can not be performed because the tape is not at load point.</p> <p>UNLOAD and remount the tape. LOAD will destroy any data already on the tape.</p>
40	<p>Cabinet over temperature has been detected.</p> <p>Take appropriate temperature lowering measures.</p>
41	<p>Fan failure has been detected in CDR cooling unit.</p> <p>Clean filters and/or replace fan assembly.</p>
50	<p>LOAD can not be performed because the tape is write protected.</p> <p>CDR will unload the reel of tape.</p> <p>Install a write enable ring in the supply reel and remount the tape.</p>

Table Continued

Table 4-A Continued
 CDR MAINTENANCE DISPLAY CODES

CODE	INTERPRETATION
51	<p>LOAD can not be performed because the tape unit is not ready. This may be because it is rewinding.</p> <p>Wait for rewind to complete and then make unit ready. Retry request.</p>
52	<p>LOAD failed after 10 tries.</p> <p>CDR will unload the tape.</p> <p>Retry the request with a different tape. If the problem persists then:</p> <ol style="list-style-type: none"> 1) Clean the tape head path. 2) The tape unit may require calibration or be faulty. 3) Suspect: QPC130 TAPE CONTROL QPC39 TIMING Cables <p>See flowchart 5-2 for tape fault clearing procedures.</p>
53	<p>UNLOAD failed after 10 tries. CDR gives up.</p>
54	<p>The restore function failed after 4 tries.</p> <p>CDR will wait 8 hours and then it will try again.</p> <p>This code indicates that the error recovery attempt has failed.</p> <p>Attempt to RESTORE the tape. If restore is successful, then use LD 42 to clear the maintenance display or press MAN INT on a multi-port CDR machine.</p> <p>If the RESTORE fails, then try cleaning the tape heads and path. Try RESTORE again. If it fails then try LOADING a new tape. If this is successful, then send the old tape for processing.</p> <p>If the LOAD fails, then suspect one or more of the following items:</p> <ol style="list-style-type: none"> 1) tape drive — may require calibration or repair 2) QPC130 tape interface 3) QPC39 timing 4) cables <p>If the problem is intermittent, suspect the items listed above.</p> <p>Since CDR will try the restore function again after an 8 hour wait, the CDR machine may be recording data if more than 8 hours has elapsed since the code was originally put in the display. This would indicate that the fault is intermittent.</p>

Table Continued

Table 4-A Continued
CDR MAINTENANCE DISPLAY CODES

CODE	INTERPRETATION
55	<p>End of tape encountered while doing a RESTORE.</p> <p>CDR will UNLOAD the tape.</p> <p>The tape can not be RESTORED since the tape mark at the end of the data can not be found.</p> <p>If the tape has data on it then it should be sent for processing and a new tape mounted and LOADED to collect new data.</p> <p>If the problem persists a faulty tape unit is suspected.</p> <p>This problem can also be caused by trying to RESTORE a blank tape.</p>
56	<p>Write protect was encountered while doing a RESTORE.</p> <p>Ensure that the write enable ring is installed.</p> <p>If the problem persists a faulty tape reel or tape drive is suspected.</p>
57	<p>End of tape encountered while writing.</p> <p>CDR will UNLOAD the tape.</p> <p>Mount a new tape and LOAD it. Send the tape that was removed for processing.</p> <p>Depending on the period of time since the end of tape occurred data may or may not have been lost.</p>
70	<p>CDR tape control pack did not respond.</p>
71	<p>CDR tape control has a stuck interrupt.</p> <p>Codes 70 and 71 suspect:</p> <p>CDR TAPE CONTROL QPC130</p> <p>If the CDR is not running, refer to flowchart 5-3.</p> <p>If the CDR is running, then an intermittent fault exists. Use overlay 42 to clear the CDR maintenance display, or press MAN INT on a multi-port CDR machine.</p> <p>If the problem recurs, then suspect the packs listed above followed by all packs listed for code 01.</p>

Table Continued

Table 4-A Continued
 CDR MAINTENANCE DISPLAY CODES

CODE	INTERPRETATION
80	SDI response time out.
81	SDI has permanent interrupt.
82	SDI line is not ready.
83	SDI lost an output interrupt.
	Codes 80, 81, 82, 83, suspect: QPC45 or QPC139 SDI. Ensure that the speed select, address select, and programming plugs are properly set. If CDR is not running, refer to flowchart 5-3. If CDR is running, then an intermittent fault exists. Use overlay 42 to clear the CDR maintenance display, or press MAN INT. If the problem recurs, then suspect the packs listed above followed by all packs listed for code 01. Code 82 can also be caused by a faulty modem or cable.
Ax (See Note)	TTY x does not exist This code will be displayed if the parameter on an ENL SL1 command refers to an SDI port which does not exist. Either the parameter was incorrectly specified or the address switches on the SDI card were set incorrectly. This error can also occur with a faulty SDI card.
Bx (See Note)	TTY x is not getting output interrupts. The SDI card should be replaced. If this does not cure the problem, act as for code 01.
c x (See Note)	“EIA Device Not Ready” status on TTY x. Unless the SDI card is faulty, this error indicates: a) There is no connector attached to the faceplate of the SDI unit, or b) A plug on the SDI plug board is in the wrong position, or c) A plug on the SDI board is missing. Check plug positions.

Table Continued

Table 4-A Continued
CDR MAINTENANCE DISPLAY CODES

CODE	INTERPRETATION
D x (See Note)	<p>Stuck interrupt on TTY x.</p> <p>Replace the SDI card.</p> <p>If this proves to be of no avail, treat the problem the same as for code 01.</p>

Note: This display code identifies a problem in port x (0 through 15), where x is indicated as a hexadecimal device number in the range 0 to F:

Hex Display	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
Port	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15

5. FAULT-CLEARING PROCEDURES

5.01 The CDR system fault clearing can be divided into three categories:

- link faults
- CDR CE faults
- faults affecting the CDR tape drive.

5.02 These faults are classified by using Flowchart 5-1. Fault clearing of these categories is covered in Flowcharts 5-2 through 5-4 at the end of this practice.

5.03 To replace circuit packs in the CDR system refer to Chart 5-1 and 5-2.

Chart 5-1
 CIRCUIT PACK REPLACEMENT PROCEDURES (Non-SD1 Pack)

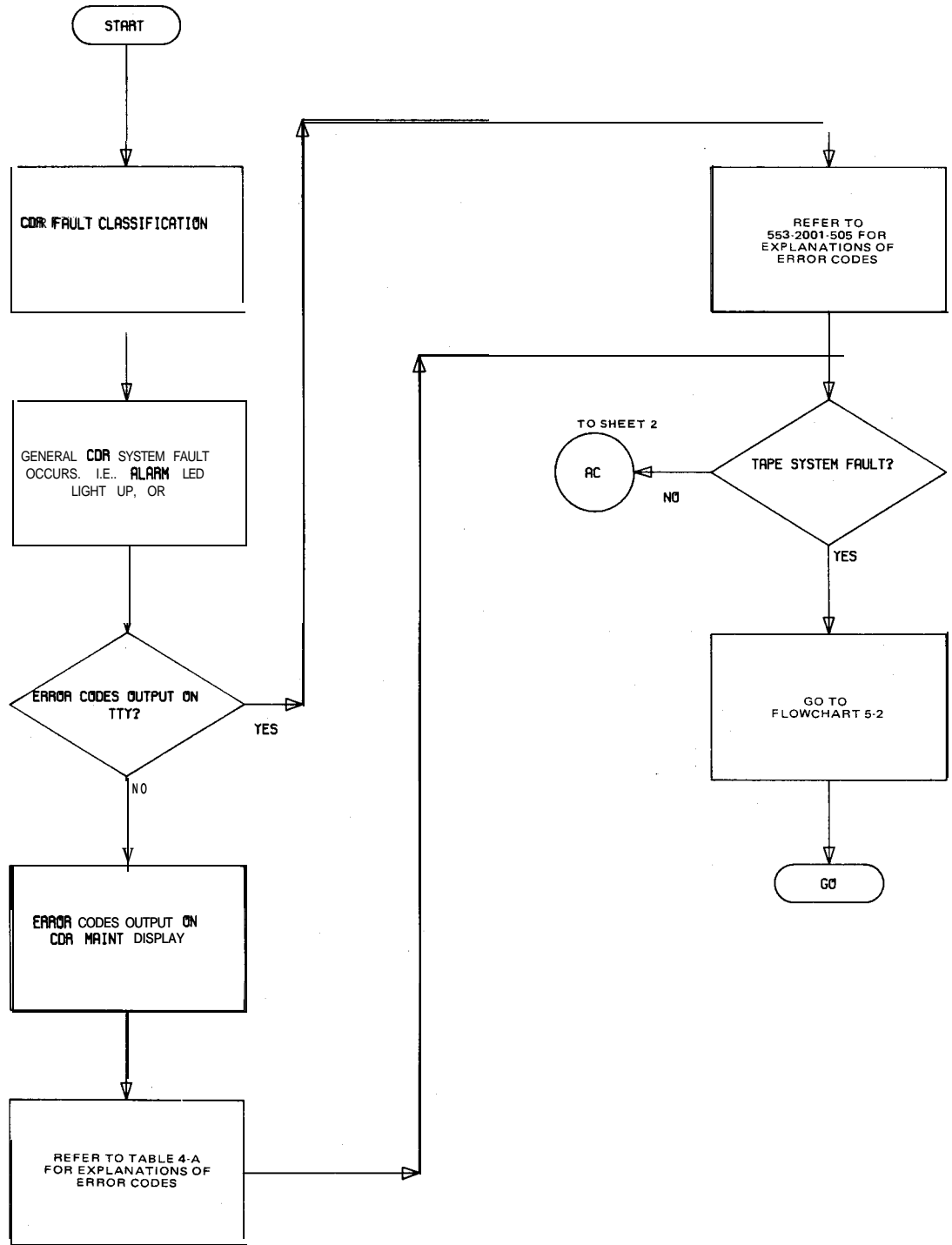
This procedure also clears the fault LEDs.

STEP	PROCEDURE	RESULT
1	Disable links by using program 42 command DIS LINK N. This should be done in every SL-1 system connected to the CDR.	
2	Press UNLOAD button on QPC39 timing pack.	Loads data onto tape and rewinds it off the reel.
3	Switch ENB/DIS switch on pack faceplate to DIS and remove pack.	
4	Replace circuit pack.	
5	Switch ENB/DIS switch on pack faceplate to ENB.	
6	Disable, then enable QPC42. The LED on the pack should extinguish. If it remains lit, a fault is indicated and fault clearing procedures are called for.	
7	Re-thread tape in MTU.	
8	Press LOAD and ON-LINE buttons on MTU.	Brings drive on line and to the LOAD POINT.
9	Press RESTORE button on QPC39 pack.	Finds end of written data on the tape, and positions the tape to write more data.
10	Enable links via program 42 command ENL LINK N in every connected SL-1 system.	This minimizes the loss of data.

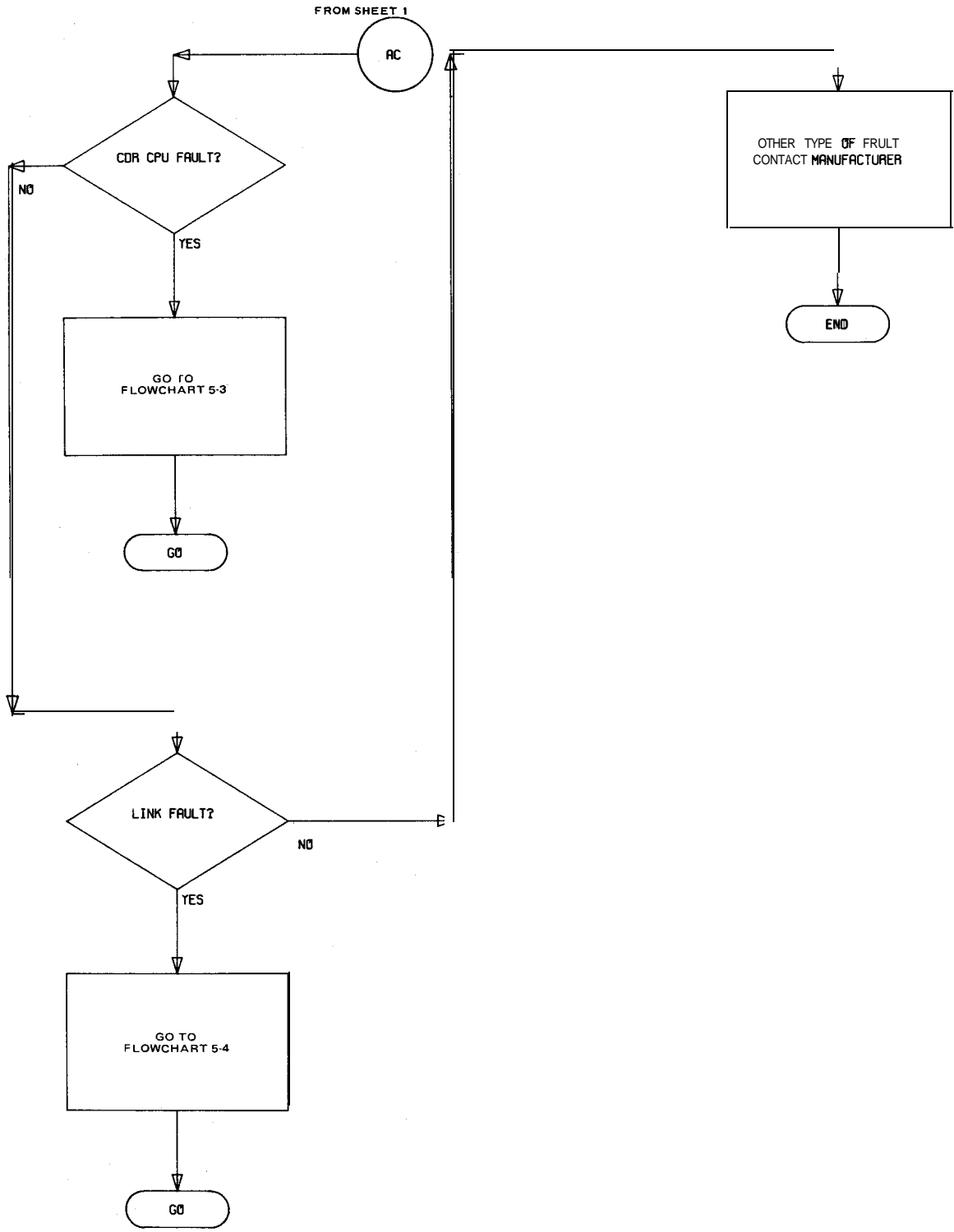
Chart 5-2
CIRCUIT PACK REPLACEMENT PROCEDURES (SDI Pack)

The following procedure is used in the multi-port CDR for replacing QPC45 or QPC139 SDI cards without interrupting service to any SL-1 system (except those connected to the effected SDI card).

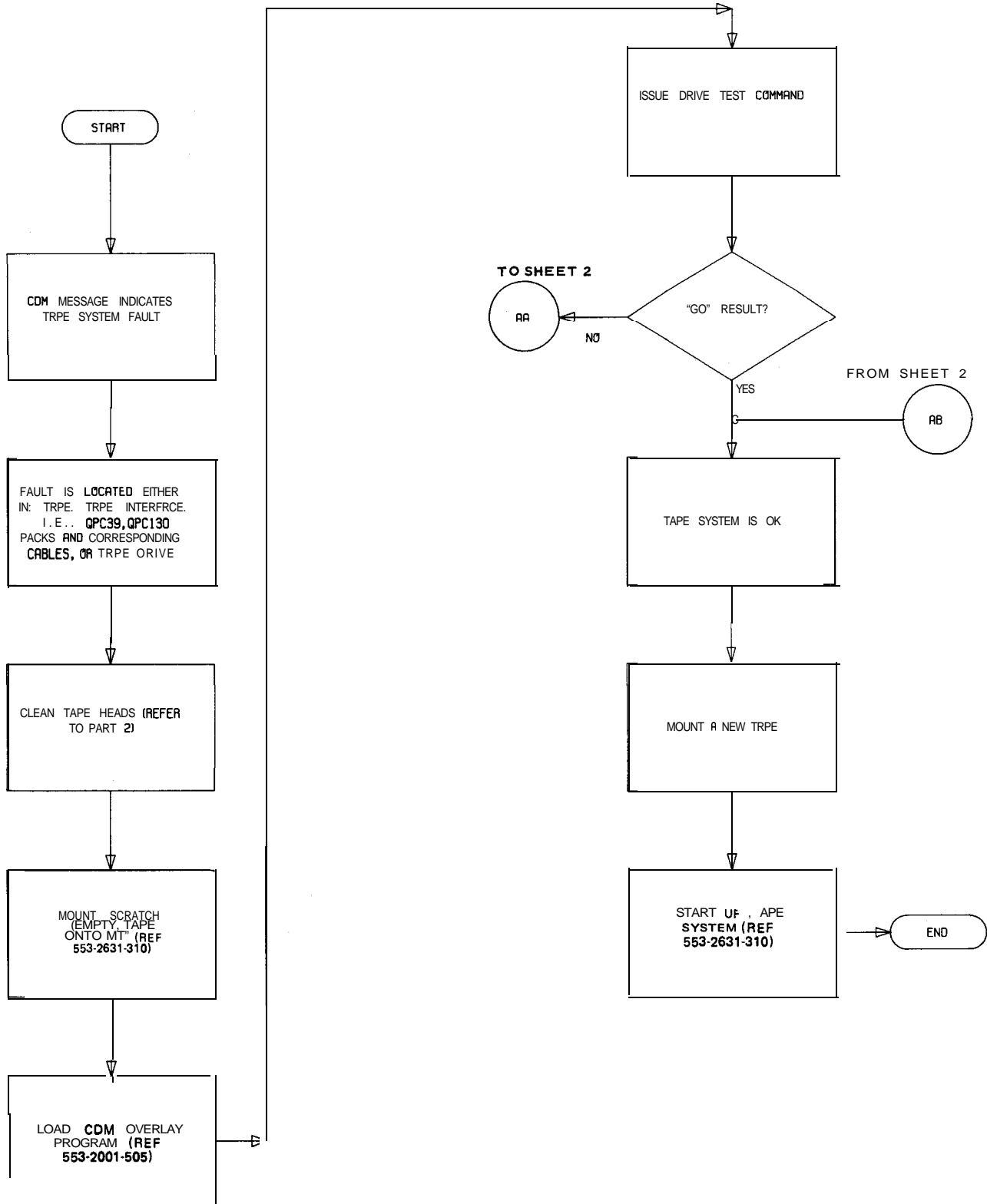
STEP	PROCEDURE
1	Disable the link in the SL-1 system by using DIS LINK N command from overlay 42. For QPC139, this has to be done in both SL-1 systems.
2	Using overlay 42 in some other SL-1 system, disable the effected SL-1 port(s) in the CDR by issuing the command DIS SL1 m, where m is the device number of the SL-1 port on the CDR SDI card.
3	Switch the ENL/DIS toggle on the QPC45 or QPC139 to DIS.
4	Replace the card.
5	Enable the new card with the toggle switch.
6	Issue ENL SL1 m from the SL-1 system that issued DIS SL1 m.
7	Issue ENL LINK n from each SL-1 system whose port in the CDR machine was affected.



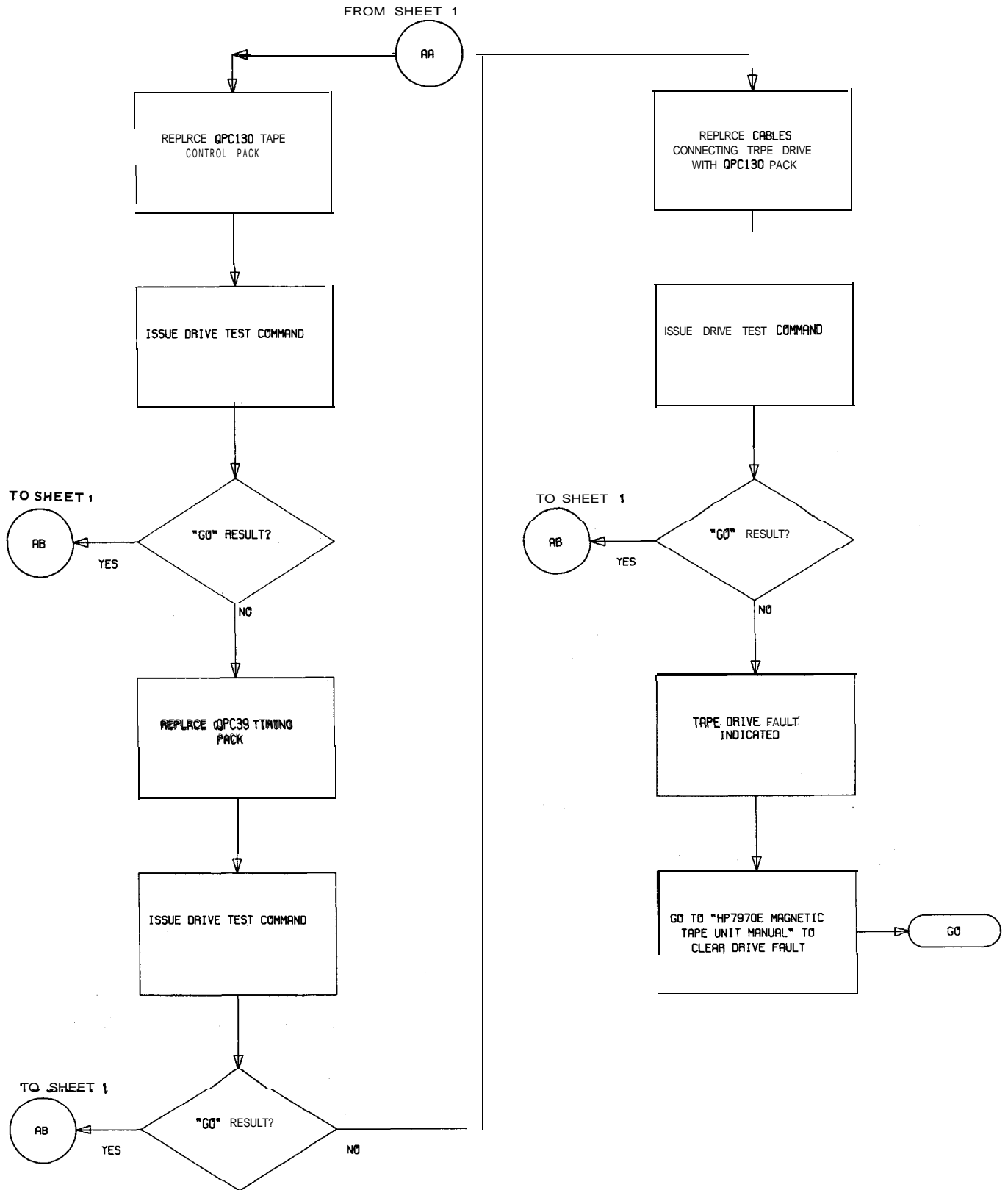
Flowchart 5-1
CDR Fault-Classification Procedures



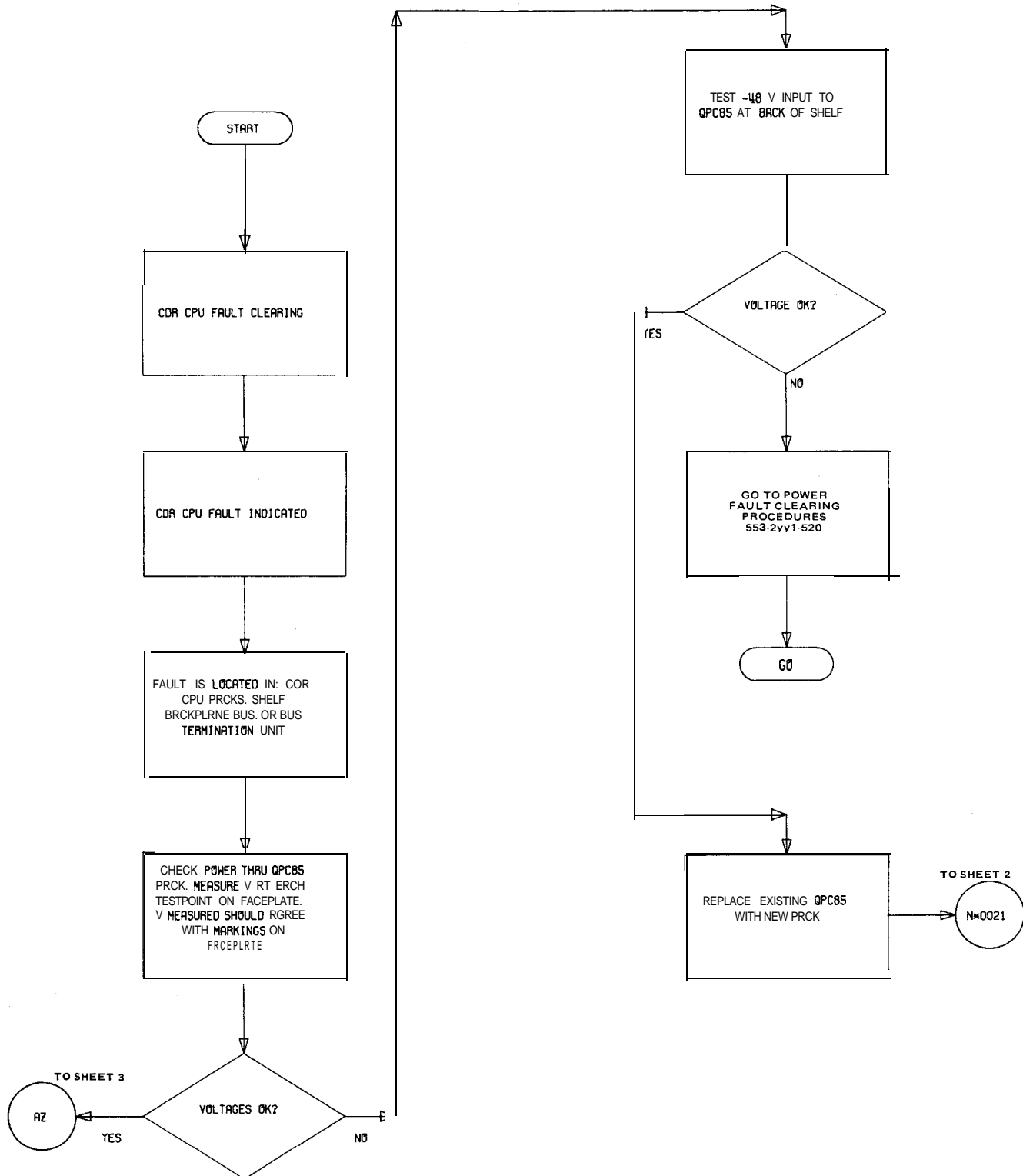
Flowchart 5-1 Continued
CDR Fault-Classification Procedures



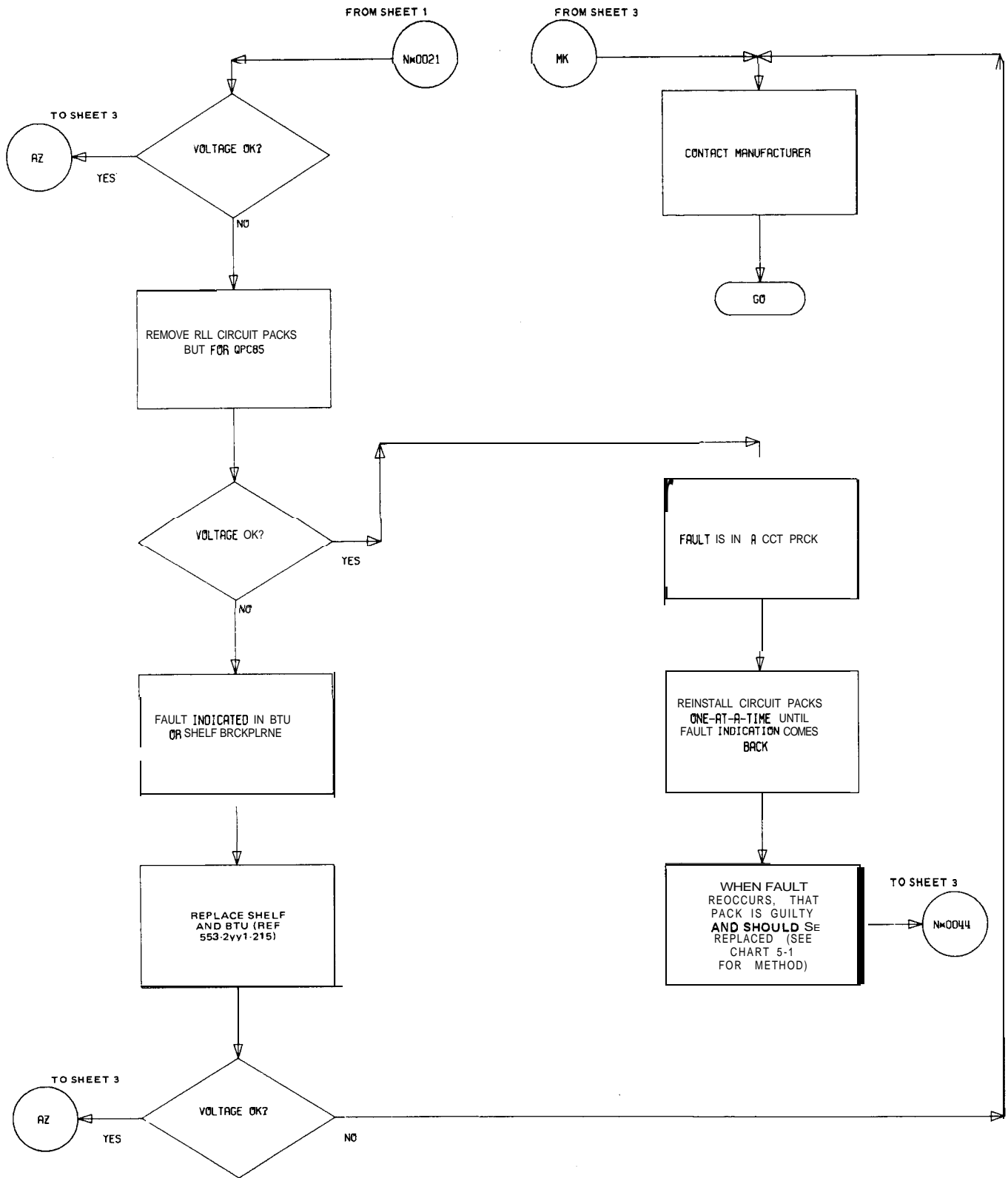
Flowchart 5-2
CDR Tape System Fault-Clearing Procedures



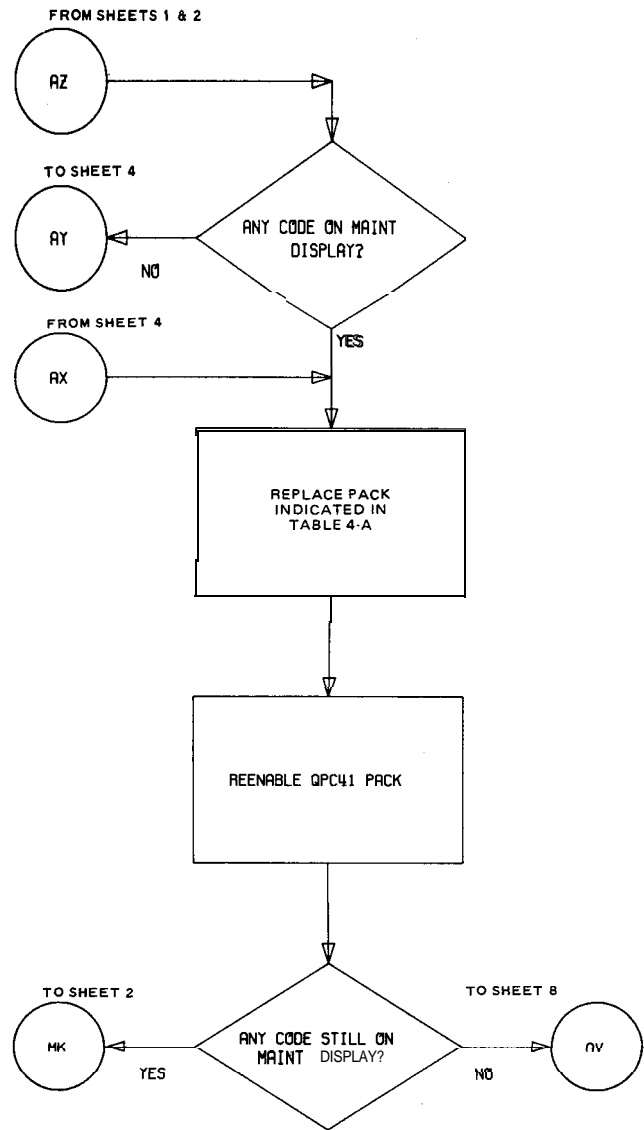
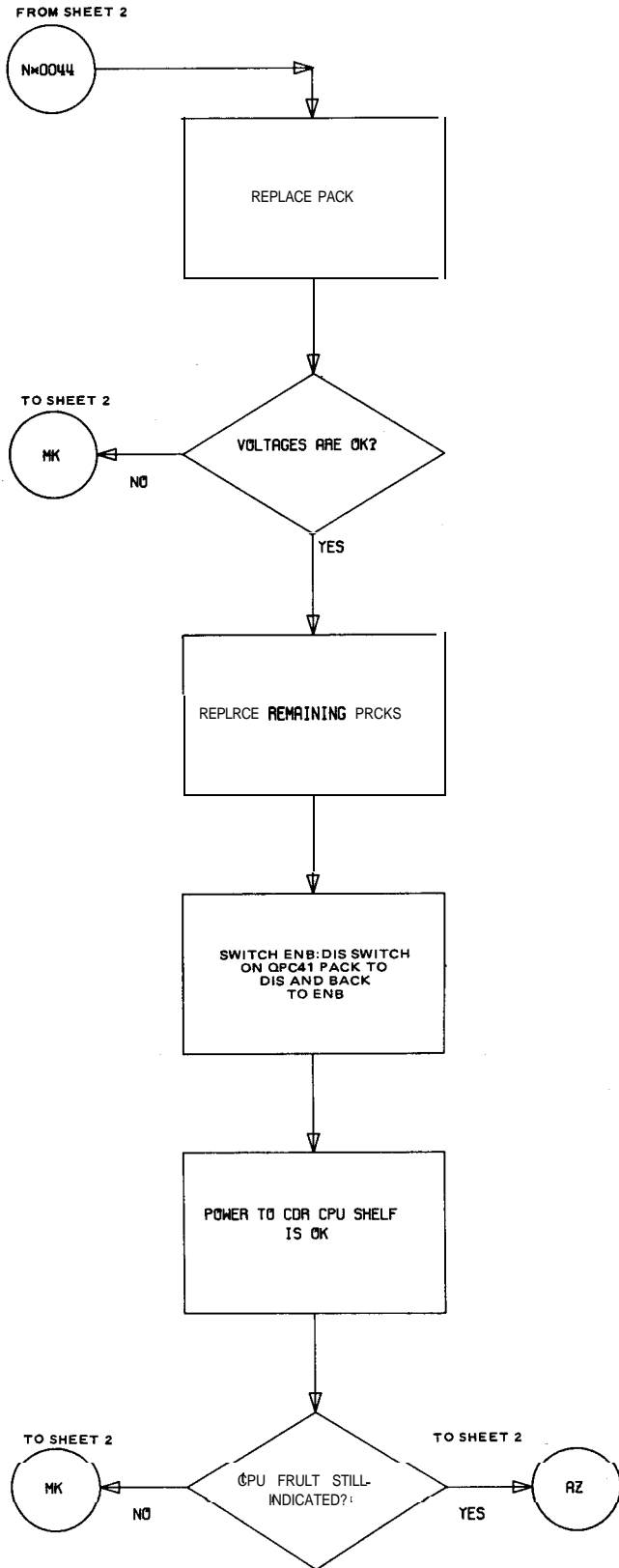
Flowchart 5-2 Continued
 CDR Tape System Fault-Clearing Procedures



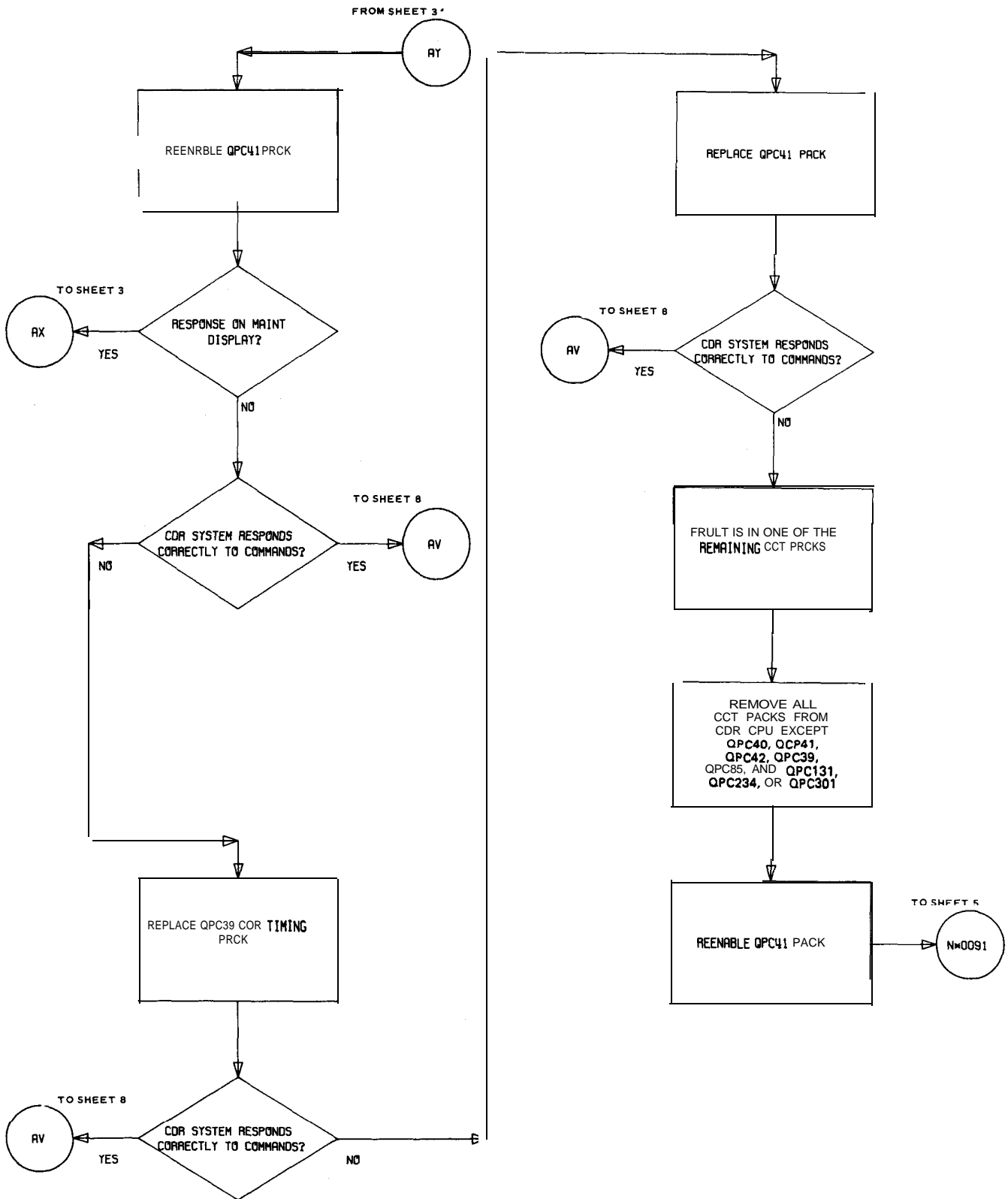
Flowchart 5-3
CDR CPU Fault Clearing Procedures



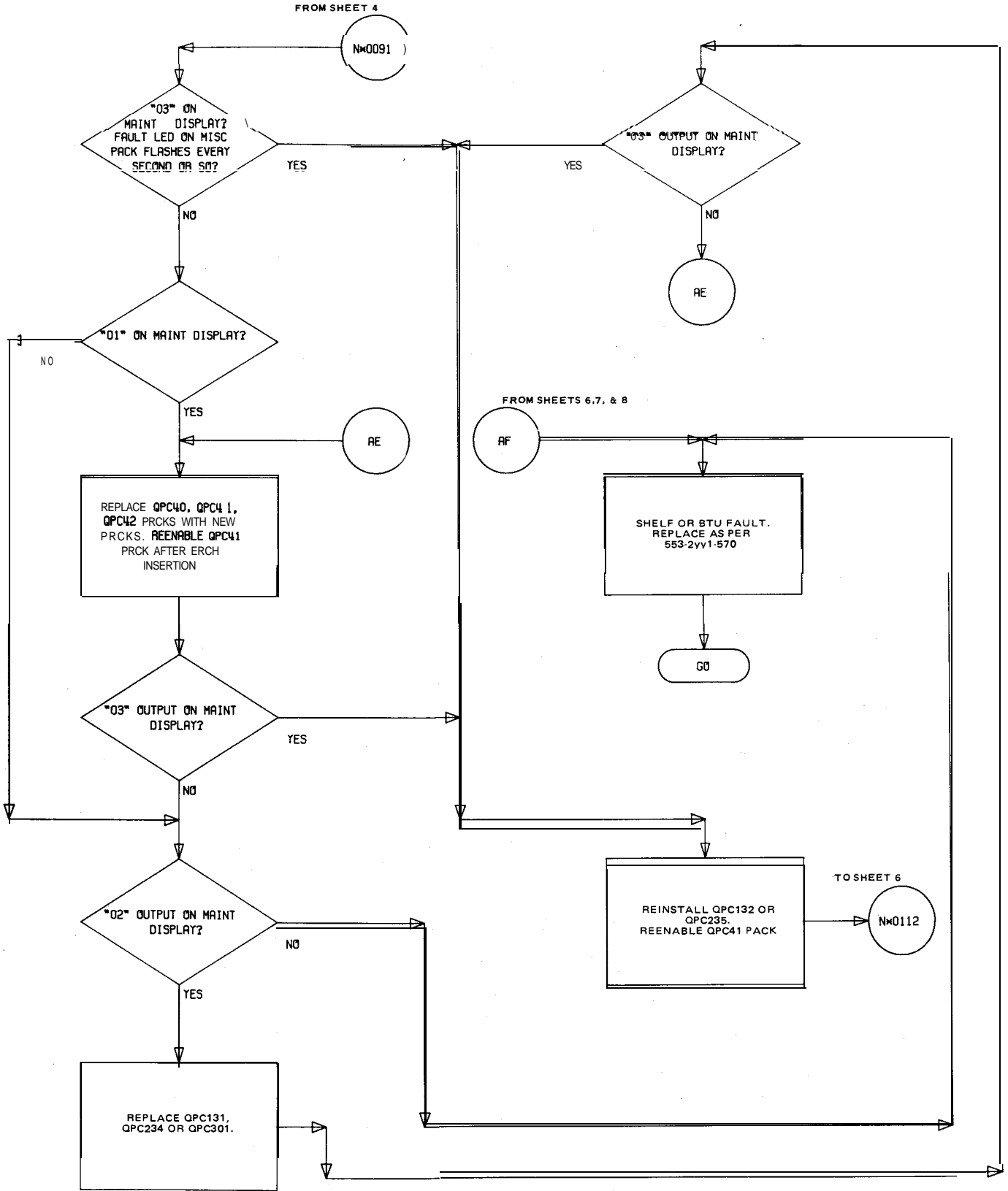
Flowchart 5-3 Continued
CDR CPU Fault Clearing Procedures



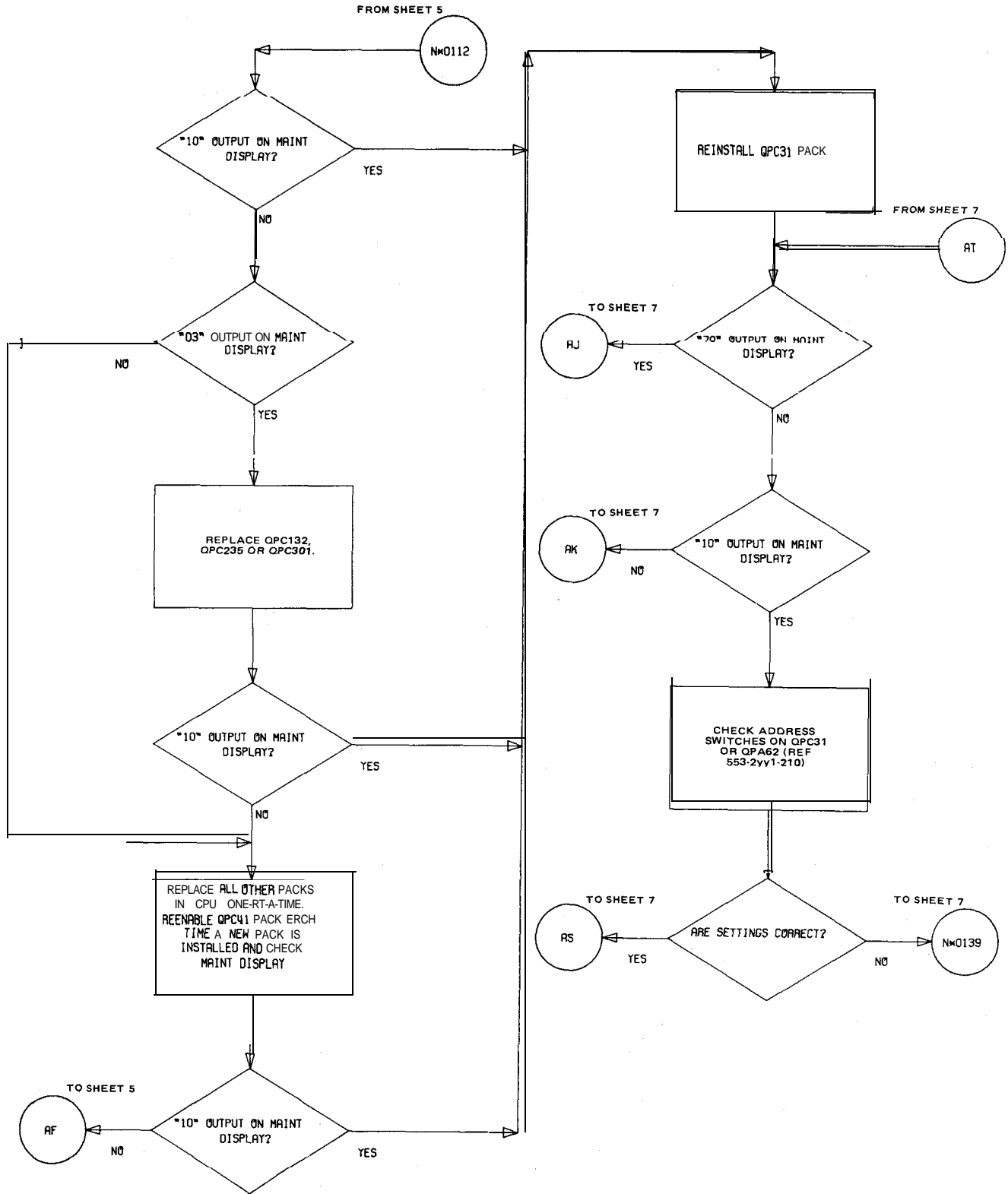
Flowchart 5-3 Continued
CDR CPU Fault Clearing Procedures



Flowchart 5-3 Continued
CDR CPU Fault Clearing Procedures

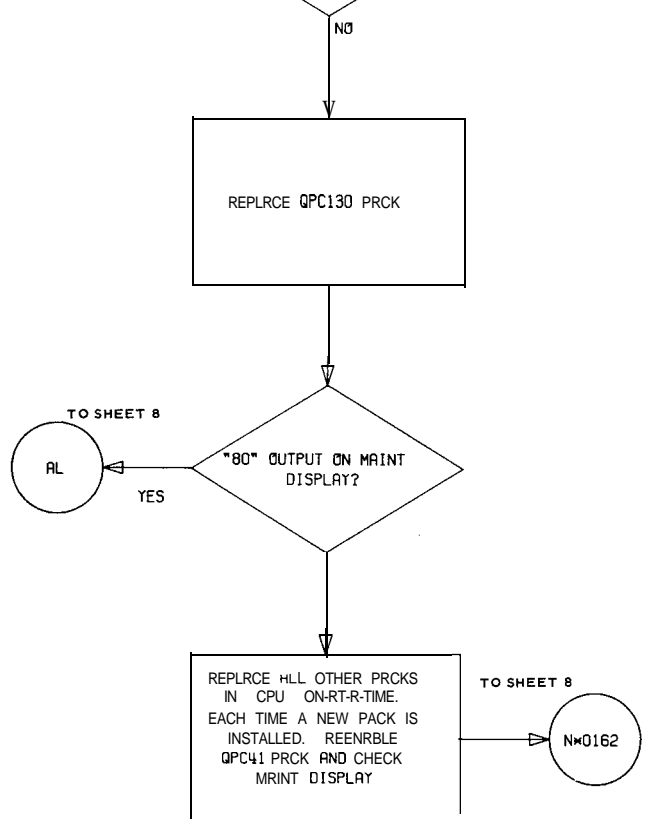
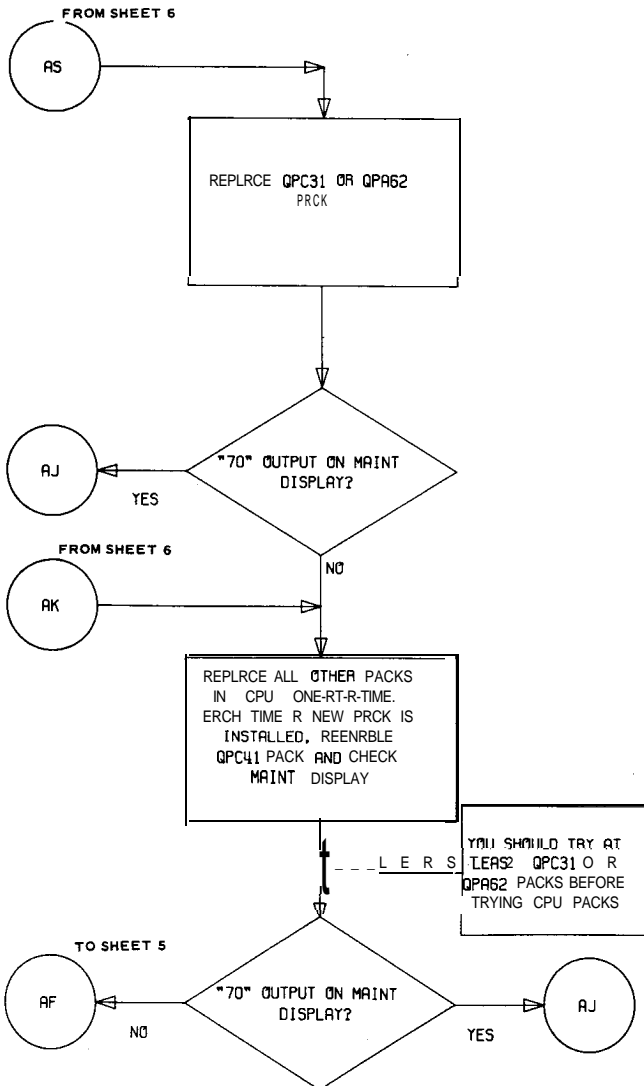
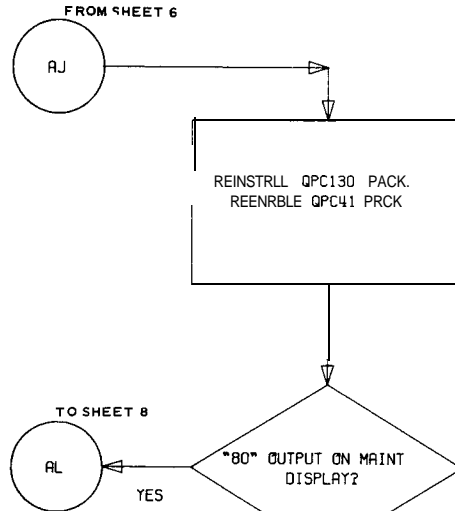
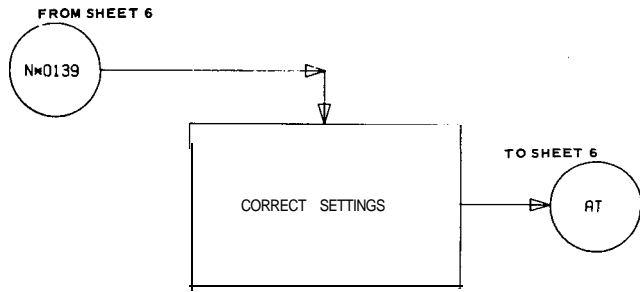


Flowchart 5-3 Continued
CDR CPU Fault Clearing Procedures

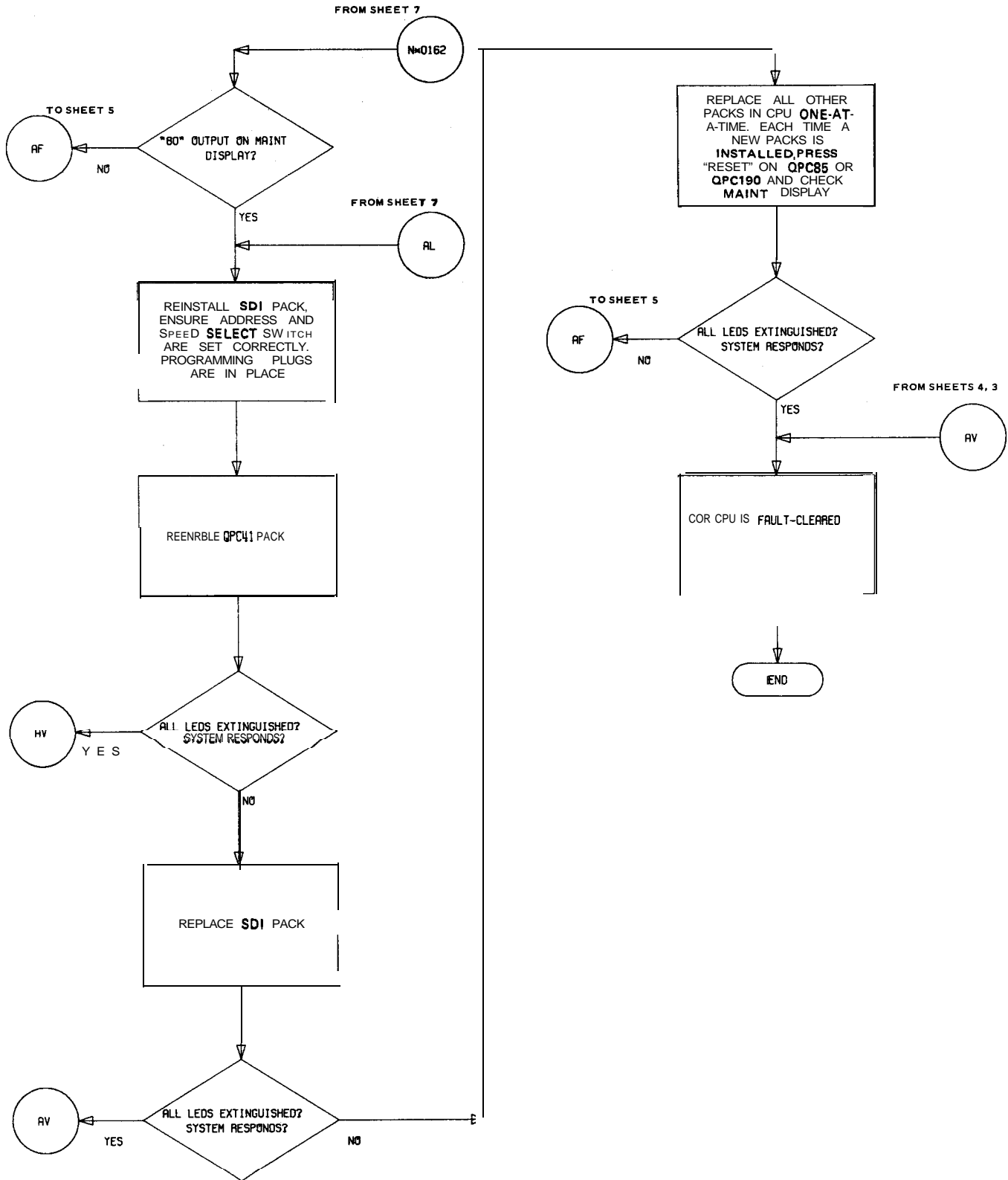


Flowchart 5-3 Continued
 CDR CPU Fault Clearing Procedures

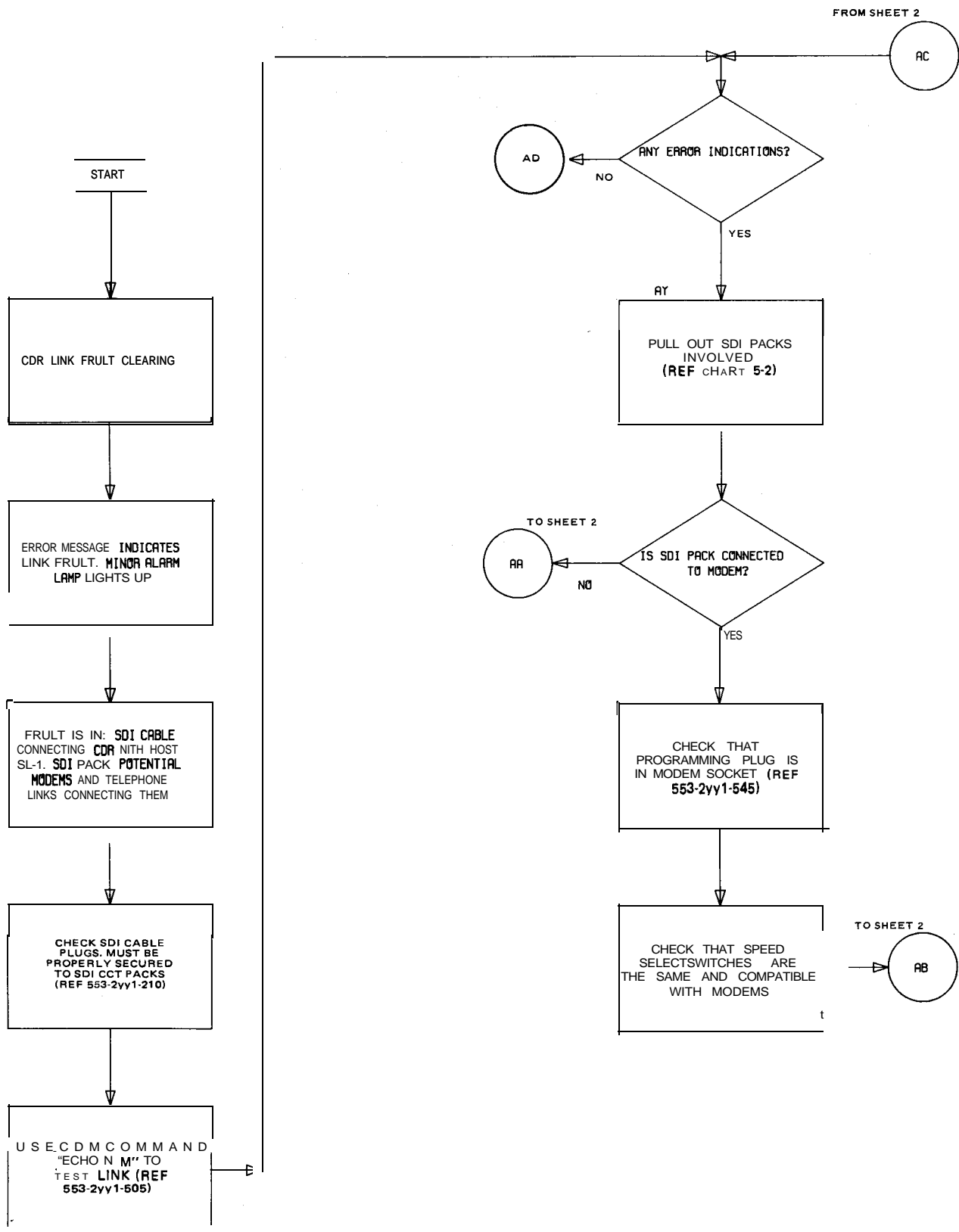
PRACTICE 553-2631-510



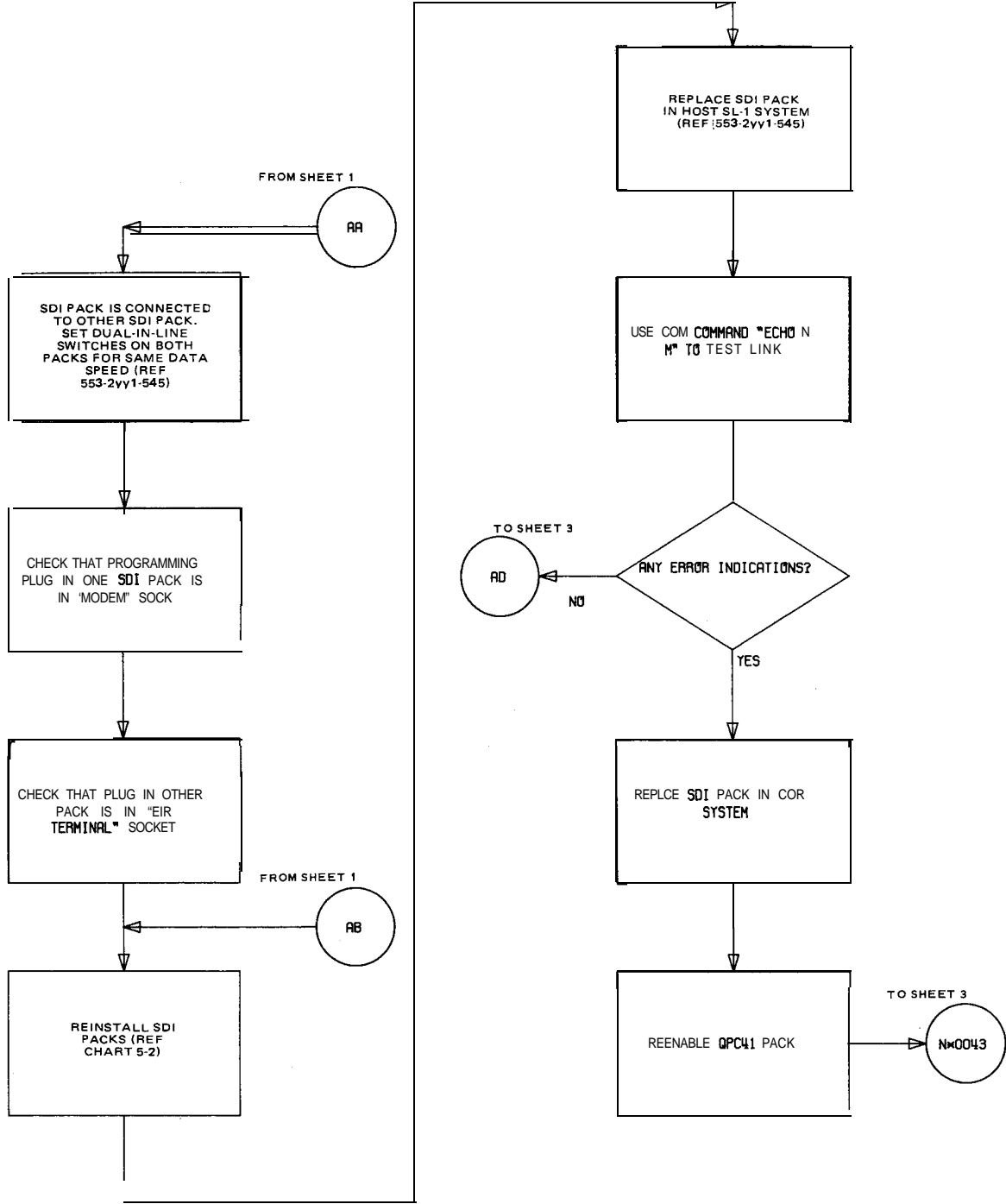
Flowchart 5-3 Continued
CDR CPU Fault Clearing Procedures



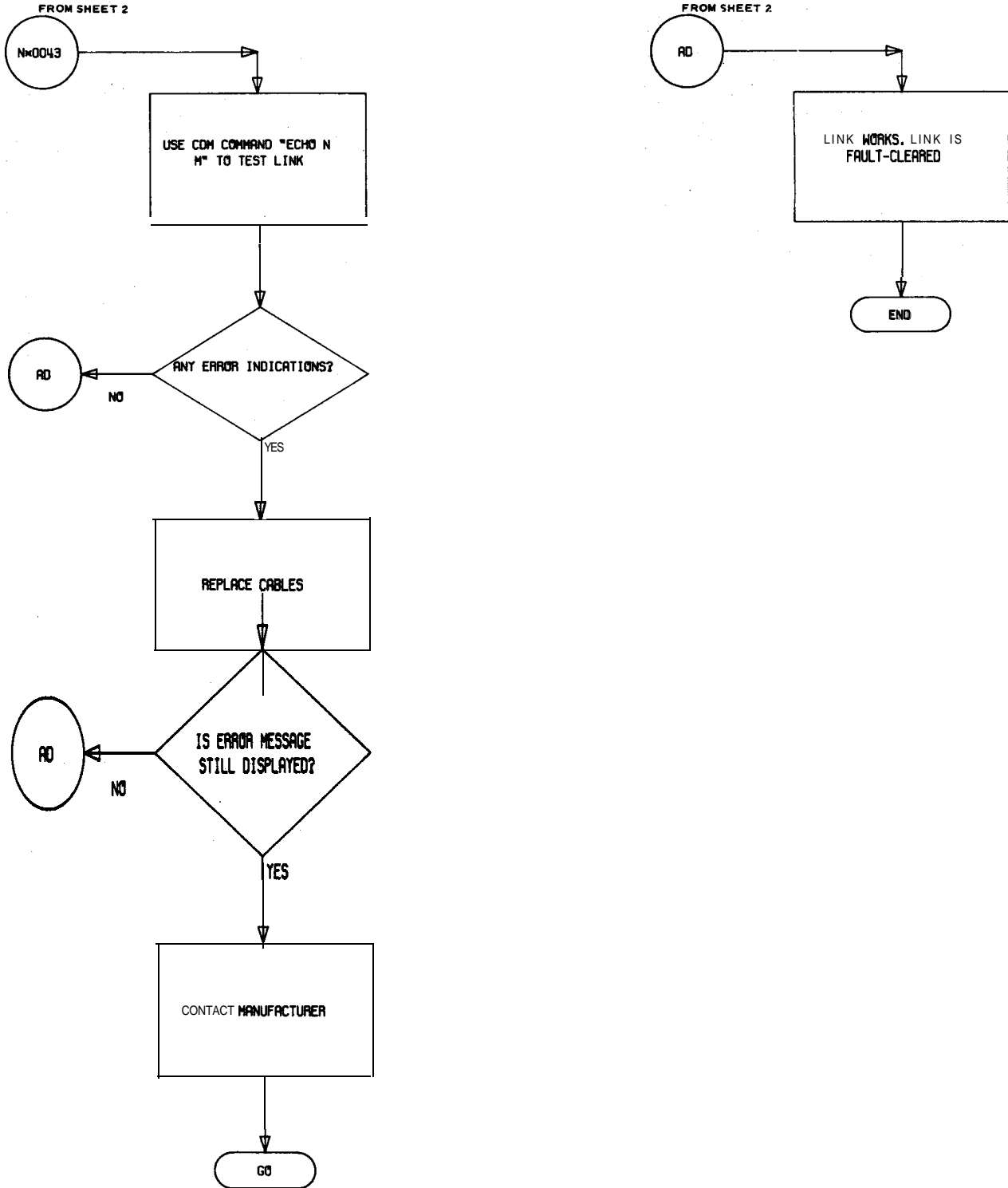
Flowchart 5-3 Continued
 CDR CPU Fault Clearing Procedures



Flowchart 5-4
CDR Link Fault Clearing Procedures



Flowchart 5-4 Continued
CDR Link Fault Clearing Procedures



Flowchart 5-4 Continued
CDR Link Fault Clearing Procedures

SL-1*
BUSINESS COMMUNICATIONS SYSTEM
AUTOMATIC ROUTE SELECTION

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* SL-1 is a trademark of Northern Telecom Limited

1. INTRODUCTION

1.01 Automatic Route Selection (ARS) is a collective term referring to several separate software packages available for the SL-1 system:

- | Main ARS
- | ARS Queuing
- | ARS Improvement for Autovon
- | ARS Traffic.

ARS improvement for Autovon is available only when the Autovon Generic (X14) is specified for the SL-1. ARS queuing is a prerequisite for ARS **traffic**. ARS **traffic** is described in 553-2001-450. **Traffic** Measurement. The main ARS feature is not a prerequisite for ARS queuing.

2. MAIN ARS FEATURE

GENERAL

2.01 The arrangement of telephone networks is often such that a call between two geographic locations can take one of several routes. The call can vary in cost according to which route is taken and the time of day at which the call is made. Thus, if a cost-conscious caller has more than one route to choose from, and wants his call to take the cheapest route when making a call at a given time, a calculation must first be made of the expense of each available route, the time of day, and all the various permutations that can result from these. This process can be irritating, if there are a lot of routes to choose from.

This exercise can be avoided by using an SL-1 system with the ARS option. An SL-1 system with the ARS feature can automatically select the cheapest route according to preprogrammed data concerning preferred routes and time-of-day. The user has only to dial a code to reach a given destination.

2.02 Instead of figuring out which one of several numbers to use in calling a certain location, the ARS user dials a code to access the ARS feature, and then another code to reach a certain location. The user then dials the local number to which he wishes to be connected. Tones, trunk routes to choose from, digits to be dialed, and any restrictions or limitations are preprogrammed into the three data blocks in the SL-1 memory.

2.03 Definition of some terms unique to the ARS feature:

- *ARS Access Code.* This is the number that is dialed by the telephone user to gain access to the ARS feature.
- *User Access Code.* This is the user-dialed code which may or may not be part of a telephone number that determines where a user's telephone call will be directed. The most appropriate means of reaching the destination is chosen by the SL-1 acting as directed by the ARS program.
- *Deletion and Insertion of Digits.* Each trunk route generally has a different dialing plan from other trunk routes. To make the single user access code compatible with all trunk routes, the SL-1 must sometimes delete digits from those dialed by the user, and replace them with other digits. The number of digits deleted and inserted are programmed by the user. The craftsman specifies how many

digits are deleted from the beginning of the user access code, and programs which digits will then be added to the front of the number.

- *Routes.* A trunk route is a group of trunks of the same type that go from a common location to a common location, as defined in the Route Data Block (see 553-2001-220/221).

FEATURE ORGANIZATION

2.04 The ARS feature consists of a computer program. No additional trunks or hardware circuits are required. Once programmed, the feature operates automatically under the control of the central processing unit in the system.

2.05 The ARS feature provides the customer with an extra overlay program to use for programming the system. This is Overlay 27 and is found only on SL-1 machines that are equipped with the ARS feature. The overlay program uses three types of data blocks:

- 1 ARS Data Block (ADB)
- ARS Schedule Blocks (SCB)
- User Access Code Block (UAC).

2.06 *ARS Data Block.* The ARS data block is a part of the SL-1 software that contains information about the customer's ARS options and general configuration. One ARS data block is created per customer. The ARS data block contains the following information :

- (a) the ARS access code (1 to 4 digits)
- (b) the maximum allowable number of:
 - schedule blocks (0 to 225)
 - user ARS codes (1 to 8000)
 - 1 maximum number of digits to be inserted for any user ARS code (1 to 11)
- (c) time of delay (0 to 32 seconds) before selecting a last choice route
- (d) whether or not dial tone is to be supplied after the ARS access code is dialed
- (e) whether or not a triple burst of delay tone is to be supplied.

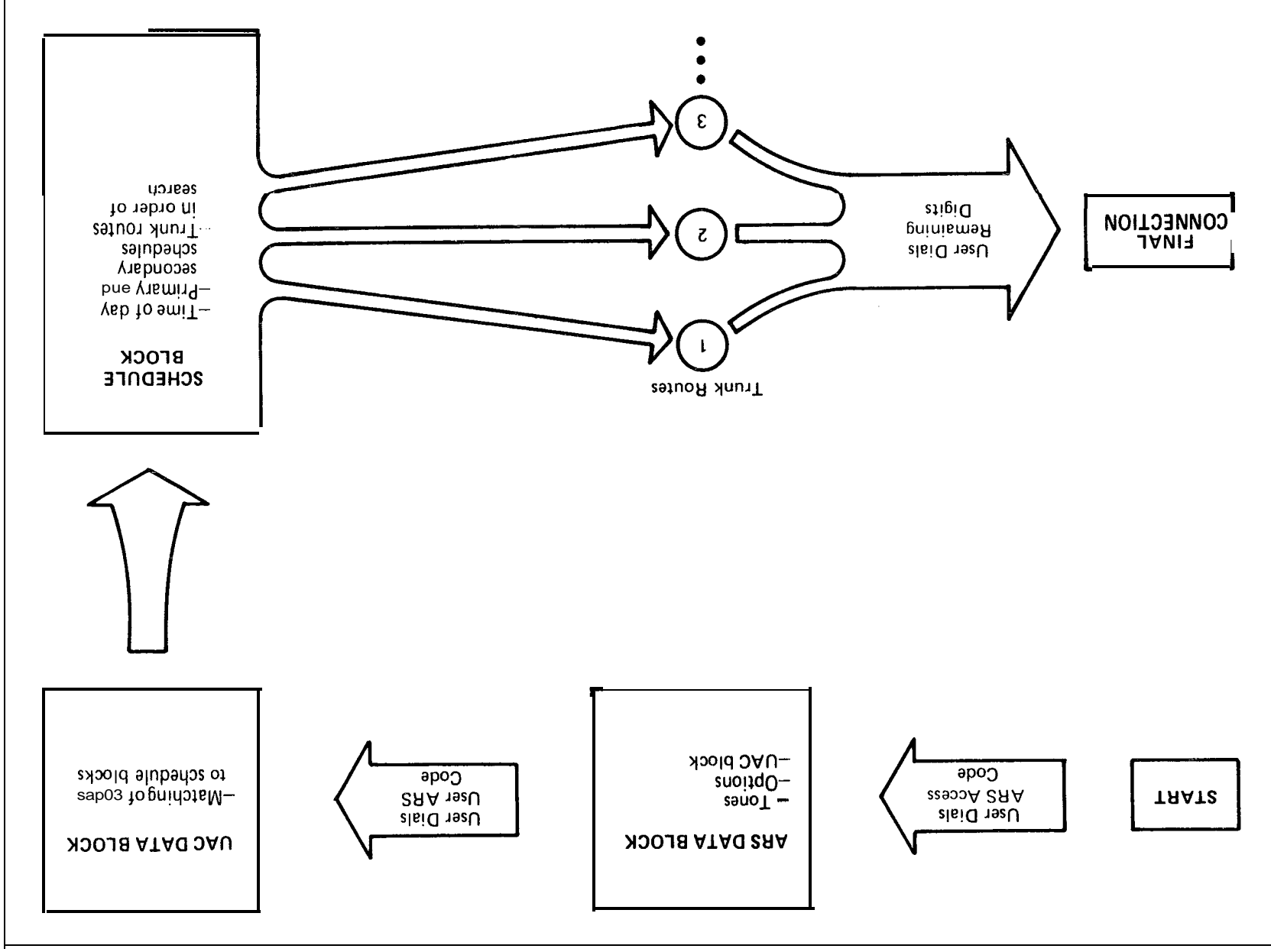


Fig. 2-1 – Block Diagram of ARS Feature Operation

2.07 *ARS Schedule Blocks.* The ARS schedule blocks contain information on how to route calls. Each schedule block contains a primary schedule, with an optional secondary schedule. A primary schedule is specified by:

- the hours during which the primary route sequence is in effect
- the permissible routes in their order of preference.

If the primary schedule is less than 24 hours long, the remaining hours make up the secondary schedule, and the routes for the secondary schedule are listed in the order in which they are searched for an idle trunk. Note that if a schedule only has one route, the triple burst of delay tone is never given, even if specified in the ARS data block.

One schedule block is reserved for the “universal call routing”, a route used for all user ARS codes not listed in the user ARS code data block. This routing is typically the local CO trunk group or other Direct Distance Dialing (DDD) facility and is assigned to schedule block 0 by the system (although the contents of schedule block 0 must be programmed by the customer).

The schedule block also contains the information for deletion and insertion of digits for each route specified in the primary and/or secondary schedules.

2.08 *User Access Code Data Block.* This data block is where the user access code dialed by the user is associated with schedule block. Only one schedule block may be associated with each user access code, but more than one user access code may be associated with the same schedule block.

WOW ARS WORKS

2.09 A telephone user in the SL-1 system picks up the telephone handset and dials the ARS access code. The system reads the ARS data block to determine which tones and delay values are in effect for this call. At this point the system provides dial tone if specified in the ARS data block. When the user dials the user access code, the system reads the user access code data block (for that customer) to determine which schedule block is to be used to put the call through.

2.10 Once the system has accessed the appropriate schedule block, the SL-1 decides whether the primary or secondary schedule is in effect by using the SE-1 internal clock and the programmed times in the schedule block. If the call is made within the primary schedule hours, the prime schedule routes listed in the data block are searched (in the order in which they are listed) for an idle trunk, except for the last route on the list. If there is only one route in the schedule, then an idle trunk on this route is sought immediately. If an idle trunk is not found, the user hears a triple burst of delay tone (if programmed in the ARS data block), and the system rescans all routes (including the last one) at the end of the delay period. If no idle trunks are found, the system returns an overflow tone to the user.

2.11 If the user activates “ring-again” during the delay period, the system continues to search all routes (except the last choice) until an idle trunk is found. If “ring-again” is activated after overflow tone has been received, all routes (including the last) are searched until an idle trunk is found.

2.12 The system does not proceed from primary to secondary schedules while searching for a route. The secondary schedule is used only outside of primary schedule hours, and is searched in the same manner as the primary schedule.

2.13 When an idle trunk has been found, the SL-1 outpulses the digits dialed by the user after the ARS access code, noting any deletions and/or insertions according to the ARS schedule block (specified for each route of the schedule). The first digit(s) of the number are used up in accessing the trunk. The SL-1 waits until the trunk is seized, then outpulses the rest of the digits, pausing only if an asterisk (*) is programmed into the number.

ARS on SL-1 only has control of the first leg of a route. A blockage or other problem on a subsequent leg of a connection is beyond the control of the SL-1.

Access Restrictions

2.14 The basic SL-1 access restrictions also apply to ARS. In particular, the ARS mechanism selects an outgoing trunk group only if the class of service of the calling station permits such a connection. Access to the ARS feature however, is allowed from all stations and tie trunks independently of their class of service. To provide a tool for controlling telephone costs a special class of service, Conditionally Toll Denied (CTD) is available. Depending on how an outgoing call is dialed, this class is equivalent to:

- Unrestricted (UNR) for calls places through ARS
- 1 Toll denied (TLD) for calls dialed directly.

The result is that a set with CTD class of service may place toll calls (over CO, FX, WATS and DID/DOD trunks) through ARS but not by directly accessing the outgoing trunks.

A toll call is defined as any call that is placed over CO, FX, WATS or DID/DOD trunk and where the first digit outpulsed is 1 or 0 (trunk group is not code restricted), or for which the first three digits outpulsed (and not absorbed by the connecting CO) represent a code that is denied for this route (trunk group is code restricted).

HOW TO USE ARS

2.15 The telephone user can make use of the ARS feature by following the procedure given in Chart 2- 1.

CHART 2-1
HOW TO USE ARS

To make use of ARS the user should know the value of the time delay before scanning the last choice route for an idle trunk.

This feature can be used in conjunction with the “ring again” feature. Users may find it useful to be familiar with “ring again” before using the ARS feature. See 552-2001-105 for information on “ring again”.

STEP	ACTION	VERIFICATION
1	Pick up the handset.	Dial tone heard from receiver.
2	Dial the ARS access code.	Dial tone may or may not be heard.
3	Dial the user access code for the destination you wish to reach.	If dial tone* is heard, go to Step 6. If warning tone is heard, go to Step 4. If overflow tone is heard go to Step 5.
4	If warning tone is heard: Either continue waiting or, activate “ring again”.	If dial tone? is heard, go to Step 6.
5	If overflow tone is heard: Activate “ring again”.	If dial tone† is heard, go to Step 6.
6	If dial tone is heard: Dial CO or extension number as required.	Ringback tone is heard, 2-way conversation when call is answered.

* The dial tone is optional, programmed in the ARS data block.

† This dial tone, if it is heard, comes from the switching machine at the far end of the trunk.

ENGINEERING ARS

2.16 *Memory and Real Time Impact.* Information regarding memory and real-time is contained in 553-2YY1-151.

2.17 *Trunk Arrangement and Selection.* The actual programming of trunk route selection and digit insertion and deletion can best be illustrated using an example. Assume the customer is a hypothetical company with branches in three cities, Northeast, Southeast (where the SL-1 is), and West. The company has tie trunks and WATS trunks as well as access to the **DDD** network. Tie calls are cheaper than WATS calls which in turn are cheaper than DDD calls, except after 6 p.m., when **DDD** calls using public telephone network are cheapest of all.

The western and eastern cities are in different time zones, the east being one **hour** ahead of the west. In this example the telephone company rates take a sharp increase at 8 a.m. Thus when a call goes through from the southeast branch to the northeast branch between 8 a.m., and 9 a.m., it **would** be

cheaper to go by way of the western branch rather than the more direct route. This is true not only for calls to the northeast branch but for any **long-distance** call before 9 a.m.

Dialing Plan

2.18 In this example the trunk arrangement for the southeast branch of the company is as follows (see Fig. 2-2).

The southeast SL-1 is equipped with the following trunk routes:

ROUTE NUMBER	DESTINATION AND TYPE
1	Tie to northeast branch
2	WATS to northeast branch
3	Tie to western branch
4	WATS to west branch
5	CO trunk; i.e., to public telephone network.

TABLE 2-A
TRUNK ARRANGEMENT OF SOUTHEAST BRANCH

DESTINATION	HOURS OF THE DAY	ROUTE NUMBERS IN ORDER OF PREFERENCE
Northeast (company)	8 a.m. to 9 a.m. 9 a.m. to 8 a.m.	3, 1, 2, 5 1, 2, 5
Northeast (local area)	8 a.m. to 9 a.m. 9 a.m. to 8 a.m.	3, 1, 2, 5 1, 2, 5
West (company)	8 a.m. to 6 p.m. 6 p.m. to 8 a.m.	3, 4, 5 5, 3, 4
West (local area)	8 a.m. to 6 p.m. 6 p.m. to 8 a.m.	3, 4, 5 5, 3, 4
West (long distance)	8 a.m. to 6 p.m. 6 p.m. to 8 a.m.	3, 4, 5 5, 3, 4

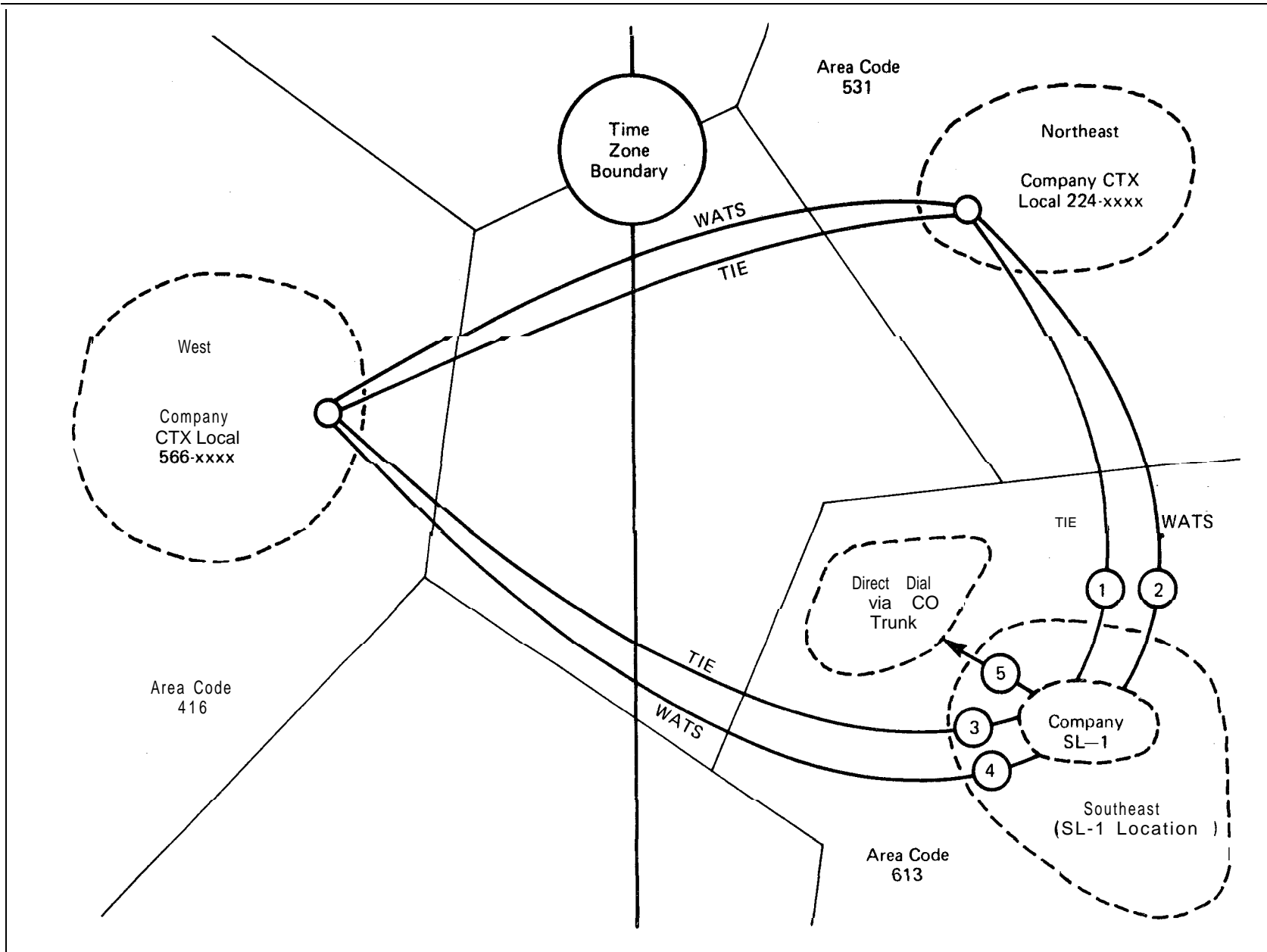


Fig. 2-2 - Geographic Layout of Hypothetical Company

2.19 Programming the SL- 1 for ARS is given in Chart 2-2.

CHART 2-2
IMPLEMENTING THE ARS FEATURE

STEP	PROCEDURE												
1	List all destinations for which ARS is to be used. For the example, see Table A.												
2	For each destination list all the alternate trunk routes used to access the destination in ascending order of cost. If the costs change according to time, list the routes in order of preference under each time period (see Table A for example).												
3	Decide on an ARS access code, for example 7.												
4	For each destination, assign a user access code. For example:												
	<table style="margin-left: auto; margin-right: auto;"> <thead> <tr> <th style="text-align: left;">DESTINATION</th> <th style="text-align: left;">USER ARS CODE</th> </tr> </thead> <tbody> <tr> <td>Northeast</td> <td>- company premises 177</td> </tr> <tr> <td></td> <td>- local calling area 177-9</td> </tr> <tr> <td>West</td> <td>- company premises 144</td> </tr> <tr> <td></td> <td>- local calling area 144-9</td> </tr> <tr> <td></td> <td>- long distance 144-9-1</td> </tr> </tbody> </table>	DESTINATION	USER ARS CODE	Northeast	- company premises 177		- local calling area 177-9	West	- company premises 144		- local calling area 144-9		- long distance 144-9-1
DESTINATION	USER ARS CODE												
Northeast	- company premises 177												
	- local calling area 177-9												
West	- company premises 144												
	- local calling area 144-9												
	- long distance 144-9-1												
5	For each trunk route, determine the digits to be deleted from and/or inserted into the dialed number to allow a call to be completed on the route. For the example, these are given in Table E1. Note that the access code for the desired trunk route must be incorporated into these deletions and insertions.												
6	The following restrictions and limitations must be kept in mind when programming ARS: <ul style="list-style-type: none"> ● Each customer can have no more than 255 different schedule blocks. In the example given there are six. ● Each schedule block has a primary and secondary schedule. Hours of operation are specified for the primary schedule, and the hours outside these are assumed to be the secondary schedule. ● For any schedule, primary or secondary, no more than eight routes may be specified. ● In any route a maximum of 15 digits may be deleted, and a maximum of 11 digits may be inserted. ● Before programming the call codes, it is first necessary to specify the maximum expected number of call codes so that memory for the customer's ARS program can be allocated in one continuous block. When the number of call codes increases above the specified maximum, a system reload may be necessary to reallocate memory. ● The ARS access code must not be the same as the beginning of any other number or access code for the customer. 												

TABLE 2-B
DIALING PLAN FOR SOUTHEAST ARS INSTALLATION

DESTINATION	ROUTE	DIGITS DIALED	DIGITS TO BE OUTPULSED BY THE SYSTEM	DIGITS DELETED	DIGITS INSERTED
Northeast (Company Centrex Local)	tie	177-XXXX	x x x x	0	none
	WATS	177-XXXX	155-53 1-224-XXXX	3	155-53 1-224
	DDD	177-XXXX	9-1-53 1-224-XXXX	3	9-1-531-224
	tie (via West)	177-XXXX	144-177-XXXX	3	144-177-
Northeast (Local Calling Area)	tie	177-9-NNX-XXXX	9-NNX-XXXX	0	none
	WATS DDD	177-9-NNX-XXXX	9-1-5155-53 1-NNX-XXXX 1-NNX-XXXX	11	9-1-531-155-53
	tie (via West)	177-9-NNX-XXXX	144-177-9-NNX-XXXX	3	144-177-9-
West (Company Centrex Local)	tie	144-XXXX	x x x x	0	none
	WATS	144-XXXX	155-416-566-XXXX	3	155-416-566
	DDD	144-x x x x	9-1-416-566-XXXX	3	9-1-416-566
West (Local Calling Area)	tie	144-9-NNX-XXXX	9-NNX-XXXX	0	none
	WATS DDD	144-9-NNX-XXXX	9-1-416-NNX-XXXX	11	155-416-
West (Long Distance)	tie	144-9-1-AAA-NNX-XXXX	9-1-AAA-NNXC-XXXX	0	none
	WATS	144-9-1-AAA-NNX-XXXX	155-416-9-1-AAA-NNX-XXXX	3	155-416-9-1-
	DDD	144-9-1-AAA-NNX-XXXX	9-1-AAA-NNX-XXXX	3	none

Programming to Save Real Time

2.20 The amount of real time the ARS feature uses depends mainly on the length of time the computer must spend searching through entries in the user access code data block each time a user access code is dialed. Arranging the programming to reduce the number of entries in the user access code data block therefore saves real time in making each ARS Call.

2.21 The next step is to program the system. First, overlay 27 must be loaded using the "LD 27" command. Once overlay 27 is loaded, the prompts and responses in Chart 2-3 are used to program the hypothetical system (see 553-2001-200/221) for a detailed description of the prompts and responses.

CHART 2-3
ARS DATA INPUT

PROMPT	RESPONSE	COMMENTS
REQ TYPE CUST ARDN MAXS MAXC MAXL	new adb 0 7 10 10 9	First, the ARE data block is created. ARS access code. Maximum schedule blocks required. Maximum user access codes required. Maximum digits to be inserted. These "maximum values" are designed to save protected data store space, while still allowing for some expansion.
DLTN DELY DYTN	yes 3 yes	Dial tone. Delay before choosing last route. Delay tone.
REQ TYPE CUST SCBN PSCH PSRT SSRT RT DEL INST	new scb 0 1 8 to 9 3 1 2 5 1 2 5 1 0 (carriage return)	First schedule block (for Northeast, Company local) Schedule block numbers 0 is reserved for the universal call routing. Routes 1,2,3,5 are programmed for order of search. Routes 1,2,5 are programmed for digits deleted and inserted.
RT DEL INST RT DEL INST RT DEL INST RT	2 3 155531224 3 3 144177 3 91531224 (carriage return)	

Chart Continued

CHART 2-3 Continued
ARS DATA INPUT

PROMPT	RESPONSE	COMMENTS	
REQ	new	Second schedule block (for Northeast, local calling area).	
TYPE	scb		
CUST	0		
SCBN	2		
PSCH	8 to 9		
PSRT	3 1 2 5		
SSRT	1 2 5		
RT	1		
DEL	0		
INST	(carriage return)		
RT	2		
DEL	3		
INST	155531		
RT	3		
DEL	3		
INST	1441779*	** causes a 3-second pause while a CO trunk is seized and dial tone returned.	
RT	5		
DEL	3		
INST	91531		
RT	(carriage return)		
REQ	new		Third schedule block (for West, Company local).
TYPE	scb		
CUST	0		
SCBN	3		
PSCH	8 to 18		
PSRT	3 4 5		
SSRT	5 3 4		
RT	3		
DEL	0		
INST	(carriage return)		
RT	4		
DEL	3		
INST	155416566		
RT	5		
DEL	3		
INST	91416566		
RT	(carriage return)		

Chart Continued

CHART 2-3 Continued
ARS DATA INPUT

PROMPT	RESPONSE	COMMENTS
REQ	new	Fourth schedule block (for West, local calling area).
TYPE	scb	
CUST	0	
SCBN	4	
PSCH	8to 18	
PSRT	3 4 5	
SSRT	5 4 3	
RT	3	
DEL	0	
INST	(carriage return)	
RT	4	
DEL	3	
INST	155416	
RT	5	
DEL	3	
INST	91416	
RT	(carriage return)	
REQ	new	Fifth schedule block (for West, long distance).
TYPE	scb	
CUST	0	
SCBN	5	
PSCH	8to 18	
PSRT	3 4 5	
SSRT	5 3 4	
RT	3	
DEL	0	
INST	(carriage return)	
RT	4	
DEL	3	
INST	1554169*1	
RT	5	
DEL	3	
INST	(carriage return)	
RT	(carriage return)	
REQ	new	Schedule block for universal call routing.
TYPE	scb	
CUST	0	
SCBN	0	
PSCH	0 to 0	
PSRT	5	
RT	5	
DEL	0	
INST	(carriage return)	
RT	(carriage return)	

Chart Continued

CHART 2-3 Continued
ARS DATA INPUT

PROMPT	RESPONSE	COMMENTS
REQ	new	Create a UAC block to associate one schedule block with each user ARS code dialed. Different codes may reference the same scb , but one code cannot have more than one scb .
TYPE	uac	
CUST	0	
UACN	177	
SCBN	2	
UACN	1770	
SCBN	3	
UACN	144	
SCBN	4	
UACN	1449	
SCBN	5	
UACN	14491	
SCBN	6	
UACN	(carriage return)	
REQ	end	The ARS programming is now complete.

To remove a number from the PSCH (prime schedule hours) type in an 'X' **preceeding** the number; e.g., typing:

X0 X1 X2 for PSCH removes 0, 1, and 2

X0 to X8 for PSCH removes 0 1 2 3 4 5 6 and 7

XI7 to **X0** for PSCH removes 17 18 19 20 21 22 and 23.

3. ARS QUEUING

GENERAL

3.01 'ARS queuing' (ARSQ) is designed to modify the ARS feature package and 'ring again' feature to allow customer control of:

- user's waiting time before being **allowed** to use the last choice route (ARS calls only),
- user's priority of access to 'ring again over trunks' (all trunk calls).

3.02 In ARSQ one group of prompts serves to modify the main ARS feature, and another **modifier** 'ring again over trunks'. ARSQ can therefore be used to modify 'ring again' whether ARS is present or not.

3.03 **ARSQ is a software feature requiring no additional hardware equipment.** Once programmed, the feature operates under system control. The user uses the feature in the same way as regular ARS and 'ring again'.

FEATURE OPERATION

3.04 The telephone user places a **call** using the ARS feature, but without ARSQ. If all available ARS routes are busy, the user receives a warning in the form of three bursts of miscellaneous tone.

The user can then:

- activate 'ring again' indefinitely over cheap routes, or
- abandon the call in T seconds, or
- 1 wait T seconds, after which time ARS will search all routes (including the most expensive). If all the routes are busy, **overflow** tone is returned and the user can either abandon the call or activate ring again over all routes.

When the ARSQ feature package is present, some other options are available to the customer.

ARS Restricted

3.05 When the ARS restricted option is specified, the most expensive route never becomes **available** for use. The user receives overflow tone, if the cheap routes are busy, and must either activate 'ring again' over cheap routes or else terminate the call. If the user does neither, the station will be locked out after 15 seconds.

Expensive Route Wait Timer (ERWT)

3.06 The ERWT determines how long the user must wait before using the last-choice route. If the user invokes 'ring again' over cheap routes, the ERWT begins to run, unless it has an infinite value (default condition). If a cheap route trunk becomes idle while the time is running, 'ring again' offers the call to the user. If no cheap routes become idle and the timer expires, one of two things can happen:

- If the least desirable route has an idle trunk, 'ring again' offers the **call** to the user.
- If no routes are available, 'ring again' is automatically activated on **all** routes, and the ERWT is not started again.

'Ring Again' Modification

3.07 'Ring Again' over trunks is also modified by ARSQ. In the normal operation of 'ring again', several users activating 'ring again' over the same routes are queued on a first-in, first-served basis. ARSQ allows modification of this rule.

When ARSQ is in effect, each call entering the 'ring again' queue is assigned a starting priority. Calls of a higher starting priority are immediately put ahead of calls with a lower starting priority. At specified intervals (promotion wait time), the call is promoted to the next higher priority and moves up in the queue movement. This continues until the call reaches its maximum priority. After this has happened the system continues to compute the priorities and normal queue movement to determine when the call is to be connected to an idle trunk.

SERVICE PROVISION AND CHANGE

3.08 Each station has associated with it an ARSQ code between 0 and 3, assigned in the **500** data block (**500/2500** sets) or the SL-1 data block (SL-1 sets). This ARSQ code determines the treatment that the set is to have when using the ARS or 'ring again' features or both.

3.09 It is in the customer data block (overlay 15) where each ARSQ code is associated with the desired parameters of **call** treatment.

3.10 For detailed description of prompts and responses, see **553-2001-200** and **553-2001-221**.

SECTION 553-265 1-200

MEMORY REQUIREMENTS

3.11 The protected data store requirement for each customer is eight words. Two bits in each station block are also used. These costs exist whether the customer has the ARSQ package, or not.

EXAMPLE

3.12 Assume that the hypothetical system in Part 2 has the ARSQ feature for four telephones in the system. The telephones have ARSQ codes 0, 1, 2, and 3 programmed into their data blocks opposite the ARSQ prompt in each block. In the customer data block the programming would take the form shown in Table 3-B (see 553-2001-221 for more details).

3.13 The telephone with ARSQ code 0 (and all other telephones with ARSQ code 0) now have the following characteristics:

- If the telephone user activates “ring again” he or she starts out at the priority level ahead of priority 0 calls entering the queue and behind priority 2 and 3 calls. If the call is not put through within 60 seconds of activating ‘ring again’, the telephone is advanced to the **next-to-top** priority for being connected to an idle trunk.
- If the call is not answered within 90 seconds of activating ‘ring again’, the last-choice route is scanned for an idle trunk.

TABLE 3-A
ARSQ ‘RING AGAIN’ PROMPTS

PROMPT	DESCRIPTION	RANGE	DEFAULT
SPRI	starting priority	0 to 3	0
MPRI	maximum priority	SPRI to 3	0
PROM	promotion wait time	1 to 999 seconds seconds intervals of 30 seconds	3

Note that PROM and ERWT are independant of each other.

TABLE 3-B
CODE 0 DATA INPUT

PROMPT	RESPONSE	COMMENTS
ARSQ	0	ARSQ code “0” to be programmed
ARSR	no	ARS restricted class of service
SPRI	1	starting priority
MPRI	2	maximum priority
PROM	60	promotion wait time
ERWT	90	expensive route wait time

3.14 Code 1 is programmed as shown in Table 3-C.

3.15 The telephones with **ARSQ** code 1 have the following characteristics:

- The last choice route can never be used for any call.

- The telephones with **ARSQ** code 1 are always at the back of the “ring again” queue, and are allowed access to an idle trunk only when no other telephones are in front of them.

Codes 2 and 3 are programmed in a similar manner.

TABLE 3-C
CODE 1 DATA INPUT

PROMPTS	RESPONSE	COMMENTS
ARSQ	1	
ARSR	yes	
SPRI	0	
MPRI	0	

The PROM is not given, since **SPRI** = MPRI.

The ERWT prompt is not given, since ARSR is “yes”.

4. ARS IMPROVEMENT FOR AUTOVON
(Generic X 14)

GENERAL

4.01 The ARS feature improvement of Autovon incorporates the following options which are used in conjunction with those present in the main ARS feature.

- l ARS processing on all **calls** with no ARS access code necessary.
- Normal ARS processing or digit manipulation only (on selected sources).
- Flexible attendant Directory Number (DN).

RESTRICTIONS

4.02 The main ARS feature is a prerequisite for the ARS improvement for Autovon.

DESCRIPTION

All-ARS Option

4.03 An ALL-ARS option in the ARS data block indicates that there is no ARS access code for this customer. If the All-ARS option is not specified, then the ARS feature operates exactly as standard.

4.04 When the ALL-ARS option is specified, all calls not directed to a specific ARS schedule are automatically directed to schedule 0. When a new ARS data block is created with the ALL-ARS option set, the system automatically assigns a default schedule 0. This is a “digit manipulation only” schedule, which deletes 0 digits and inserts nothing.

Digit Manipulation

4.05 The sole purpose of a “digit manipulation only” schedule is to delete digits from those dialed by the user and replace them with other preprogrammed digits. A different deletion and replacement program can be specified for each sequence of digits dialed, by using different “digit manipulation only” schedules. No route selection takes place, and no primary or secondary schedule hours are input for a “digit manipulation only” schedule.

4.06 The Autovon-improved ARS feature uses two types of schedule block; ‘normal ARS’ and ‘digit manipulation only’. The prompt TRNK in the schedule block determines which kind of schedule block it will be.

- A ‘yes’ response to the TRNK prompt results in a ‘normal ARS’ schedule. In this type of schedule, calls that use the schedule must terminate on a trunk and are processed using the main ARS feature.
- A ‘no’ response to the TRNK prompt results in all calls using the schedule being processed by ‘digit manipulation’ only.

Flexible Attendant DN

4.07 This feature allows a DN taken from the normal numbering scheme of the system to be assigned as the attendant DN. A station calls the SL-1 attendant by dialing this number instead of ‘0’. The flexible attendant DN is stored in the protected customer data block and is specified via service change on that block (prompt ATDN) it defaults to ‘0’.

EXAMPLE OF AUTOVON – IMPROVED ARS

4.08 The following example deals with the situation in which it is desired to use the Autovon ARS improvement for digit manipulation only. For a complete list of prompts and error codes and their meanings see 553-2661-220 and -221.

4.09 In the hypothetical SL-1 installation, there are 3 trunk routes:

ROUTE NUMBER	ACCESS CODE
1	91
2	92
3	93

The desired call treatment is given in Table 4-A.

TABLE 4-A
CALL TREATMENT FOR SAMPLE INSTALLATION

DIGITS DIALED	ACTION REQUIRED
0	- outpulse 0 on route 3
118	- outpulse 118 on route 2
S X X X	- outpulse XXX on route 3
7 X X X	- intercom call
all others,	
1000 to 1179,	- outpulse XXXX on route 1
1190 to 1999	
2 X X X	
3 X X X	
4 X X X	
6XXX	
8XXX	
9XXX	

4.10 The proper service changes to result in this call treatment are given in Chart 4-1.

CHART 4-1

STEP	ACTION
1	<p>Load 27 and create an ARS data block (ADB, overlay overlay 27) with ALL-ARS (AARS) option set to 'yes'.</p> <pre> LD 27 ARS000 REQ NEW TYPE ADO CUST 2 AARS YES MAXS 10 MAXC 10 MAXL 11 DELY 20 DYTN YES </pre> <p>These maxima are designed to minimize protected data store use while allowing for some expansion.</p>
Chart Continued	

CHART 4-1 Continued

STEP	ACTION
------	--------

- 2 Create 5 ARS schedule blocks (SCB, overlay 27).

SCB NO.	TRNK?	DIGITS DELETED	INSERT DIGITS:
0	NO	0	91
1	NO	0	93
2	NO	0	92
3	NO	1	93
4	NO	0	0

Note: When specifying “digits to be deleted”, you are specifying *how many* digits are to be deleted from the beginning of the number. If nothing is to be deleted, enter “0”.

When specifying “digits to be inserted”, you are specifying *which* digits are to be inserted at the front of the number. If it is desired to insert no digits, press RETURN.

PROMPT	RESPONSE	COMMENTS
--------	----------	----------

REQ	NEW	
TYPE	SCB	
CUST	2	“Universal” SCB
SCBN	0	
TRNK	NO	
DEL	0	
INST	91	

REQ	NEW	
TYPE	SCB	
CUST	2	
SCBN	1	first SCB
TRNK	NO	
DEL	0	
INST	93	

REQ	NEW	
TYPE	SCB	
CUST	2	
SCBN	2	second SCB
TRNK	NO	
DEL	0	
INST	92	

Chart Continued

CHART 4-1 Continued

STEP	ACTION																																													
	<table border="1"> <thead> <tr> <th>PROMPT</th> <th>RESPONSE</th> <th>COMMENTS</th> </tr> </thead> <tbody> <tr> <td>REQ</td> <td>NEW</td> <td></td> </tr> <tr> <td>TYPE</td> <td>SCB</td> <td></td> </tr> <tr> <td>CUST</td> <td>2</td> <td>third SCB</td> </tr> <tr> <td>SCBN</td> <td>3</td> <td></td> </tr> <tr> <td>TRNK</td> <td>NO</td> <td></td> </tr> <tr> <td>DEL</td> <td>1</td> <td></td> </tr> <tr> <td>INST</td> <td>93</td> <td></td> </tr> <tr> <td>REQ</td> <td>NEW</td> <td></td> </tr> <tr> <td>TYPE</td> <td>SCB</td> <td></td> </tr> <tr> <td>CUST</td> <td>2</td> <td>fourth SCB</td> </tr> <tr> <td>SCBN</td> <td>4</td> <td></td> </tr> <tr> <td>TRNK</td> <td>NO</td> <td></td> </tr> <tr> <td>DEL</td> <td>0</td> <td></td> </tr> <tr> <td>INST</td> <td>0</td> <td></td> </tr> </tbody> </table>	PROMPT	RESPONSE	COMMENTS	REQ	NEW		TYPE	SCB		CUST	2	third SCB	SCBN	3		TRNK	NO		DEL	1		INST	93		REQ	NEW		TYPE	SCB		CUST	2	fourth SCB	SCBN	4		TRNK	NO		DEL	0		INST	0	
PROMPT	RESPONSE	COMMENTS																																												
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CUST	2	fourth SCB																																												
SCBN	4																																													
TRNK	NO																																													
DEL	0																																													
INST	0																																													

3 Create 4 ARS code data blocks (UAC, overlay 27).

ARS CODE	SCHEDULE BLOCK NO.
0	1
118	2
5	3
7	4

PROMPT	RESPONSE
REQ	NEW
TYPE	UAC
CUST	2
UACN	118
SCBN	2
UACN	0
SCBN	1
UACN	5
SCBN	3
UACN	
SCBN	a
UACN	
REQ	END

4.11 This completes the programming for this example of Autovon improved ARS.

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

AUTOVON FEATURE (GENERIC X11) DESCRIPTION

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* SL-1 is a trademark of Northern
Telecom Limited

1. GENERAL

1.01 The Automatic Voice Network (AUTOVON) is the name given to the major non-secure military telephone network in the U.S.A. It is linked to an equivalent Canadian network, the Canadian Switched Network (CSN). Both of these networks are completely separate from the public telephone network, and have their own numbering plans and special calling capabilities.

1.02 The two calling capabilities unique to AUTOVON are:

- (a) **Precedence.** Each call involving the AUTOVON network is identified with one of five priority levels. This priority level is determined when dialing the call.
- (b) **Preemption.** If all required trunks in a route are busy, a call on the route in any stage can be taken down or 'preempted' to free the trunk for a call of a higher priority.

1.03 The **SL-1** fills the role of an AUTOVON Private Branch Exchange (PBX) by having one or more trunk routes between the AUTOVON network and the **SL-1**. The **SL-1** call processing software for these routes is modified to provide proper signaling for AUTOVON calling as well as precedence and preemption. For non-AUTOVON trunk routes, the **SL-1** uses standard call processing software.

1.04 The **SL-1** is not intended to operate as a terminating PBX serving as an AUTOVON node. This type of operation requires significantly increased translation and class-marking capability and is more suited to CO equipment.

Associated NTP

1.05 The NTP describing the AUTOVON feature for Generic XI 1 Release 4 and its operation are:

553-2661-101	AUTOVON General Description
553-2661-181	AUTOVON Engineering Data
553-2661-301	Station sets and Attendant Console Operation for Autovon features.
553-2661-181	Engineering Consideration
553-2661-301	Operational Tests
553-2001-181	QPC71 2-Wire E&M/DX Trunk Description
553-2001-190	QPC237 4-Wire E&M/DX Trunk Description
553-2YY 1-220	Features and Services I: Data Administration
553-2YY1-221	Features and Services II: Service Change Input/Output
553-2311-310	Data Administration I: Feature Implementation
553-231 1-31 1	Data Administration II: Input/Output Reference

553-2751-103 Network and Basic Authorization Codes
Description

1.06 In addition to the documents listed above, other documentation describes the AUTOVON network as a whole:

I DCA Circular 370-V130-1: AUTOVON Switching Center Requirements

I DCA Circular 370-V175-6: AUTOVON Network specifications

1.07 The previous AUTOVON Generic, X14, is described in 553-2661-100.

2. AUTOVON FEATURES

PRECEDENCE

2.01 On the AUTOVON network, each call is marked with one of five priority levels according to the urgency of the call. The levels, from highest to lowest, are:

- (a) *Flash Override*. Flash Override (FO) calls are the highest priority calls in the AUTOVON system. They can not be preempted by any other call but can preempt any call except other FO calls.
- (b) *Flash*. Flash (F) calls are the second-highest priority calls in the AUTOVON system. They can be preempted only by FO calls. They cannot be preempted by other F calls or lower-level calls.
- (c) *Immediate*. Immediate (I) calls are the third-highest priority calls. They can be preempted by FO and F calls but not by other I or lower-level calls.
- (d) *Priority*. Priority (P) calls are the fourth-highest level calls. They can be preempted by FO, F, and I calls, but not by other P or lower-level calls.
- (e) *Routine*. Routine (R) calls are the lowest-level calls. They can be preempted by all but other R calls.

2.02 AUTOVON calls of a priority level above Routine (i.e., a 'precedence' call) can be initiated by:

- the attendant
- a station with the Class of Service (COS) required to access the AUTOVON trunk, and the Network COS (NCOS) for the required precedence level.

2.03 All AUTOVON calls initiated from TIE trunks are sent out over the AUTOVON trunks as Routine calls.

Attendant

2.04 The SL-1 attendant console is equipped with keys for precedence levels above Routine. The attendant can initiate calls of any precedence level, or can extend a call from a station to the AUTOVON network using any precedence level (including Routine).

Stations

2.05 Like the attendant console, SL-1 stations can be equipped with precedence keys for dialing precedence calls. The 500/2500 stations can dial precedence calls using the Special Function Prefix (SPRE).

2.06 Every station is assigned a Network Class of Service (NCOS) value, in which is defined a maximum allowable precedence level. A station may direct-dial calls of any precedence level up to and including this maximum precedence. Calls of any higher precedence must be extended to the AUTOVON network via the SE-1 attendant.

Tie Trunks

2.07 Tie trunks are assigned Trunk Group Access Restriction (TGAR) codes which govern their access to SL-1 facilities. Subject to this restriction, TIE trunks with the Fully Restricted 1 (FR1) COS or higher can access AUTOVON trunk routes.

PREEMPTION

2.08 AUTOVON trunk routes have a 'preemption' facility whereby any trunk that is not idle can be forcibly disconnected. The trunk may or may not be part of an established call (i.e., it may be in an established, signalling, or any other **nonidle** state). A **nonidle** trunk can be forcibly disconnected for one of two reasons:

- to use the subsequently freed trunk for another call (in which case the trunk will be held busy for that call), or
- to free a link for some other purpose, such as testing, in which case the freed trunk may or may not be used again immediately.

2.09 Trunk routes going out from the SL-1 to an ASC can be one of two types:

- *Nonpreemptable*. This trunk route can be used for routine-level calls only. It cannot be preempted from the SL-1 end but can be preempted from the ASC end. Over nonpreemptable trunk routes, precedence calls are sent out as routine.
- *Preemptable*. This trunk route can be used for any level of call and can be preempted from either the ASC or SL-1 ends.

2.10 When a call is preempted, both parties involved in the call (or all parties in the case of a conference) hear 3 s of preempt warning tone. The parties then have a certain length of time to disconnect. If the trunk is not disconnected within this time, the trunk is forcibly disconnected by whichever end the preempting call originates from. If the trunk is to be used for the preempting call, it is held busy until the preempting call is ready to access it. If the trunk is not to be used, it is made idle.

2.11 A trunk can be preempted only if it meets the following conditions:

- not in maintenance busy
- not in line lockout
- not access denied
- | not in use by an attendant
- not in any stage of preemption or outpulsing
- the precedence level of the call being preempted is less than the precedence level of the preempting call
- not a test line or in any stage of testing
- | not in end-to-end signaling configuration.

PROMOTION

2.12 After making a call, the caller can raise the precedence level of the call one or more times until the call is completed or the caller's maximum allowable precedence level is reached. For example, if a station or an attendant makes an AUTOVON call and an idle or preemptable trunk is not found, reorder tone is returned to the caller. At this point the caller can 'promote' the call by pressing a higher precedence key. The call is then placed again by the system. If there are still no idle or preemptable trunks available, the process can be repeated until a preemptable trunk is found or the station or attendant reaches its maximum allowable precedence level. (Calls cannot be promoted using SPRE codes.)

AUTOVONTRUNK TYPES

2.13 An SL-1 can have as many AUTOVON trunk routes as required (subject to normal system limits). These routes can be incoming only, 2-way, or outgoing only, or any combination thereof depending on system requirements.

Nonpreemptable

2.14 Nonpreemptable trunks routes between the ASC and the SL-1 can use Dual-Tone Multifrequency (DTMF) or Dial Pulse (DP) signaling for outgoing calls and DP only for incoming calls. This trunk route can be used only for Routine-level calls, which can be preempted from the ASC but not from the SL-1.

Preemptable

2.15 Preemptable trunk routes to the ASC use DTMF for outgoing calls and DP for incoming calls. This trunk route can be used for any level of call from Flash-Override to Routine. (Traditionally, however, this trunk route is only used for precedence calls.) Calls on this trunk route can be preempted from either the ASC or the SL-1.

AUTOVON TRUNK ACCESS

2.16 In general, AUTOVON trunks are treated similarly to TIE trunks, with Class of Service (COS) restrictions provided by Trunk Group Access Restrictions (TGAR) for trunks and Trunk Access Restriction Groups (TARG) for stations.

2.17 AUTOVON to AUTOVON trunk connections are allowed, subject, to TGAR and TARG in addition to any other limitations imposed by supervision or signaling incompatibilities.

2.18 Call *Modification*. Some types of access are allowed only through 'call modification'. This refers to such features as call transfer and call forward. With these features the AUTOVON call can be allowed or denied either on the basis of the originating station or trunk COS/TGAR, or on the basis of the forwarding/transferring station COS/TGAR. This option is selected in the Customer Data Block (overlay 15, prompt OPT).

Incoming AUTOVON Calls

2.19 Incoming AUTOVON trunks can be assigned any one of the following restriction levels for access to other trunk facilities:

- *Unrestricted (UNR)*. No restrictions.
- *Conditionally Unrestricted (CUN)*. Same as toll denied.
- *Conditionally Toll Denied (CTD)*. Same as toll denied.
- *Toll Denied (TLD)*. Local CO calls and TIE trunk calls allowed, toll CO calls denied unless made through the attendant or call modification (Note).

- Semi-Restricted (**SRE**). CO calls allowed only through the attendant or call modification; TIE trunk calls allowed.
- **Fully Restricted (FRE)**. CO calls allowed only through call modification. TIE trunk calls allowed.
- **Fully Restricted, Subgroup 1 (FR1)**. CO calls denied, TIE trunk calls allowed.
- | **Fully Restricted, Subgroup 2 (FR2)**. CO calls denied, TIE trunk calls denied.

Note: If Code Restriction is in effect, calls to CO trunks can be selectively allowed or denied. If the New Flexible Code Restrictions feature is in effect, calls to all trunk routes can be selectively allowed or denied on a per route basis.

Outgoing AUTOVON Calls

2.20 Dial access to outgoing AUTOVON trunks is allowed from any station or trunk, subject to any TGAR codes and Trunk Access Restriction Levels assigned to station, trunks, or AUTOVON trunks. It is recommended that direct dial access to the preemptable trunk group be limited to attendants and stations with allowed precedence level higher than routine. Outgoing routine calls on the nonpreemptable trunk group can automatically advance to this trunk group when all nonpreemptable trunks are busy (overlay 16, prompt STEP).

2.21 Stations and Non-AUTOVON trunks can be assigned any one of the following levels for access to outgoing AUTOVON trunks.

- | **Unrestricted (UNR)**. No restrictions, except that DID trunks may only obtain access through the attendant or call modification.
- | **Conditionally Unrestricted (CUN)**. Same as UNR.
- **Conditionally Toll Denied (CTD)**. Same as UNR.
- **Toll Denied (TLD)**. Same as UNR
- **Semi Restricted (SRE)**. Same as UNR.
- **Fully Restricted (FRE)**. Calls from CO allowed only through call modification, TIE trunk calls allowed.
- **Fully Restricted, Subgroup 1 (FR1)**. Calls from CO denied, tie trunk calls allowed.
- | **Fully Restricted, Subgroup 2 (FR2)**. Calls from CO denied, TIE trunk calls denied.

AUTOVON TONES, DIGITS, AND RINGING

2.22 To allow for precedence operation of the AUTOVON network, special AUTOVON signaling and progress tones are provided. These tones are provided by the Flexible Tone And Digit Switch (QPC251). In addition to AUTOVON tones, the QPC251 also supplies all standard North American tones.

Precedence Digits

2.23 DTMF signaling uses, short bursts of tone pairs to represent digits. For AUTOVON, the tone pairs are treated by the system as special digits denoting the precedence level of a call (also included are the DP equivalents for DP signaling):

- FO (flash Override): 697-l- 1633 Hz (10 pulses, i.e., digit '0')
- F (Flash): 770-l-1633 Hz (1 pulse, i.e., digit '1')
- I (Immediate): 852-l- 1633 Hz (2 pulses, i.e., digit '2')
- l P (Priority): 941-i-1633 Hz (3 pulses, i.e., digit '3')
- l R (Routine): the absence of any of the above tones in DTMF outpulsing indicates that the call is Routine-level. For DP outpulsing, 4 pulses (i.e., digit '4') represents Routine.

Preemption Warning Tone

2.24 Preemption Warning Tone is given to all parties in an AUTOVON call that is about to be preempted. It consists of 440 i-620 Hz.

AUTOVON DIALING PLAN

2.25 SL-1 internal DN are identical for internal calls, incoming AUTOVON calls, and direct dialed calls from the public network. SL-1 internal directory numbers can consist of two, three, or four digits.

Incoming Calls

2.26 The AUTOVON dialing plan assigns three-digit NXX codes to ASC. The SL-1 is considered part of this plan and can be assigned its own NXX code. Depending on how the ASC translates dialed digits, it can **outpulse** up to and including the seven digits of the PBX code and extension, not including the precedence digit, where applicable. Therefore, the digits outpulsed to the SL-1 can consist of:

{P}{NXX} DXXX

Where: P = precedence digit, NXX = PBX code, DXXX = SL-1 extension DN, and { } denotes that the digits are optional.

2.27 The precedence digit, P, is never included on incoming calls over nonpreemptable trunk routes, since all calls on this route are assumed to be routine (note). It is always included on incoming calls over **preemptable** trunk routes and can be the digit O(FO), 1(F), 2(I), 3(P), or 4(R).

Note: The only exception to this rule is if a call on this route is preempted by the ASC, in which case the P digit is included after preemption has occurred.

2.28 If the NXX digits are outpulsed from the ASC but not required by the SL-1 up to three digits can be absorbed through a customer defined option (overlay 16, prompt IABS). The precedence digit is not absorbed.

Outgoing Calls

2.29 **Nonpreemptable Routes.** Dialing to an ASC over a nonpreemptable trunk route can be DTMF or DP. Since all calls placed by an SL-1 over this type of trunk route are routine and cannot be preempted by the SL-1, precedence digits are not outpulsed. Thus digits outpulsed over a nonpreemptable AUTOVON trunk route can consist of:

{RC} {NYX} NXX DXXX

Where RC is the route code, NYX is the AUTOVON area code (required if calling to another AUTOVON area), NNX is the code of the ASC to which the call is ultimately destined, and DXXX is the required extension. { } indicates optional digits. (The trunk access code is not shown.)

2.30 Route codes indicate the routing treatment to be given to the call over the ASC network and may or may not be included depending on the configuration of the ASC. For example, SL-1 Data calls using an Add-On data module may require line conditioning or switching out of echo suppressers.

2.31 The AUTOVON area code is required if calling to another AUTOVON area.

2.32 The user may dial some or all of the above codes. Whatever dialing plan is used must be coordinated with the ASC and AUTOVON network.

2.33 *Preempt&e Routes.* Dialing to an ASC over a preemptable trunk route is DTMF only. The P digit is not outpulsed for Routine calls, but is outpulsed for Priority or higher-level calls, and is derived from:

- the P, I, F and FO feature keys on the attendant console or SL-1 station set, or
- the equivalent SPRE and precedence code sequence.

2.34 These digits outpulsed over a preemptable trunk route can consist of:

{P} {RC} {NYX} NXX DXXX

(The trunk access code is not shown).

CALL DETAIL
RECORDING
ENHANCEMENT

2.35 The precedence tracking Call Detail Recording (CDR) option can be used to maintain a record of stations making precedence calls. When the AUTOVON CDR Enhancement package is equipped, an additional field is appended to CDR records pertaining to call initiation, modification and termination for the station making the call. This field contains the precedence level of the call, in the form of a blank and a single digit:

- 0 - AUTOVON Flash Override
- 1 - AUTOVON Flash
- 2 - AUTOVON Immediate
- 3 - AUTOVON Priority
- 4 - AUTOVON Routine
- 5 - Non-AUTOVON.

2.36 Codes 4 and 5 are stored on magnetic tape but are not printed out on TTY. A more detailed description of the CDR magnetic tape format and TTY format modification is given in 553-2631-100.

SL-1 AUTOVON
TRUNK INTERFACE

2.37 The AUTOVON precedence level is not appended to the CDR initialization and time stamp records.

2.38 The AUTOVON network uses 4-wire transmission throughout. The types of SL-1 trunk that can be used for AUTOVON trunk routes are:

- QPC237 4-wire E&M/DX trunk or
- QPC71 2-Wire E&M/DX trunk, with add-on 24V4 repeater.

2.39 SL-1 software programs have been tailored to suit the signaling and call processing requirements of AUTOVON, including detection of 'precedence' digits, preemption, and special treatment of precedence calls. The sequence and timing of signals on AUTOVON trunks is discussed in 555-2661-181. A description of the QPC237 4-wire E&M/DX trunk is contained in 553-2001-190. A description of the QPC71 2-wire E&M/DX trunk is contained in 553-2001-187.

DISTINCTIVE
RINGING AND
RINGBACK

2.40 Incoming AUTOVON precedence calls are distinguished by a special triple-ringing signal. For SL-1 sets this involves a special ringing cadence and ring duration. For 500/2500 sets, special ring timing is provided.

2.41 SL-1 sets obtain the distinctive ringing cadence from the QPC251 TDS card. The special cadence for tone ringing and ringback is 1.64 s on and 0.36 s off. The normal tone ringing frequencies are used, but the special ringback tone consists of 440-1-480 Hz.

2.42 For 500/2500 sets the distinctive ringing cadence is 1.54 s on and 0.38 s off. The special ringback tone of 440-1-480 Hz is also provided.

2.43 This distinctive ringing, if selected, is retained by the call regardless of how it is modified (e.g., Attendant Intercept, Night Intercept, Call Park, etc.).

Ringing Power Supply

2.44 Because the distinctive ringing cycle involves 2.5 times as much 'on' ringing time as normal ringing, the number of 500/2500 set ringers that may be rung at any one time is reduced. The amount of the reduction depends on what percentage of calls are expected (at maximum) to receive distinctive ringing.

3. ATTENDANT FEATURES

3.01 To allow the attendant to handle various precedence levels of calls, the attendant console in an AUTOVON equipped SL-1 can be equipped with several keys unique to AUTOVON operation.

3.02 Incoming Call *Indicators* (ICI):

- *AUTOVON Routine (A VR)*. Routine level calls coming into the SL-1 from AUTOVON trunk routes appear on this ICI key. Calls presented on this ICI key are transferred to a free Loop Pickup Key (LPK) when their turn in the LPK queue arrives.. AUTOVON Routine calls are queued as normal TIE trunk calls.
- *AUTOVON Precedence (PRC)*. Precedence calls coming into the SL-1 from AUTOVON trunks routes appear on this ICI key. These calls go to the front of the attendant queue in order of precedence (all FO calls first, F calls second, etc.).
- *AUTOVON Precedence Intercept (PIN)*. Calls appearing on this ICI key are precedence calls from the AUTOVON network that have been misoperated, i.e., dialed towards vacant or maintenance busy numbers, blocked in the network, slow-answer recalled or Distinctive ringing No Answer (DFNA) calls.

3.03 *Precedence Keys*. In the AUTOVON network, the normal AUTOVON telephone set has a 16-key dial pad. This comprises a normal 12-key pad with keys added for the FO, F, I, and P precedence digits. Since SL-1 does not provide the 16-key pad, separate keys are used to outpulse the FO, F, I, and P digits. (These digits can also be outpulsed using SPRE+ access codes.)

INCOMING CALLS

3.04 *Digit Display*. Incoming Routine calls from an AUTOVON trunk route cause the digit display to display the trunk access code and member number as in normal operation. For incoming precedence calls, the precedence level is displayed, followed by the trunk access code and member number. The precedence level for digit display purposes is coded as follows: 0 = FO, 1 = F, 2 = I, and 3 = P. For example a Flash call from trunk member 7 of an AUTOVON trunk route with access code 21 would be displayed as 1-21 -7. A Routine call would be displayed as 21-7.

Routine Calls

3.05 A routine call from an AUTOVON trunk route appears on the LPK key or the AUTOVON Routine (AVR) ICI key if a LPK is not available. When the call reaches the LPK key, the digit display shows the trunk access code and member number of the trunk used for the incoming call. The attendant can then treat the call in the same way as a normal TIE trunk call.

Precedence Calls

3.06 A precedence call from an AUTOVON trunk route appears on a LPK or on the AUTOVON Precedence (PRC) ICI key if a LPK key is not available. When a LPK key becomes available, the precedence call is presented first, regardless of whether its turn in the LPK queue has come or not. The attendant can then extend the call in the same way as a TIE trunk call.

Precedence Intercept
Calls

3.07 Precedence calls from the AUTOVON network that do not complete for various reasons are re-routed to the attendant via the PIN ICI key. When a LPK key becomes free, the precedence call is presented even if other calls have been waiting in the queue longer. The digit display shows the DN that was originally dialed by the caller. The attendant can display the precedence level, access code and member numbers of the AUTOVON trunk by pressing the 'Display Source' key. The attendant can then transfer the call or perform other functions as normal.

Precedence Calls
Waiting

3.08 The attendant is informed of the arrival of a precedence call even if busy processing some other call. When the precedence call arrives in the attendant queue, the CW lamp begins to flash and the attendant tone ringer sounds. The CW lamp continues to flash, and the tone ringer continues to sound, as long as there are precedence calls in the queue.

Precedence Completion
To Busy

3.09 If a precedence call is made to a busy station, the call is returned to the attendant via the PIN key. The attendant can then dial the originally dialed DN to complete the call. If the precedence level of the incoming call is greater than that of the established call, the attendant can then enter the conversation on the busy line and 'talk off' the connection to allow the precedence call to complete.

3.10 If the attendant decides not to complete the incoming call to the busy station, pressing the Release Destination key leaves the established call unmodified.

3.11 The attendant cannot enter the conversation at the busy station if any of the following conditions are encountered. In these cases, overflow tone is returned to the attendant if:

- either party of the current call is on hold
- the call on the dialed station is parked
- the dialed station has Make Set Busy activated
- the dialed station has Group or Individual Do Not Disturb active
- the dialed set is locked out
- a connection between the dialed set and the incoming call is 'access denied' because of TGAR, TARG, COS etc.,
- there is an attendant associated with the call at the busy station
- the call at the busy station is already in the process of being preempted
- the current call at the busy station is 'Camped On' or 'Call Waiting' on another set.
- the busy station is dialing
- the incoming call was unable to complete because of network blocking
- the precedence level of the current call is equal to or greater than that of the incoming call

- there is already another 'Call Waiting' to talk to the station
- the dialed set is maintenance busy.

3.12 If any of the following conditions are encountered at the dialed station, the call is modified as indicated:

- **Station to Station Calls.** If the busy station is engaged in a call to another station, the call is assumed by the system to be Routine, and completion to busy is permitted.
- **Order of Disconnection.** If the busy station hangs up before the attendant tries to complete to busy, the attendant must reextend the call as normal.
- l **Ring Again.** If the dialed station has 'Ring Again' activated on a trunk or another station, it will be held away from the 'Ring Again' call until the precedence call is completed.

OUTGOING CALLS

3.13 Traditionally, AUTOVON access from PBX systems has been controlled by the PBX attendant. When a caller requires access to the AUTOVON network, the caller dials the attendant and asks the attendant to place the call. The SL-1 attendant has the capability to fulfill this role.

Routine Calls

3.14 The attendant can extend Routine calls to AUTOVON trunks in the same way as for normal trunks, i.e., the attendant dials the AUTOVON trunk access code, receives dial tone, and dials the required extension number. If when dialing the trunk access code all trunks in the route are busy, overflow tone is returned to the attendant.

Precedence Calls

3.15 To extend a precedence call, the attendant dials whichever of the four precedence codes is required, then the AUTOVON trunk access code, then the AUTOVON extension number. The dialing of precedence codes can be done in one of two ways:

- (a) **Precedence Feature Keys.** The attendant console in an AUTOVON SL-1 can be equipped with one feature key for each of the four precedence levels: FO, F, I, and P. To dial the precedence digit when making an AUTOVON call, the attendant simply presses the appropriate feature key. This code can be entered either before or after the AUTOVON preemptable trunk access code is dialed.
- (b) **SPRE Codes.** The attendant can also dial a precedence digit by dialing the special function Prefix (SPRE), then the 2-digit code for the required precedence level as follows:
 - FO: SPRE + 8 + 2
 - l F: SPRE + 8 + 5
 - I: SPRE + 8 + 8
 - o P: SPRE + 8 + 0

3.16 When the attendant dials a precedence call, the digits are shown on the digit display as in the case with other types of calls. The precedence digits are shown on the digit display in their translated form, i.e., in the same form as the ASC would outpulse them to the SL-1 for incoming calls:

- FO = 0
- F = 1
- I = 2
- l P=3.

3.17 Thus if the attendant were to dial a flash call with trunk access code 21 to extension 737-2772, regardless of whether feature keys or SPRE codes were used to dial the F digit, the digit display would show 1-217372772.

3.18 If all AUTOVON trunks are busy with calls of equal or higher precedence than the one being extended by the attendant, re-order tone is returned to the attendant. At this point, the attendant may 'promote' the call by pressing a higher precedence key. (Calls cannot be promoted by dialing SPRE + precedence code.) The SL-1 removes the re-order tone and scans the trunks in the route again for a preemptable call. This procedure may be repeated until the call goes through or is abandoned.

Dial Service Assistance
Recall

3.19 Stations engaged in AUTOVON calls may need to recall the AUTOVON Dial Service Assistance (DSA) operator for special call completion or conferencing purposes. When a station engaged in an AUTOVON call recalls the attendant and requests a DSA recall, the attendant presses the Signal Destination key. This causes the SL-1 to send a switch-hook flash (of programmable length: overlay 15, prompt SDFL) to the AUTOVON network, resulting in a recall of the DSA operator. When the DSA operator answers, the attendant can connect the operator with the station using normal procedures.

Speed Call

3.20 For Speed Call, System Speed Call (SSC), and Network Speed Call (NSC), the precedence digits may be stored in the speed call lists in SPRE code form (i.e., the stored number is SPRE + 8 + x). Precedence keys cannot be used to store precedence digits. When the speed call entry is invoked, the call is placed as if the precedence digit had been dialed from the keypad.

3.21 A Precedence key cannot be pressed before a Speed Call key.

3.22 If the call is blocked by a busy trunk route, the attendant may promote the call in the normal manner.

4. STATION FEATURES

4.01 Routine calls to the ASC may be directly dialed from any station for which access to nonpreemptable trunk groups is allowed by the restrictions of COS, TGAR, etc. Station users may also gain access to AUTOVON facilities through the attendant. Stations allowed access to a preemptable trunk route may also direct dial precedence calls up the the maximum precedence level specified in the Network Class Of Service (NCOS) assigned to the station.

4.02 The four precedence keys FO, F, I, and P that can be assigned to an attendant can also be assigned to an SL-1 set. SL-1 sets can place routine and precedence calls in the same way as the attendant console, i.e., dial the precedence digit, (if required), trunk access code, and extension number. Stations may also promote calls in the same way as attendants.

5. FEATURE INTERACTIONS

	5.01 The operation of the following features may be affected when they are used in conjunction with AUTOVON.
Attendant Administration	5.02 AUTOVON cannot be administered using the Attendant Administration feature.
Attendant Extended Calls	5.03 Attendant extended precedence calls distinctively ring terminating sets.
Attendant Intercept	5.04 <i>Busy</i> or <i>No-Answer DN</i> . Precedence calls are automatically routed to the attendant if the called station is busy or does not answer within a specified period of time. If the called station has invoked Hunting or Call Forward (all calls), these act in the normal manner, but if the call still cannot be completed it is routed to the attendant Precedence Intercept (PIN) ICI. The CFNA timer for precedence calls has its own separate timer in customer data (prompt DFNA), as distinct from the normal (routine) CFNA. 5.05 <i>Maintenance DN</i> . Misdialed but complete precedence calls to maintenance DN (which are preceded by the Special Function Prefix (SPRE) digit(s)) are also intercepted to the attendant. 5.06 <i>Other Calls</i> . The following types of calls may have intercept treatment specified in the customer data block. If they are precedence calls, they are intercepted to the attendant regardless of this specified treatment: <ul style="list-style-type: none"> ● calls to DN for which access is denied ● calls to vacant DN ● calls to maintenance-busy DN ● code or toll-restricted calls from toll-denied AUTOVON trunks.
Authorization Codes	5.07 Station users can raise the level of restrictions, including precedence levels, by using the 'Basic Authorization Codes' (BAUT) capability. Each authorization code stored in the system is assigned on NCOS value which defines the maximum allowable precedence. This feature allows station users to transport their own station's capabilities to another station, even though the other station may be more restricted. Network Authorization Codes (NAUT) apply only to Electronic Switched Network (ESN) and therefore cannot be used to access AUTOVON Routes. 5.08 <i>Routine Calls</i> . To place a Routine AUTOVON call using an authorization code, the user dials: <ol style="list-style-type: none"> (1) SPRE + 6 (2) an authorization code that allows access to the nonpreemptable AUTOVON trunk route (3) the trunk access code

(4) the AUTOVON network number.

5.09 Precedence Calls. To place a precedence call to the AUTOVON network using an authorization code, the user dials:

- (1) SPRE + 6
- (2) an authorization code that allows access to the preemptable AUTOVON trunk route at the desired precedence level
- (3) SPRE + precedence level code (or press the appropriate precedence key) for desired precedence level
- (4) trunk access code for the preemptable AUTOVON trunk route
- (5) the AUTOVON network number.

Automatic Answerback

5.10 If a precedence call terminates on a set in Automatic Answerback (AAB) mode, Distinctive Ringing is applied until AAB occurs.

5.11 Automatic Answerback answers precedence calls the same as as routine calls. If there is nobody at the station set to speak to the caller when answer back occurs, this could cause confusion and call processing difficulty. Users are advised to turn off this feature when away from the set.

Automatic Call Distribution

5.12 Precedence calls to an ACD-DN are not permitted to queue for an Automatic Call Distribution (ACD) agent. If an idle agent is not immediately available, such a call is intercepted to the attendant. The attendant may then transfer the call to the ACD-DN and stay on the line waiting for an answer. In this case the call is placed in the ACD high-priority queue.

Auto Dial

5.13 The Auto Dial key may not be programmed to contain SL-1 precedence keys. If an Auto Dial is to contain precedence the SPRE code must be used. However, Auto Dial calls may only be promoted using precedence keys.

Buzz for SL-1 Sets

5.14 If a precedence call terminates on an SL-1 set and buzzing is to be applied, the buzzing is continuous. Routine calls result in the normal 2-s buzz.

Call Detail Recording

5.15 Since TTY printouts from AUTOVON enhanced Call Detail Recording (CDR) are 2 characters wider than the conventional CDR TTY printout, there is a possibility that the enhanced format may not fit on some printers.

5.16 CDR tapes and CDR data are subject to many types of downstream processing, both from outside firms and by Northern Telecom e.g., for the Communication Mangement Center. If the AUTOVON enhancement for CDR is equipped, any downstream processing systems must be modified to accommodate the changed format.

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Call Forward	<p>5.17 AUTOVON calls from the AUTOVON network cannot be forwarded back into the AUTOVON network. AUTOVON trunk calls to stations which are call forwarded to outgoing trunks are subject to TGAR and COS restriction. For example, if the the AUTOVON trunk group is denied access to the CO trunk group (either by TGAR restriction or by assigning an FR1 or FR2 COS to the AUTOVON trunks group), Routine calls from an AUTOVON trunk to a station forwarding calls over a CO trunk are intercepted according to the specified intercept treatment. Precedence calls are always intercepted to the attendant. AUTOVON calls coming into a call-forwarded DN either retain the COS/TGAR assigned to the incoming AUTOVON trunk, or take those of the forwarding set, whichever is specified in the Customer Data Block (overlay 15, prompt OPT).</p> <p>5.18 Distinctive Ringing is preserved for forwarded precedence calls.</p>
Call Park	<p>5.19 Calls that are station-parked and timeout and are recalled to a busy station are not intercepted back to the attendant, even if the call is precedence call. Distinctive Ringing is preserved for precedence calls.</p>
Call Pickup	<p>5.20 Precedence calls to a station in a Call Pickup group can be picked up from another station in the same group, provided the connection is allowed by TGAR, COS, etc. The maximum allowable precedence level of the station performing the pickup can be lower than the precedence level of the incoming call.</p>
Call Transfer	<p>5.21 Distinctive Ringing is preserved for transferred precedence calls.</p>
Called Party Disconnect Control	<p>5.22 Called Party Disconnect Control (CPDC) is not permitted for AUTOVON trunks.</p>
Centralized Attendant Service	<p>5.23 Precedence digits cannot be transmitted across Release Link trunks. Thus a Centralized Attendant Service (CAS) Main attendant cannot extend an AUTOVON precedence call over AUTOVON trunks at a CAS Remote location. A precedence call from a CAS Remote station can be extended by a CAS Main attendant only over AUTOVON trunks at the CAS Main location. In this case, precedence and promotion keying can only be done by the CAS Main attendant.</p>
Conference	<p>5.24 All stations conferenced into a precedence call are rung distinctively.</p>
Direct Inward System Access	<p>5.25 Precedence calls to a Direct Inward System Access (DISA) DN are intercepted to the attendant. Routine calls to a DISA DN receive normal intercept treatment. It is not possible to place precedence calls from a DISA DN.</p>
Electronic Switched Network Features	<p>5.26 AUTOVON trunk routes are not permitted in the route lists for Network Alternate Route Selection (NARS), Basic Alternate Route Selection (BARS), or Coordinated Dialing Plan (CDP), due to possible preemption interactions. The facilities of NARS/BARS/CDP can be used to access another PBX where AUTOVON trunks terminate. Routine AUTOVON calls can then be placed using the facilities at the other PBX, and AUTOVON precedence calls can be placed only through the attendant at the other PBX.</p> <p>5.27 <i>Speed Call</i>. The precedence level of the call cannot be greater than the maximum precedence level defined for the Network Class of Service (NCOS) of the call.</p>

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- For System Speed Call (SSC), the NCOS of the call is either the originator's NCOS or the NCOS of the SSC list, whichever has the higher Facility Restriction Level (FRL).
- For Network or regular Speed Call, the NCOS of the call is the originator's NCOS.

Flexible Hot Line

5.28 ASC incorporate a feature called Off-Hook, or Hot Line Service as outlined in DCA Circular 370-V130-1. This feature is required for applications where a call must be established through the AUTOVON network automatically as a result of lifting a handset. SL-1 incorporates a similar feature, the 'Flexible Hot Line'. Flexible Hot Line service can be provided to designated 500/2500 sets via Data Administration (service change). Each station so designated must be assigned the Manual Line (MNL) COS and digits must be stored via service change to dial the required DN. For AUTOVON, these may include precedence digits (in SPRE form), route codes, AUTOVON trunk access codes, and AUTOVON network number. The DN cannot be modified by the user.

Forced Charge Account

5.29 It is not possible to automatically promote the precedence of a call using Forced Charge Account (FCA). The call must be re-dialed at the higher precedence level.

Hunting

5.30 Precedence calls terminating on a hunt DN are rung distinctively.

Make Set Busy

5.31 Routine calls to a set which is made busy are given busy tone. Precedence calls are routed to the attendant.

Message Center

5.32 Precedence calls to busy or non-answering stations are not forwarded to the Message Center, Integrated Messaging System (IMS), or Integrated Voice Messaging System (IVMS), even if this is specified by CFNA. These calls are intercepted to the attendant. Precedence calls dialing the message center directly are handled as other ACD calls if the message center is an ACD type. Calls to a DN-type message center are treated as those to a normal DN.

Multiple Precedences

5.33 If a station user dials multiple precedence codes before accessing an AUTOVON trunk the last precedence code dialed is used. This allows Speed Call and Auto Dial numbers containing precedence digits to be promoted to higher precedence levels after dialing.

Night Intercept

5.34 Incoming AUTOVON calls can be automatically re-routed to an assigned night station when the attendant position is not manned. Such calls are queued for service and the night station can transfer the calls as applicable. The queuing is in order of precedence. Thus, an incoming call of higher precedence than those already in the queue moves to the front of the queue. However, a 'call waiting' or 'camping on' the night station cannot be displaced by a call of higher precedence. Listed Directory Number (LDN) calls are treated the same as Routine calls for queuing purposes. The night station does not have local preemption capabilities, and outgoing calls from the night station to the AUTOVON trunk are treated like those from any other subscriber station.

Office Data
Administration System

5.35 Overlay 81 and 83 recognize the SL-1 Precedence Keys.

PRACTICE 553-2661-101

Precedence Attendant Access	5.36 Station users may dial a precedence SPRE code and then dial '0' for the attendant. This call is then placed in the attendant priority queue and appears on the PRC ICI. This provides two advantages: it offers priority access to the attendant, and helps the attendant verify that the calling station does have precedence access.
Recorded Overflow Announcement	5.37 Incoming calls of Priority or higher-level precedence do not receive Recorded Overflow Announcement (ROA) even if the customer has this option enabled. Routine calls receive whatever ROA treatment is specified for normal calls.
Ring Again	5.38 Upon finding the AUTOVON trunk group busy, station users may use the Ring Again feature. The precedence level is stored for the call and used when outpulsing to the ASC.
Special Function Prefix	5.39 Calls incoming over AUTOVON trunks cannot use dial access features which require the use of Special Function Prefixes (SPRE).
Speed Call	5.40 The precedence SPRE codes may be stored in Speed Call lists (note). When a speed call list is programmed the precedence digits must be dialed (precedence keys are not stored). Calls are placed as if the precedence access code was dialed from the keyboard. If the call is blocked, SL-1 set users may promote the call to a higher precedence.
Station to Station Calling	5.41 Station users may dial a precedence SPRE code and then an SL-1 DN. The call is completed as normal and receives no special treatment.
Stored Number Redial	5.42 If a user places a precedence call using SL-1 set precedence keys, the precedence of the call is not stored. The call may, however, be promoted in the usual way.



 BUSINESS COMMUNICATIONS SYSTEM

SL-1*

AUTOVON FEATURE (GENERIC X11) INTERFACE DESCRIPTION AND OPERATION

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1. GENERAL

1.01 The Automatic Voice Network (AUTOVON) is the name given to the major non-secure military telephone network in the U.S.A. It is linked to an equivalent Canadian Network, the Canadian Switched Network (CSN). Both of these networks are completely separate from the public telephone network, and have their own numbering plans and special calling capabilities.

1.02 The SL-1 fills the role of an AUTOVON Private Branch Exchange (PBX) by having one or more trunk routes between the AUTOVON network and the SL-1. This practice describes the trunks, signaling and tones used for AUTOVON trunk routes, and in general provides information useful when planning the SL-1 interface to an AUTOVON Switching Center (ASC).

Note: This practice describes the AUTOVON feature for Generic X1 1 Release 4. Engineering data for the Generic X14 AUTOVON feature can be found in 553-2661-180.

1.03 *Reference Documentation:* The following documents may also be useful when planning the SL-1 interface to an ASC:

553-2001-151 553-2201-151	Engineering and Assignment of Equipment
553-2001-182	Summary of Transmission Parameters
553-2001-187	2-wire E&M/DX Trunk Description
553-2001-190	4-wire E&M/DX Trunk Description
553-2001-450	Traffic Measurement
DCA Circular 370-V130-1	AUTOVON Switching Center Requirements
DCA Circular 370-V175-6	AUTOVON Transmission Specifications

2. AUTOVON TRUNK INTERFACE

2.01 The AUTOVON network uses 4-wire transmission throughout. The two types of SL-1 trunks that can be used for AUTOVON trunk routes are:

- QPC237 4-wire E&M/DX trunk (this is the preferred type), or
- l QPC71 2-wire E&M/DX trunk, with add-on 24V4 repeater to convert to 4-wire operation.

2.02 SL-1 software programs have been tailored to suit the signaling and call processing requirements of AUTOVON, including detection of 'precedence' digits, preemption, and special treatment of precedence calls. For a complete description of the QPC71 and QPC237 circuit packs, read 553-2001-187 and 553-2001-190 respectively.

2.03 An SL-1 can have as many AUTOVON trunk routes as required (subject to normal system limits). These routes can be incoming only or outgoing only (recommended for heavy traffic routes to minimize the possibility of glare), or 2-way (suitable for light traffic routes). The trunk routes can be of two types: preemptable and non-preemptable.

2.04 Nonpreemptable. Nonpreemptable trunk routes between the ASC and the SL-1 use Dual Tone Multifrequency (DTMF) or Dial Pulse (DP) signaling for outgoing calls and DP only for incoming precedence calls. This trunk route can be used only for routine-level calls (see Note). Any precedence calls made on this route by the SL-1 are sent out as routine.

Note: The only exception to this rule is if the ASC preempts a call on this route and sends another call of higher precedence. The SL-1 cannot preempt calls on the route.

2.05 Preemptable. Preemptable trunk routes to the ASC are DTMF for outgoing calls and DP for incoming calls. This trunk route can be used for any level of call from Flash Override to Routine (traditionally, however, this trunk route is reserved for precedence calls). Calls made on this trunk route can be preempted from either the ASC or the SL-1.

AUTOVON Trunk Options

2.06 AUTOVON trunk routes require the following options:

- *Wink Start.* ASC use wink start operation for trunk seizure and preemption. Wink start is selected in the trunk data block, overlay 14, prompt STAR, response 'wnk'.
- *Either End Disconnect Control.* This is required to allow preemption of AUTOVON trunks from either the ASC or the SL-1. Disconnect control is selected in the route data block, overlay 16, prompts NEDC and FEDC. Response to both is 'eth'.

TRANSMISSION

2.07 The QPC237 trunk circuit is the SL-1 4-wire E&M/DX trunk interface which does not require external 2- to 4-wire conversion equipment. In addition, the QPC237 can also provide control signals for external echo suppressors.

PRACTICE 553-2661-181

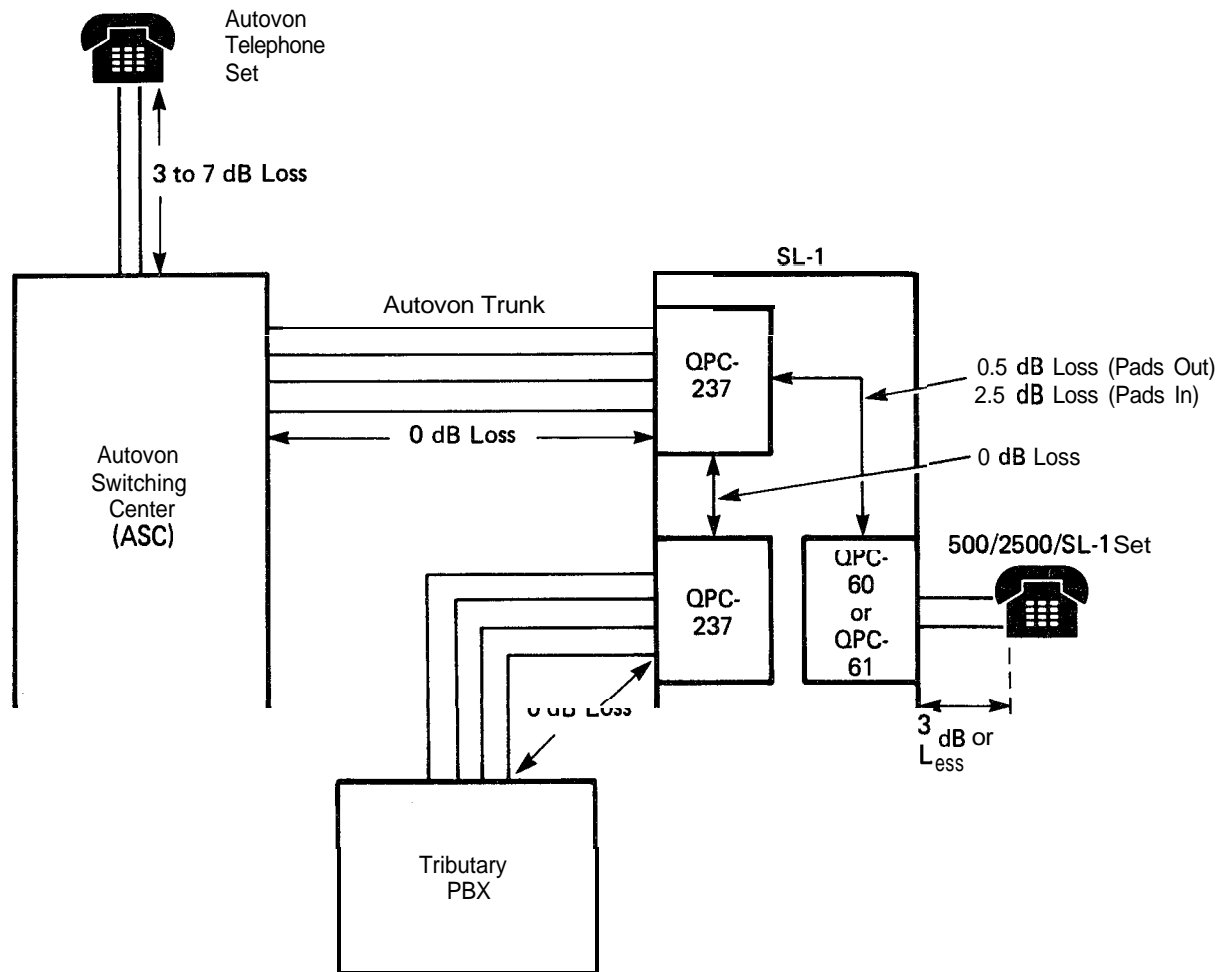


Fig. 2-1
AUTOVON Transmission Losses

2.08 Figure 2-1 shows an SL-1 arranged for interfacing to an ASC. DCA circular 370-V175-6 specifies a loss of 0 dB on PBX access lines from 4-wire PBX to the ASC. 2-wire user loops from the ASC have a design loss averaging 3 to 7 dB maximum. A QPC237 trunk introduces a loss of 0.5 dB (pad out) on connections between the QPC237 and QPC60 or QPC61 line cards to SL-1 stations. With the pads switched in at the QPC237, the loss is 2.5 dB. Typical SL-1 subscriber loops have a loss of 3 dB or less. Thus the loss introduced between an SL-1 station and the ASC conforms to ASC requirements. On trunk-to-trunk connections between an ASC and a PBX the SL-1 has a 0 dB loss between QPC237's, as required by AUTOVON.

TRUNK SEIZURE SIGNALING

2.09 The following paragraphs outline the trunk signaling sequences for incoming and outgoing AUTOVON calls, and call preemption.

SL-1 Seizure Sequence
(Idle Trunk)

2.10 Figure 2-2 shows the signaling from the time a free trunk has been identified and is required for use until the talking state is reached. The SL-1 seizure sequence is as follows:

- (1) The SL-1 seizes the trunk by going Off-Hook (battery on M Lead).
- (2) The SL-1 starts Glare timing (default 48 ms) and Seizure Supervision timing (default 5 s). The Glare timing interval is programmable between 0 and 2 s in 8 ms increments to accommodate both No. 1 ESS start pulse (which can be as short as 40 ms after SL-1 seizures), and satellite facilities (where up to 800 ms delay can be expected).
 - (a) If an Off-Hook is received from the ASC within the Glare interval, the SL-1 assumes a request for service and switches to the incoming call, rerouting the outgoing call on another trunk.
 - (b) If Off-Hook is not received within the Seizure Supervision period, the SL-1 assumes Trunk failure and flags diagnostic routines.
- (3) The ASC sends back a Start Pulse (Off-Hook). If the pulse is too short (less than 100 ms), it is ignored by the SL-1. If the pulse is too long (more than 350 ms), the SL-1 assumes Glare, in which case it reroutes the outgoing call on another trunk and locks the trunk out for 30 s. The ASC returns dial tone when the Start Pulse is sent.
- (4) **The SL-1 delays 128-256 ms after the start Pulse before sending digits.**

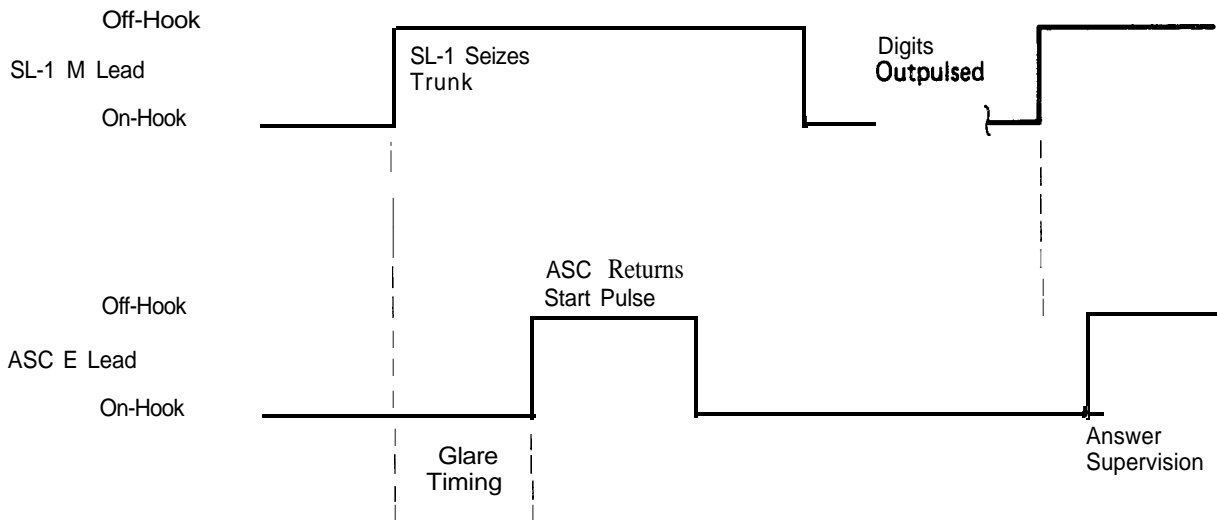


Fig. 2-2
SL-1 Seizes AUTOVON Trunk

ASC Seizure Sequence

2.11 Figure 2-3 shows the signaling from the time of incoming trunk seizure by the ASC until the digits are received by the SL-1. The ASC seizure sequence is as follows:

- (1) An **Off-Hook** from the far end (ground on E lead) is detected by the **SL-1**.
- (2) The **SL-1** marks the trunk busy and determines that the seizure persists for at least 128 ms.
- (3) The **SL-1** sends a start pulse (Off-Hook) for 256 ms and prepares to receive digits. It also starts the Stuck Sender timer.
 - (a) If the first digit is not received within the Stuck Sender period, a faulty trunk is assumed and the trunk is put in a line lock-out state. Diagnostics are flagged, and a minor alarm is raised.
 - (b) If the first digit is received within 5 s, incoming digit absorption occurs (if specified for the route), excluding the precedence digit.

ASC PREEMPTION SIGNALING

2.12 The following sequences describe the events occurring when the ASC preempts a call established or in the process of being established on a trunk. The trunk may or may not be used again. The initial conditions on the trunk can be as follows:

- 1 SL-1 On-Hook/ASC Off-Hook
- SL-1 Off-Hook/ASC Off-Hook
- SL-1 Off-Hook/ASC On-Hook.

2.13 If both ends are On-Hook, Preemption is not valid since the trunk is already idle.

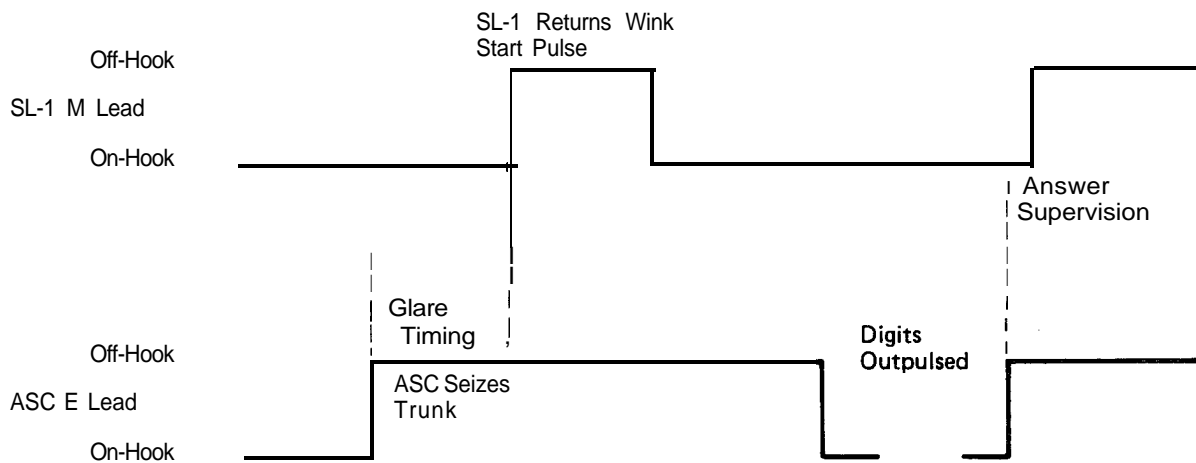


Fig. 2-3
ASC Seizes Trunk to SL-1

2.14 *SL-1 On-Hook/ASC Off-Hook*. This state occurs when a call is in the process of being established, i.e., the ASC is waiting for a Start Pulse or is in the middle of sending digits. The sequence for preemption by the ASC is as follows (see Fig. 2-4):

- (1) The ASC sends a Preempt Wink (280-416 ms On-Hook).
 - (a) If the circuit is to be reused, the ASC remains Off-Hook and waits for a Start Dial pulse from the SL-1.
 - (b) If the circuit is not to be reused, the ASC remains Off-Hook for 95-105 ms, then returns On-Hook for a valid disconnect period.

2.15 *SL-1 Off-Hook/ASC Off-Hook*. This is the normal talking state. The signaling sequence for preemption by the ASC is as follows (see Fig. 2-4):

- (1) The ASC sends a Preempt Wink (On-Hook) for 280-416 ms, then Preempt Tone for up to 3 seconds.
 - (a) If the circuit is to be reused, the ASC remains Off-Hook waiting for the Start Dial pulse from the SL-1.
 - (b) If the circuit is not to be reused, the ASC returns Off-Hook for 95-105 ms, then On-Hook for a valid disconnect period.

2.16 *SL-1 Off-Hook/ASC On-Hook*. The situation occurs on an outgoing call from the SL-1 after the ASC has returned a Start Dial pulse but before the called station answers. The signaling sequence for preemption by the ASC is as follows (see Fig. 2-5):

- (1) The ASC transmits a momentary Off-Hook for 95-105 ms, then Preempt Wink (On-Hook) for 280-416 ms.
 - (a) If the circuit is to be reused, the ASC remains Off-Hook waiting for the Start Dial pulse from the SL-1.
 - (b) If the circuit is not to be reused, the ASC returns Off-Hook for 95-105 ms, then On-Hook for a valid disconnect period.

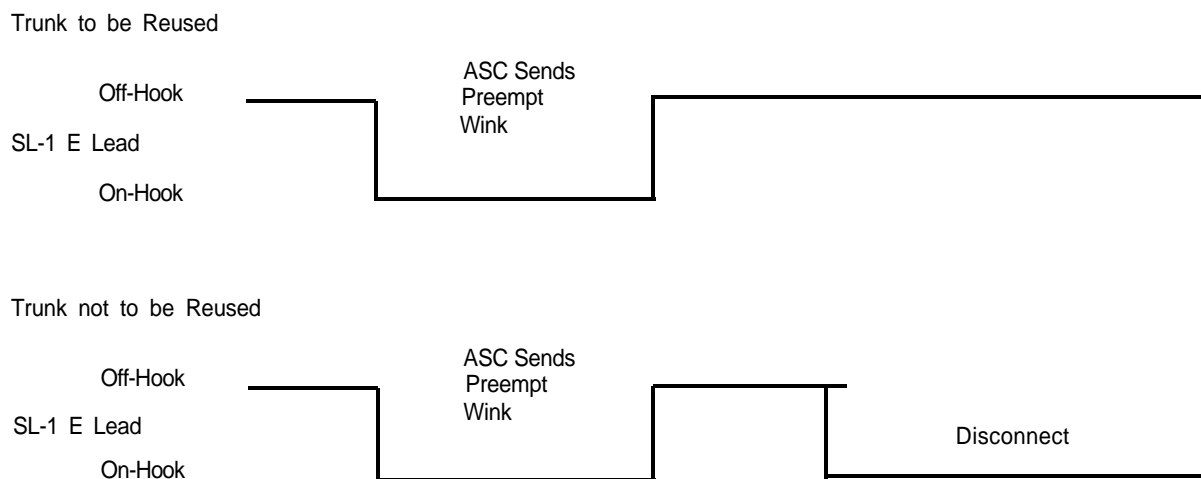


Fig. 2-4
ASC Preempt sequence: ASC Off-Hook

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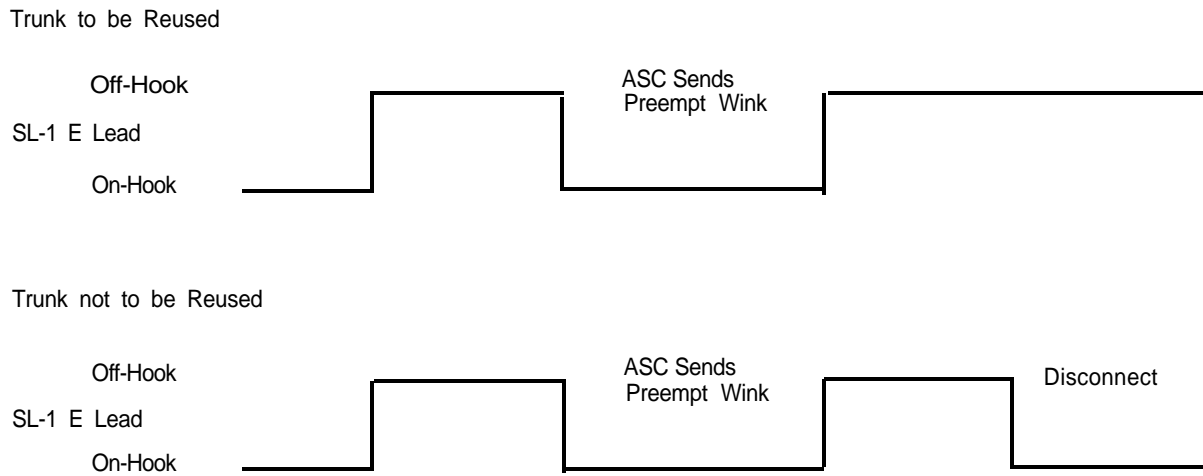


Fig. 2-5
ASC Preempt Sequence: ASC On-Hook

DISCONNECT
SIGNALING

2.17 *SL-1 Disconnect.*

- (1) A station goes On-Hook. The **SL-1** transmits On-Hook to the ASC and sets the Disconnect Supervision Timer (default 35 s).
- (2) The ASC responds by going On-Hook for at least a valid disconnect period.
- (3) The SL-1 times for at least 700 ms before reseizure may start.

2.18 *ASC Disconnect.*

- (1) The ASC goes On-Hook for at least a valid disconnect period.
- (2) **The SL-1** goes On-Hook and times for 800 ms before reseizure is allowed.

Note 1: A 'Valid disconnect period' is determined by the programmable timers ICF (incoming calls) or OGF (outgoing calls), as appropriate.

Note 2: The Guard Interval Timers for SL-1 disconnect (700 ms) or ASC disconnect (800 ms) are programmable to accomodate satellite facilities. The 700 ms and 800 ms values are nominal, real values being 640-1536 ms for SL-1 disconnect and 768-1664 ms for ASC disconnect (both are programmable in 128 ms steps). For both, the lowest values are the system defaults.

3. AUTOVON TONES

3.01 An **SL-1** equipped with the AUTOVON feature can produce all the tones required by AUTOVON, as well as all other tones used by North American Networks.

TONES GENERATED

3.02 Table 3-A gives the DTMF tones generated by the SL-1 QPC251 Flexible Tone and Digit Switch (TDS) and transmitted by the QPC237 4-wire trunk or the QPC71 %-wire trunk.

Table 3-A
DTMF SIGNALS PROVIDED BY SL-1

DTMF SIGNAL	FREQUENCY (Hz) ($\pm 1/32$)	NOTES
1	697 + 1209	
2	697 + 1336	
3	697 + 1477	
4	770 + 1209	
5	770 + 1336	
6	770 + 1477	
7	852 + 1209	
8	852 + 1336	
9	852 + 1477	
*	941 + 1209	1
0	941 + 1336	
#	941 + 1477	2
FO	697 + 1633	3
F		3
I	850 + 1633	3
P	941 + 1633	3

Note 1: This signal is not presently used.

Note 2: The SL-1 interprets this signal as an end-of-dialing indicator.

Note 3: Legend:

- FO = Flash Override
 - F = Flash
 - I = Immediate
 - P = Priority
-

3.03 Table 3-B gives the voice frequency tones generated by the SL-1 flexible TDS and transmitted by the SL-1 QPC237 4-wire trunk or the QPC71 2-wire trunk.

RECEIVING DATA

3.04 The Digitone Receiver (DTR) of the SL-1 will respond to tones and frequencies defined for AUTOVON and North American signaling that meet the following criteria.

3.05 *Signal Level.* Signal level can be over a range of + 5 to -22 dB V per frequency (across 600 Ω) with a maximum difference of 4 dB between the frequency pair.

3.06 *Signal Duration.* The two frequencies must be present for at least a 40 ms period between successive signals.

3.07 *Signal to Noise Ratio.* The DTR operates satisfactorily with a signal to noise ratio of 22 dB per frequency (thermal noise flat weighting) with the minimum allowable received level and maximum allowable frequency error.

3.08 *Dial Pulse.* DP signals must be 9.5 to 10.5 pps at 57% break.

ERROR HANDLING AND DIAGNOSTICS

3.09 The timers, signal checking, and maintenance diagnostic programs provided by SL-1 ensure that signaling malfunctions are detected and appropriate action (Such as ignoring the erroneous signal, automatic testing of the circuit if required, and/or printing an error message on the TTY) is taken.

3.10 *False Seizures.* E-lead seizures of shorter duration than 150 ms are ignored by SL-1.

Table 3-B
VOICE FREQUENCY SIGNALS PROVIDED BY SL-1

SIGNAL	FREQUENCY (Hz)	COMPOSITE LEVEL (dBm0)	INTERRUPTION
Dial Tone	350 + 440	- 1 0	Continuous
Busy Tone	480 + 620	- 2 1	60 ipm 0.5 s on 0.5 s off
Busy Tone Trunk	480 + 620	- 2 1	120 ipm 0.2 s on 0.3 s off
Reorder Tone	480 + 620	- 2 1	120 ipm 0.2 s on 0.3 s off
Routine Ring Tone (Ring Back)	440 + 480	- 1 3	10 ipm 2.0 s on 4.0 s off
Precedence Ring Tone (Ring Back)	440 + 480	- 1 3	30 ipm 1.64 s o 0.36 s off
Preempt Tone	440 + 620	- 1 5	Continuous

3.11 **Glare (Incoming Calls)**. Once a valid seizure is detected, the SL-1 marks the trunk busy to outgoing calls within 100 ms of the seizure. (This is normally done immediately.)

3.12 **Glare (Outgoing Calls)**. Once an M-lead seizure (request for service) is initiated by SL-1, any E-lead response within the glare timing interval is considered glare. If glare is detected, the SL-1 responds by releasing the connection and resending the outgoing call on another trunk. If the E-lead response persists beyond 150 ms, SL-1 assumes a valid incoming request for service and sends a wink start signal to AUTOVON.

3.13 **Stuck Senders or Insufficient Digits (Incoming calls only)**. If after receiving a wink start signal:

- no digits are received in 5 s (the 5-s timer is programmable),

or

- the interdigital interval exceeds 5 s.

3.14 The SL-1 responds by returning an off-hook signal of 3 s duration followed by on-hook (disconnect). If the ASC does not disconnect within a additional 15 s, a maintenance message is printed on the teletype and a minor alarm raised on the attendant console. If the ASC does disconnect the trunk within the 15 s, the trunk is idled in the normal way with no message ensuing.

3.15 **Wink Start Timeout (Outgoing Calls Only)**. On outgoing calls originated from SL-1, failure to receive a wink or start dial signal from the ASC within 5 s causes SL-1 to increment threshold measurement. In this case, supervision by the originating party determines disconnect. The trunk is held busy by the SL-1 for 30 s, during which time SL-1 makes no attempt to re-seize it. At the end of the 30 s, it is placed back into service.

3.16 **Preemption**. Preemption by the ASC can occur at any time other than when the trunk is idle. Thus SL-1 is prepared to detect the preemption wink signal on incoming calls prior to outpulsing by the ASC, during the interval between digits, or any time after outpulsing is complete (before or after the station answers). On outgoing calls from SL-1, the preemption signal can occur at any time after the ASC recognizes request for service and before the ASC disconnects upon completion of calls.

3.17 Once the preempt signal is received (i.e., the ASC remains off-hook following preempt wink), and SL-1 responds by returning a wink start signal, the trunk is held busy to outgoing calls to permit the incoming precedence call to occur. If in this case, a new call originated by the ASC is not received within 5 s, SL-1 provides treatment similar to that for the stuck sender time-out.

3.18 **Disconnect Timeout**. Once the SL-1 disconnects from a call, the ASC should return on-hook supervision. If this does not occur within 35 s, SL-1 prints a maintenance message and raises a minor alarm.

3.19 Due to the purpose of the AUTOVON system and few number of trunks to each SL-1, any failure of an AUTOVON trunk constitutes a serious degradation in communication capability between the SL-1 and the ASC. Therefore, for any AUTOVON trunk fault, a minor alarm is raised on the attendant console, without invoking emergency transfer.

3.20 *Input/Output.* AUTOVON input and output messages applying to the SL-1 maintenance diagnostics can be found in the Maintenance Diagnostic practice for the applicable **SL-1** system.

4. MEMORY AND REAL TIME

Memory

4.01 The following is a summary of the program memory estimate for the AUTOVON feature package of Generic XI 1 Release 4:

FEATURE	STORAGE (Words)
AUTOVON Package	16000
Conversion (XI 1.3 to XI 1.4)	200
Data Dump	200
System Load	200

4.02 Protected Data Store.

1 word per protected Route Data Block

1 word per protected Customer Data Block.

4.03 Unprotected Data Store.

3 words per unprotected AUTOVON trunk block

2 words per call register

3 words per unprotected attendant block

1 word per unprotected station block.

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

**AUTOVON FEATURE (GENERIC X11)
 ATTENDANT CONSOLE AND TELEPHONE SET
 OPERATION**

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* SL-1 is a trademark of Northern Telecom Limited

1. GENERAL

1.01 The Automatic Voice Network (AUTOVON) is the name given to the major non-secure military telephone network in the U.S.A. It is linked to an equivalent Canadian network, the Canadian Switched Network (CSN). Both of these networks are completely separate from the public network and have their own numbering plans and special calling capabilities.

1.02 The SL-1 fills the role of an AUTOVON Private Branch Exchange (PBX) by having one or more trunk routes between an AUTOVON Switching Center (ASC) in the AUTOVON network and the SL-1. This practice describes the procedures used to place and handle various types of calls to and from the AUTOVON network, using attendant consoles, SL-1 sets, and 500/2500 sets.

Note: This practice describes the AUTOVON feature for Generic X11 Release 4. Procedures for the Generic X14 AUTOVON feature can be found in 553-2661-300.

1.03 Reference Documentation. This practice describes only those types of operations that are unique to, or different for, AUTOVON as opposed to normal operation. Operation of features not covered in this practice can be found in the operation and test practices for the relevant SL-1 system.

- (b) *AUTOVON Routine (AVR)*. Indicates that an incoming AUTOVON routine call is in the queue.
- (c) *Precedence Intercept (PIN)*. An incoming precedence call is routed to the attendant queue and causes this lamp to light when it encounters any of the following:
 - a busy station (after hunting)
 - | a vacant number
 - no answer within a specified (i.e., programmable) time period
 - | a maintenance busy condition
 - network blocking
 - misoperation of call modification feature.

Precedence Keys

2.08 In addition to the operating keys and lamps provided for conventional attendant call-processing, the following keys are required to process AUTOVON calls. Four keys corresponding to the Priority (P), Immediate (I), Flash (F) and Flash Override (FO) precedence levels augment the dial pad and permit the attendant to place outgoing precedence calls. The attendant selects one of these keys during the dialing sequence to place a precedence call. The precedence level functions can be assigned to any operating key having flexible assignment capability. The attendant may also make or extend precedence calls by dialing the SPRE code corresponding to the appropriate precedence digit.

Note: Lamps associated with precedence keys P, I, F, and FO do not light when the key is depressed.

Call Waiting (CW) Lamp

2.09 This lamp is located to the right of the NIGHT key and flashes at 60 ipm whenever there is a precedence call waiting in the attendant queue. The tone ringer sounds when a precedence call arrives in the queue, and continues to sound until the precedence call is answered.

OPERATING PROCEDURES

2.10 All calls except those originating or terminating on AUTOVON trunks are processed using conventional **SL-1** procedures described in the console operating manuals provided with the **SL-1**.

2.11 AUTOVON calls require the modified procedures described in the following charts. Routine or higher precedence outgoing calls may be placed by the attendant, although routine outgoing calls are normally dialed directly by station users.

2.12 Incoming calls are shown on the ICI keys, of the attendant console and are automatically presented to a LPK key (if one is available).

2.13 Routine incoming calls are presented to the loop keys in order of arrival. Incoming precedence calls are presented to the loop keys in order of precedence, highest to lowest.

2.14 Routine outgoing calls class-marked for access to outgoing trunks normally bypass the attendant. In cases where the class of service of a station dictates attendant assistance, the procedure is the same as for non-AUTOVON calls. Precedence calls bypass the attendant if the station is allowed precedence outward dialing. If not, the attendant must extend the call (if the call is permitted by COS and TGAR restrictions).

How to Use the Charts

2.15 Each type of call is processed in steps. Reside each step is given the appropriate action in the Action column. In the Verification column is given the various indicators that the step was performed properly and the system is operating correctly.

Chart 2-1
INCOMING ROUTINE CALLS

STEP	ACTION	VERIFICATION
1	Signal from incoming trunk.	Tone ringer sounds. SRC lamp flashes at 60 ipm. If call is presented to a LPK, LPK flashes at 30 ipm and digit display shows trunk access code and member number. ICI (AVR) is lit steadily.
2	Press LPK key.	Tone ringing stops, 2-way conversation. SRC lamp steadily lit.
3	To terminate call, go directly to Step 5. To extend the call to a station, key in the station DN on the key pad.	Ringback is heard, calling party excluded during dialing. Digit display shows the dialed DN. EXC SRC lamp lit during dialing. DEST lamp flashes at 30 ipm until called party answers.
4	Called station answers (see Note).	Ringback stops, 3-way conversation. DEST lamp steadily lit.
5	Press RLS key.	RLS lamp lights, all other lamps used in the call go out. Digit display blanked.
<p>Note: Attendant may release before called station answers by pressing RLS key. Call will complete if called station answers. If called station does not answer within the recall period, the call is presented to the attendant on the ATT RECALL ICI.</p>		

Chart 2-2
INCOMING PRECEDENCE CALLS

STEP	ACTION	VERIFICATION
1	Signal from incoming trunk.	Tone ringer sounds. SRC lamp flashes at 120 ipm. CW lamp flashes at 60 ipm. If call is presented to a LPK, LPK flashes at 30 ipm and digit display shows precedence code (Note 1), trunk access code and member number. ICI (PRC) is lit steadily.
2	Press LPK key.	Tone ringing stops, 2-way conversation. SRC lamp steadily lit.
3	To terminate call, go directly to Step 5. To extend the call to a station, key in the station DN on the key pad.	Ringback is heard, calling party excluded during dialing. Digit display shows the dialed DN. EXC SRC lamp lit during dialing. DEST lamp flashes at 30 ipm until called party answers.
4	Called station answers (see Note 2).	Ringback stops, 3-way conversation. DEST lamp steadily lit.
5	Press RLS key.	RLS lamp lights, all other lamps used in the call go out. Digit display blanked.
<p><i>Note 1:</i> The precedence digit is shown on the digit display as one of 0 to 3 where 0=FO, 1=F, 2=I, 3=P.</p>		
<p><i>Note 2:</i> Attendant may release before called station answers by pressing RLS key. Call will complete if called station answers. If called station does not answer within the recall period, the call is presented to the attendant on the PIN ICI.</p>		

Chart 2-3
INCOMING PRECEDENCE INTERCEPT CALL

STEP	ACTION	VERIFICATION
1	Recall presented to Attendant.	Tone ringer sounds. SRC and DEST lamps flash at 120 ipm. CW lamp flashes at 60 ipm. If call is presented to a LPK, LPK flashes at 30 ipm and digit display shows originally dialed digits (Note 1). ICI (PIN) is lit.
2	Press LPK key.	Tone ringing stops, 2-way conversation. SRC lamp steadily lit (Note 2).
3	To terminate call, go directly to Step 5. To extend the call to a station, key in the station DN on the key pad.	Ringback is heard, calling party excluded during dialing. Digit display shows the dialed DN. EXC SRC lamp lit during dialing. DEST lamp flashes at 30 ipm until called party answers.
4	Called station answers (see Note 3).	Ringback steps, 3-way conversation. DEST lamp steadily lit.
5	Press RLS key.	RLS lamp lights, all other lamps used in the call go out. Digit display blanked.
<p><i>Note 1:</i> The precedence digit is shown on the digit display as one of 0 to 3 where 0=FO, 1=F, 2=I, 3=F.</p> <p><i>Note 2:</i> Press the Display Source key to see the precedence, trunk access code, and member number</p> <p><i>Note 3:</i> Attendant may release before called station answers by pressing RLS key. Call will complete if called station answers. If called station does not answer within the recall period, the call is presented to the attendant on the PIN ICI.</p>		

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Chart 2-4
PRECEDENCE CALL COMPLETION TO BUSY

STEP	ACTION	VERIFICATION
1	Incoming call to attendant.	Tone ringer sounds. If call is presented to LPK, LPK, and SRC flashes at 60 ipm. ICI (PIN) is lit.
2	Press LPK key.	2-way conversation with calling party. SRC lamp steadily lit. Digit display shows precedence level (Note), trunk access code and member number of incoming call.
3	Dial DN of busy station.	If precedence level of incoming call is higher than that of the call currently in progress on busy station, barge-in tone is heard, and 3-way conversation. DEST lamp lit steadily. Digit display shows dialed DN. If precedence level of incoming call is lower than or equal to that of call currently in progress on the busy station, overflow tone is heard.
4	Busy station goes on-hook.	2-way conversation. At this point the formerly busy station is rung automatically. Ring back tone is heard, DEST lamp flashes at 60 ipm. When called station answers, 3-way conversation.
5	Press RLS key.	Incoming call remains in conversation with station. Console is idle.

Note: The precedence digit is shown on the digit display as one of 0 to 3 where 0=F0, 1=F, 2=I, 3=P.

Chart 2-5
EXTENDING A ROUTINE CALL

STEP	ACTION	VERIFICATION
1	Call from SL-1 station.	Tone ringer sounds. SRC lamp flashes at 60 ipm. If call is presented to a LPK, LPK flashes at 30 ipm and digit display shows DN of calling station. ICI (DIAL 0) is lit steadily.
2	Press LPK key.	Tone ringing stops, 2-way conversation. SRC lamp steadily lit.
3	Enter AUTOVON routine trunk access code at key pad.	Source party excluded during dialing, EXC SRC lamp lit during dialing. Dial tone heard from trunk, DEST lamp steadily lit.
4	Enter AUTOVON extension number required by calling party.	Ringback heard. Called party answers. 3-way conversation.
5	Press RLS key.	Console idle.

Chart 2-6
EXTENDING A PRECEDENCE CALL

STEP	ACTION	VERIFICATION
1	Call from SL-1 station.	Tone ringer sounds. SRC lamp flashes at 60 ipm. If call is presented to a LPK key, LPK flashes at 30 ipm and digit display shows DN of calling party. ICI (DIAL 0) is lit steadily.
2	Press LPK key.	Tone ringing stops, 2-way conversation. SRC lamp steadily lit.
3	Enter AUTOVON precedence code (SPRE code, Note 1).	Source Party excluded EXC SRC lamp steadily lit. Digit display shows precedence digit (Note 2).
	or	
	Press feature key for precedence level.	
4	Enter AUTOVON trunk access code on the key pad (Note 3).	Source party excluded, EXC SRC lamp steadily lit during display. Digit Display shows dialed digits. Dial tone from trunk, DEST lamp steadily lit. (If overflow tone is heard, go to Chart 2-7.)

Chart continued —

PRACTICE 553-2661-301

Chart 2-6 Continued
EXTENDING A PRECEDENCE CALL

STEP	ACTION	VERIFICATION
5	Enter AUTOVON network number at key pad.	Ringback heard. When called party answers, 3-way conversation.
6	Press RLS key.	Console idle.

Note 1: SPRE = Special Function Prefix. Code is one of: FO= SPRE+ 8-t-2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0.

Note 2: Precedence digits are shown on the digit display as one of 0 to 3 where 0=FO, 1=F, 2=I, 3=P.

Note 3: The AUTOVON trunk access code may be entered before or after the precedence digit.

Chart 2-7
PROMOTION OF A CALL

Note: Promotion is used to increase the precedence level of a call when all trunks in a route are busy. In this case it probably means that a call will be preempted. Promotion can only be done on preemptable trunk routes.

STEP	ACTION	VERIFICATION
1	Overflow tone is heard after dialing trunk access code of preemptable trunk route.	DEST lamp flashes at 30 ipm. 2-way conversation with calling party.
2	Press feature key for a higher precedence level.	Overflow tone steps. Calling party excluded. Digit display shows new precedence level (Note). If dial tone is heard, promotion was successful. DEST lamp steadily lit. Call may proceed normally. If overflow tone is heard again, promotion was not successful. Repeat Step 2 using a higher precedence.

Note: Precedence levels are shown as one of 0 to 3 where 0=FO, 1 =F, 2=I, 3=P.

Chart 2-8
DSA RECALL

Note: Occasionally a station user engaged in an AUTOVON call may want to recall the Dial Service Assistance (DSA) operator. This can be done via the attendant.

STEP	ACTION	VERIFICATION
1	Recall presented to attendant.	Tone ringer sounds. If call is presented to LPK, LPK, SRC, and DEST flash at 60 ipm. ICI (ATT RECALL) is lit.
2	Press LPK key.	2-way conversation with calling party. SRC lamp steadily lit. Digit display shows DN of calling party.
3	Press SIG DEST key.	Ringback tone heard. Calling party excluded. EXC SRC lamp steadily lit. When DSA operator answers, 3-way conversation.
4	Press RLS key.	Calling party remains in conversation with DSA operator, but attendant console is idle.

Chart 2-9
AUTODIAL PRECEDENCE CALL STORAGE

Note 1: A Routine AUTOVON call can be stored in Autodial and used in the same way as TIE trunk calls. See 553-2001-300.

Note 2: For precedence calls, the precedence digit can be stored with the Autodial number and thus called up automatically when Autodial is invoked. The number can be stored without the precedence digit, but in this case the call is sent out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP	ACTION	VERIFICATION
STORAGE (With Precedence Digit)		
1	Press RLS key to ensure that the console is idle.	Console idle, all LPK extinguished.
2	Press AUTODIAL key.	AUTODIAL lamp flashes at 60 ipm.
3	Press Display Source key.	Number currently stored against AUTODIAL key is displayed.
4	Dial SPRE + precedence code (Note 1).	Digit display shows entered precedence level (Note 2).

Chart continued ➡

Chart 2-9 Continued

AUTODIAL PRECEDENCE CALL STORAGE

STEP	ACTION	VERIFICATION
5	Dial the trunk access code.	Dialed number appears on digit display.
6	Dial AUTOVON network number.	Dialed number appears on digit display.
7	Press AUTODIAL key again.	AUTODIAL lamp extinguished. Digit display is cleared. Number is now stored in Autodial.
VERIFICATION OF STORED NUMBER		
8	Press RLS key to ensure that console is idle.	Console idle, all LPK unlit.
9	Press AUTODIAL key.	AUTODIAL lamp flashes at 60 ipm.
10	Press Display Source key.	The number stored with the AUTODIAL key appears on the digit display (Note 2).
11	Press AUTODIAL key.	AUTODIAL lamp is extinguished, digit display is cleared. Console is idle.

Note 1: SPRE = Special Function Prefix. Codes are as follows: FO= **SPRE**+ 8+2, **F**=**SPRE**+8+5, **I**=**SPRE**+8+8, **P**=**SPRE**+8+0. Precedence feature keys cannot be stored in Autodial.

Note 2: The precedence digit is displayed on the digit display as follows: 0= FO, 1 =**F**, 2=**I**, 3=**P**.

Note 3: ***** when entered in **Autodial** denotes a 3-second pause in digit outputting. This gives a trunk access code time to result in a trunk seizure before continuing with outputting.

Chart **2-10**
 PLACING A PRECEDENCE CALL USING **AUTODIAL**

Note 1: If the precedence digit is stored as part of the **Autodial** number, pressing the **AUTODIAL** key results in the outpulsing of the precedence digit. If the precedence digit is not stored as part of the **Autodial** number, the call is sent out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP ACTION	VERIFICATION
IF PRECEDENCE DIGIT IS STORED IN AUTODIAL	
1 Station dials attendant.	Tone ringer sounds. If call is presented to LPK, LPK and SRC lamps flash at 60 rpm, and Digit display shows DN of calling station. ICI (DIAL 0) is lit.
2 Press an idle LPK key.	Tone ringing stops. 2-way conversation with calling party. SRC lamp is steadily lit. Digit display shows DN of calling party (Note).
3 Press AUTODIAL key.	Source excluded during dialing. EXC SRC lamp is steadily lit. DEST lamp steadily lit. Digit display shows number stored against AUTODIAL key. When outpulsing is finished, ringback is heard. Called party answers, 3-way conversation.
4 Press RLS key.	Calling party remains in conversation with destination, but console is idle.
<i>Note:</i> The precedence digit is displayed on the digit display as follows: 0= FO , 1 = F , 2= I , 3= P .	

Chart 2-11
SPEED CALL PRECEDENCE CALL STORAGE

Note 1: Routine AUTOVON calls can be stored in Speed Call and used in the same way as TIE trunk calls. See 553-2001-300.

Note 2: For precedence calls, the precedence digit can be stored with the Speed Call number and thus called up automatically when Speed Call is invoked. The number can be stored without the precedence digit, but in this case the call is sent out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP ACTION	VERIFICATION
STORAGE (With Precedence Digit)	
1 Press RLS key to ensure that the console is idle.	Console idle, all LPK unlit.
2 Press SPEED CALL key.	SPEED CALL lamp flashes at 30 ipm.
3 Dial the one- to three-digit code to be associated with the number, then press the Display Source key.	Number currently stored against SPEED CALL key is displayed.
4 Dial SPRE + precedence code (Note 1).	Digit display shows entered precedence level (Note 2).
5 Dial trunk access code.	Dialed number appears on digit display.
6 Dial AUTOVON network number.	Dialed number appears on digit display.
7 Press SPEED CALL key again.	SPEED CALL lamp extinguished. Digit display is cleared. Number is now stored in Speed Call.
VERIFICATION OF STORED NUMBER	
8 Press RLS key to ensure that console is idle.	Console idle, all LPK extinguished.
9 Press SPEED CALL key.	SPEED CALL lamp flashes at 30 ipm.
10 Dial the one- to three-digit code associated with the number, then press the Display Source key.	The number stored against the Speed Call code appears on the digit display.
11 Press SPEED CALL key.	SPEED CALL lamp is extinguished, digit display is cleared. Console is idle.

Note 1: SPRE = Special Function Prefix. Codes are as follows: FO=SPRE+8+2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0. Precedence feature keys cannot be stored.

Chart continued

Chart 2-11 Continued
SPEED CALL PRECEDENCE CALL STORAGE

STEP ACTION	VERIFICATION
<p>Note 2: The precedence digit is displayed on the digit display as follows: 0=FO, 1= F, 2=I, 3=P.</p> <p><i>Note jr:</i> ‘*’ when entered in Speed Call denotes a 3-second pause in digit outpulsing. This gives a trunk access code time to result in a trunk seizure before continuing with outpulsing.</p>	

Chart 2-12
PLACING A PRECEDENCE CALL USING SPEED CALL

Note: If the precedence digit is stored as part of the Speed Call number, pressing the SPEED CALL key results in the outpulsing of the precedence digit. If the precedence digit is not stored as part of the Speed Call number, the call goes out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP ACTION	VERIFICATION
<p>IF PRECEDENCE DIGIT IS STORED IN SPEED CALL</p>	
1 Station dials attendant.	Tone ringer sounds. If call is presented to LPK, LPK and SRC lamps flash at 60 ipm, and Digit display shows DN of calling station. Otherwise, ICI (DIAL 0) is lit.
2 Press an idle LPK key.	Tone ringing stops. 2-way conversation with calling party. SRC lamp is steadily lit. Digit display shows DN of calling party.
3 Press the SPEED CALL key, then dial the one- to three-digit speed call code for the desired number.	Source excluded during dialing. EXC SRC lamp is steadily lit. DEST lamp steadily lit. Digit display shows the speed call code entered (Note 1). When outpulsing is finished, ringback is heard. Called party answers. 3-way conversation.
4 Press RLS key.	Calling party remains in conversation with destination but console is idle.
<p>Note 1: The digits stored against the dialed Speed Call code are not displayed unless Display Source or Display Destination key is pressed.</p> <p>Note 2: The precedence digit is displayed on the digit display as follows: 0=FO, 1 =F, 2=I, 3=P.</p>	

3. STATION FEATURES

3.01 The AUTOVON feature of Generic X1 1 Release 4 enables station users to place precedence calls directly from a station set. This can be done from any SL-1, NE-500 or NE-2500 set which:

- has the required Network Class of Service (NCOS) value for the precedence level to be used, and
- | which is not barred from AUTOVON trunks by the restrictions of COS, TGAR, etc.

3.02 A precedence digit may be dialed using a Special Function Prefix (SPRE) and the appropriate 2-digit code for the required precedence level. This can be done at any station set; however, SL-1 sets can define feature keys to function as the precedence digit keys.

3.03 Precedence calls may be dialed directly, but they can also be made using the following features:

- Authorization *Code*. The Basic Authorization Code (BAUT) feature of Generic X1 1 may be used to access AUTOVON and the required precedence level.
- | *Autodial and Speed Call*. The precedence digits must be stored in the SPRE format. Precedence keys cannot be used to store precedence digits.

PRACTICE 553-2661-301

Chart 3-1
STATION PRECEDENCE CALL PLACEMENT

Note: Routine-level calls over AUTOVON trunks are made in the same way as normal TIE trunk calls.

STEP	ACTION	VERIFICATION
SL-1 SETS		
1	Go off-hook or press DN key.	Dial tone is heard. DN lamp is lit.
2	Enter SPRE + precedence code (Note 1). or Press feature key for desired precedence level.	Dial tone stops. Digit display shows precedence digit (Note 2).
3	Enter trunk access code for preemptable AUTOVON trunk route.	Digit display shows dialed digits. Dial tone heard from trunk.
4	Enter AUTOVON network number.	Dial tone stops. Ringback is heard. When called party answers, 2-way conversation.
5	Press RLS key or go on-hook.	Set is idle.
500/2500 SETS		
1	Go off-hook.	Dial tone is heard.
2	Dial SPRE + precedence code (Note 1).	Dial tone stops.
3	Dial trunk access code for preemptable AUTOVON trunk route.	Dial tone heard from trunk.
4	Dial the desired AUTOVON network number.	Dial tone stops. Ringback is heard. When called party answers, 2-way conversation.
5	Go on-hook.	Set is idle.
<p>Note 1: SPRE = Special Function Prefix. Code can be one of: FO= SPRE+ 8+2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0.</p> <p>Note 2: Precedence levels are shown on the digit display as follows: 0=FO, 1 =F, 2= I, 3=P.</p>		

Chart 3-2
CALL PROMOTION


Note 1: Promotion is used to increase the precedence level of a call when all trunks in an AUTOVON preemptable route are busy. In that case, it probably means that another call will be preempted.

Note 2: Because Promotion requires the use of precedence feature keys, AUTOVON calls from 500/2500 sets cannot be promoted.

STEP	ACTION	VERIFICATION
SL-1 SETS		
1	Dial access code of preemptable AUTOVON trunk route.	Digit display shows dialed digits. Overflow tone is heard.
2	Press feature key for a higher precedence level.	Digit display shows dialed digits. Overflow tone stops. If dial tone is heard, promotion was successful. Call may proceed as normal. If overflow tone is heard again, promotion was not successful. Repeat this step using a higher precedence level.

Chart 3-3
PRECEDENCE CALL USING AUTHCODE

STEP	ACTION	VERIFICATION
SL-1 SETS		
1	Go off-hook or press DN key.	Dial tone heard, DN lamp lights steadily.
2	Dial SPRE+6.	Digit display shows dialed digits. Dial tone stops.
3	Dial a valid authorization code that has access to the AUTOVON trunk route and the desired precedence level.	Digit display shows dialed digits. If authorization code is valid, dial tone is heard again. (If authorization code is not valid, overflow tone is heard. Go on-hook and try again.)
4	Dial SPRE + precedence code (Note).	Digit display shows dialed digits.
	<i>or</i>	
	Press the precedence feature key, if equipped.	
5	Dial the AUTOVON trunk access code.	Digit display shows dialed digits. Dial tone from trunk heard.

Chart continued 

PRACTICE 553-2661-301

Chart 3-3 Continued
PRECEDENCE CALL USING AUTHCODE

STEP	ACTION	VERIFICATION
6	Dial the AUTOVON network number.	Ringback is heard. When called party answers, 2-way conversation.
7	Press RLS key.	Set idle, call released.

500/2500 SETS

1	Go off-hook.	Dial tone heard.
2	Dial SPRE + 6.	Dial tone stops.
3	Dial a valid authorization code that has access to the AUTOVON trunk route and the desired precedence level.	If authorization code is valid, dial tone is heard again. (If authorization code is not valid, overflow tone is heard. Go on-hook and try again.)
4	Dial the precedence SPRE code (Note).	Dial tone stops.
5	Dial the AUTOVON trunk access code.	Dial tone from trunk heard.
6	Dial the AUTOVON network number.	Ringback is heard. When called party answers, 2-way conversation.
7	Go on-hook.	Set idle.

Note: SPRE = Special Function Prefix. Code is one of: FO=SPRE+8+2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0.

Chart 3-4

AUTODIAL PRECEDENCE CALL STORAGE

Note 1: Routine AUTOVON calls can be stored in **Autodial** and used in the same way as TIE trunk calls. See 553-2001-300.

Note 2: For precedence calls, the precedence digit can be stored with the **Autodial** number and then called up automatically when **Autodial** is invoked. The number can be stored without the precedence digit, however in this case the call goes out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP	ACTION	VERIFICATION
STORING A PRECEDENCE CALL		
1	Press AUTODIAL key (with handset on-hook).	AUTODIAL lamp flashes at 60 ipm. Number currently stored against AUTODIAL key is displayed.
2	Dial SPRE + precedence code (Note 1).	Digit display shows entered precedence level (Note 2).
3	Dial trunk access code.	Dialed number appears on digit display.
4	Dial AUTOVON network number.	Dialed number appears on digit display.
5	Press AUTODIAL key again.	AUTODIAL lamp is extinguished. Digit display is cleared. Number is now stored in Autodial .
VERIFICATION OF STORED NUMBER		
1	Press AUTODIAL key (with handset on-hook).	AUTODIAL lamp flashes at 60 ipm. The number stored with the AUTODIAL key appears on the digit display.
2	Press AUTODIAL key.	AUTODIAL lamp is extinguished, digit display is cleared. Set is idle.

Note 1: SPRE = Special Function Prefix. Codes are as follows: **FO**=SPRE+8+2, **F**=SPRE+8+5, **I**=SPRE+8+8, **P**=SPRE+8+0. Precedence feature keys cannot be stored.

Note 2: The precedence digit is displayed on the digit display as follows: 0=**FO**, 1=**F**, 2=**I**, 3=**P**.

Note 3: '**' when entered in **AUTODIAL** denotes a 3-second pause in digit outputting. This gives a trunk access code time to result in a trunk seizure before continuing with outputting.

PRACTICE 553-2661-301

Chart 3-5
PLACING A PRECEDENCE CALL USING AUTODIAL

Note: If the precedence digit is stored as part of the Autodial number, pressing the AUTODIAL key would result in the outpulsing of the precedence digit. If the precedence digit is not stored as part of the Autodial number, the call goes out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP ACTION

VERIFICATION

IF PRECEDENCE DIGIT IS STORED IN AUTODIAL

- | | | |
|---|------------------------------|---|
| 1 | Go off-hook or press DN key. | Dial tone is heard, DN lamp is steadily lit. |
| 2 | Press AUTODIAL key. | Dial tone stops. Stored number is shown on digit display. If call is successful, ringback tone is heard. If unsuccessful, overflow tone is heard. |
| 3 | Press RLS key or go on-hook. | Set is idle. |
-

Note 1: SPRE = Special Function Prefix. Code may be one of: FO=SPRE+8+2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0.

Note 2: The precedence digit is displayed on the digit display as follows: 0= FO, 1 = F, 2=I, 3=P.

Note 3: If overflow tone is heard, the call may be promoted as described in Chart 3-2.

Chart 3-6
SPEED CALL PRECEDENCE CALL STORAGE

Note 1: Routine AUTOVON calls can be stored in Speed Call and used in the same way as for TIE trunk calls.

Note 2: For precedence calls, the precedence digit can be stored with the speed call number and thus called up automatically when Speed Call is invoked. The number can be stored without the precedence digit, however in this case, the call is sent out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP ACTION

VERIFICATION

SL-1 SETS

- | | | |
|---|--|------------------------------------|
| 1 | Go on-hook and press the RLS key to ensure that the set is idle. | RLS lamp lit, set is idle. |
| 2 | Press the SPEED CALL key. | SPEED CALL lamp flashes at 30 ipm. |
| 3 | Dial the one- to three-digit code to be associated with the dialed digits. | |
-

Chart continued ---

Chart 3-6 Continued
SPEED CALL PRECEDENCE CALL STORAGE

STEP	ACTION	VERIFICATION
4	Dial the precedence SPRE code to be associated with the number (Note 1).	Digit Display shows precedence digit (Note 2).
5	Dial the trunk access code (Note 3).	
6	Dial the AUTOVON network number.	
7	Press the SPEED CALL key again.	SPEED CALL lamp is extinguished. Number is stored in Speed Call list. Set is idle.
500/2500 SETS		
1	Lift handset.	Dial tone.
2	Dial system prefix (#).	Dial tone removed.
3	Dial access code (2).	Dial tone (overflow tone if code is not allowed).
4	Dial speed call entry number (1-3 digits).	
5	Dial SPRE + precedence code (Note 1).	
6	Dial trunk access code (Note 3).	
7	Dial AUTOVON network number.	
8	Go on-hook.	Set idle. Number is stored in speed call list.

Note 1: SPRE = Special Function Prefix. Code may be one of: FO=SPRE+8+2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0.

Note 2: The precedence digit is displayed on the digit display as follows: 0=FO, 1 =F, 2=I, 3=P.

Note 3: An asterisk '*' stored as part of a Speed Call list entry results in a three-second pause at that point when the number is called up from the Speed Call list. This allows time for such things as seizure of trunks and returning of dial tone from trunks before outpulsing the next set of numbers.

Chart 3-7
 PLACING A PRECEDENCE CALL USING SPEED CALL (SL-1 Sets)

Note: If the precedence digit is stored as part of the Speed Call number, pressing the SPEED CALL key results in the outpulsing of the precedence digit. If the precedence digit is not stored as part of the Speed Call number, the call goes out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP	ACTION	VERIFICATION
IF PRECEDENCE DIGIT IS STORED IN SPEED CALL		
1	Go off-hook or press DN key.	Dial tone heard, DN lamp lights steadily.
2	Press SPEED CALL key.	Dial tone stops. SPEED CALL lamp lights steadily.
3	Dial the one- to three-digit speed call code for the desired number.	Desired number is outpulsed by system, SPEED CALL lamp is extinguished. Digit Display shows dialed digits. If call is successfully completed, ringback is heard. When called party answers, 2-way conversation.
4	Press RLS key.	Set idle.

Note : SPRE = Special Function Prefix. Code can be one of: FO=SPRE+8+2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0.

Chart 3-8

PLACING A PRECEDENCE CALL USING SPEED CALL (2500 Sets)

Note: If the precedence digit is stored as part of the Speed Call number, pressing the SPEED CALL key results in the outpulsing of the precedence digit. If the precedence digit is not stored as part of the Speed Call number, the call is sent out as Routine and can only be promoted if it encounters network blocking (overflow tone).

STEP	ACTION	VERIFICATION
IF PRECEDENCE DIGIT IS STORED IN SPEED CALL		
1	Go off-hook.	Dial tone heard.
2	Dial '#'.	Dial tone stops.
3	Dial '3'.	Dial tone heard (overflow tone if code is not allowed).
4	Dial one- to three-digit code for desired speed call number.	Required number is outpulsed by the system. If call was successfully completed, ringback is heard. When call party answers, 2-way conversation.
5	Go on-hook.	Set is idle.
<p><i>Note:</i> SPRE = Special Function Prefix. Code can be one of: FO=SPRE+ S-1-2, F=SPRE+8+5, I=SPRE+8+8, P=SPRE+8+0.</p>		

INTEGRATED SERVICES NETWORK

MERIDIAN SL-1*

AUTOMATIC CALL DISTRIBUTION
BASIC FEATURES DESCRIPTION

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Reason for Reissue: To correct the references to 'Music-On-Delay' and update the the practice. The changes are indicated by arrows in the margin.

* Meridian and SL-I are trademarks of Northern Telecom Limited

1. GENERAL

1.01 Automatic Call Distribution (**ACD**) is an optional feature available with Meridian SL-1. The Meridian SL-1 ACD feature is used when a large volume of incoming calls is to be answered by a group of telephones allocated for this purpose. Incoming calls are served on a first-in first-out basis and are distributed among the available telephones (called Agent Positions) such that the agent position that has been idle the longest is presented with the first call. This ensures an equitable distribution of the incoming calls. Possible applications of this feature include airline reservation offices and telephone ordering centers.

1.02 This practice describes the package of basic ACD features which includes the following

. IN-CALLS key

- Directory Number (**DN**) key

- NOT READY key

- release (**RLS**) key

- MARE BUSY key

- | call source identification

- AGENT status lamps

- | DISPLAY AGENTS key

- DISPLAY QUEUE keys and queue status lamps

- recorded announcements (**RAN**)

- | night treatment of calls

- priority trunks.

1.03 'Music on hold' and 'ACD/CDR Connection Record Option can also be provided as separate options, when required.

1.04 The Meridian **SL-1** ACD feature is available in several packages, with options to suit different levels of user need and complexity:

- Package A - ACD Basic Features

- Package B - ACD Advanced Features (**553-2671-101**)

- Package C1 - ACD Management Reports (**553-2671-102**)

- | Package C2 - ACD Load Management (**553-2671-103**)

- | Package D - ACD with an Auxiliary Data System (**553-2671-W**). ←

PRACTICE **553-2671-100**

1.05 Allowable package combinations are A, **A+B**, **A+B+C1**, and **A+B+C1+C2**. Package D provides all features of packages A, B, and C2, and replaces package C1.

2. DESCRIPTION

2.01 A Meridian SL-1 with the ACD feature is basically divided into two parts: the ACD part and the PBX part.

2.02 Calls that come into the system on incoming trunks dedicated to the ACD part of the system are lined up in a 'queue' in the Meridian SL-1 computer, and are answered by a group of SL-1 type telephones allocated for that purpose. Each of these telephones is dedicated to a particular ACD-DN (queue) and is referred to as an agent position. Incoming calls are lined up in priority first, and then in the order in which they arrive. They are then distributed equally to all the manned agent positions currently assigned to that particular queue. The system does this by creating two queues (Fig. 2-2 and 2-3) for each ACD-DN, one composed of incoming calls, and the other composed of agent positions which are ready to receive calls. The system matches free agents with incoming calls to arrive at an equitable call distribution.

2.03 Each Meridian SL-1 can serve up to 32 customers, each customer having up to 240 queues. Each queue is defined by its 'ACD-DN', which is a unique 3- or 4-digit number taken from the customer's PBX numbering plan. Up to 240 (Meridian VLE, XL, XN, XT), 120 (Meridian N, NT, LE) or 70 (Meridian M, S, MS) agent and supervisor positions can be assigned to each ACD-DN.

ACD Agent Telephone Types

2.04 An ACD agent position is an SL-1 telephone assigned to an ACD-DN. Some of the agent positions can be designated "supervisor" and can be switched between agent and supervisor status by their users. Any version of the SL-1 type telephone can be used as an agent position, depending on customer needs. NE-500/2500 type telephones cannot be used as ACD agent positions.

2.05 **QSU1**. The QSU1 telephone is the basic SL-1 type telephone and is equipped with:

- handset
- push button dial pad
 - | VOLUME UP and VOLUME DOWN keys
- built-in loudspeaker
- ten programmable feature keys, eight of which have an associated lamp for visual indications
 - | HOLD key.

2.06 The QSU1 fulfills the minimum requirement for an agent position when neither digit display nor headset operation are required. The number of programmable feature keys can be increased by providing one or more QMT1 or QMT2 add-on modules. The QMT1 has 10 buttons, the QMT2 has 20 buttons.

2.07 **QSU3**. The QSU3 telephone has, in addition to the features of the **QSU1**, a 16-number digit display used to show call source and other information related to call-processing and other optional features. This type of telephone should be provided when the agent needs to know either the source of a call or that a call has been overflowed from another queue. It is also required if the agent position is to be equipped with 'Date and Time' or other display features.

2.08 **QSU6**. This SL-1 type telephone has no digit display, but is equipped with two jacks for plug-in headset operation. **One** jack is to accommodate the agent's headset, and the other 'listen-only' pair accommodates another headset for supervisory or training purposes. **QSU6B** and later vintage telephones do not have a built-in handset. Customers who want the option of using handsets or headsets at an individual position must, therefore, provide a plug-in headset.

2.09 **QSU7**. This SL-1 type telephone has both digit display and two jacks, but does not have a built-in handset (see Fig. 2-1).

ACD Supervisor
Telephone

2.10 The ACD supervisor should use a digit display SL-1 telephone (**QSU3** or **QSU7**), and be equipped with sufficient **10-** or **20-key** add-on modules (**QMT1** or **QMT2**, respectively) to provide:

- | one key/lamp pair for each ACD agent in the supervisor's group (up to **40**)
- one key/lamp pair for each call queue (ACD-DN) for which the supervisor's agents are responsible (up to 5).

2.11 Headset Operation. The QSU6 or QSU7 telephones can be equipped with a plug-in headset. **QSU1** or **QSU3** telephones can be modified to provide single headset jacks by using the **QKN1** kit. When agent positions are equipped for headset operation, tone ringing is replaced by a 3-s burst of buzz tone from the loudspeaker.

Operation

2.12 ACD Queuing. The ACD software program creates two queues for each ACD-DN: one for incoming calls and one for ACD agent positions.

- | incoming Calls Queue. When all agents are busy the incoming calls are placed in the appropriate incoming call queue on a priority and order-of-arrival basis. They are then presented to agents as the agents become available.
- | Agent Queue. If agents are available but there are no incoming calls waiting, the available agents are placed in the designated agent queue on a first-in, first-out basis. The agent who has been waiting the longest receives the first incoming call.

2.13 Incoming Call Queue. Calls in the incoming call queue may result from one of the following:

- seizing an incoming CO, FX, or WATS trunk arranged to auto-terminate on an ACD-DN rather than the PBX attendant
- | dialing the ACD-DN over an incoming TIE or DID trunk
- | dialing the ACD-DN from a telephone within the Meridian SL-1

- | an attendant extending a call to the ACD-DN.

2.14 An individual trunk is assigned to a specific incoming call queue using Meridian SL-1 service change procedures (553-2YY1-220/-221 or -310/-311). A trunk may be assigned to only one ACD-DN at a time.

2.15 The following rules apply to agent queues:

- | An agent can be assigned to only one agent queue (ACD-DN) at a time. The internal association of a call queue with an agent queue determines which group of agents serves a particular group of incoming calls.
- | The length of an agent queue is determined by the number of agents waiting to serve ACD calls.
- | The maximum number of agents that can be assigned to an ACD-DN in order to ensure efficient call processing for each Meridian SL-1 type, is as follows:

LE, N, NT	- maximum 120	←
VLE	- maximum 240	
XL, XN, XT	- maximum 240	←
M, S, MS	- maximum 70.	

- An ACD-DN is not the same as a multiple-appearance DN. Thus, the agent positions are not restricted to a single SL-1 network loop.

2.16 An agent is removed from the queue by:

- | answering an ACD call presented to the agent position
- | originating or answering a call on any other key of the agent's position
- | operating the NOT BEADY key
- | operating the MAKE BUSY key
- | unplugging the headset/handset (QSU7 telephones).

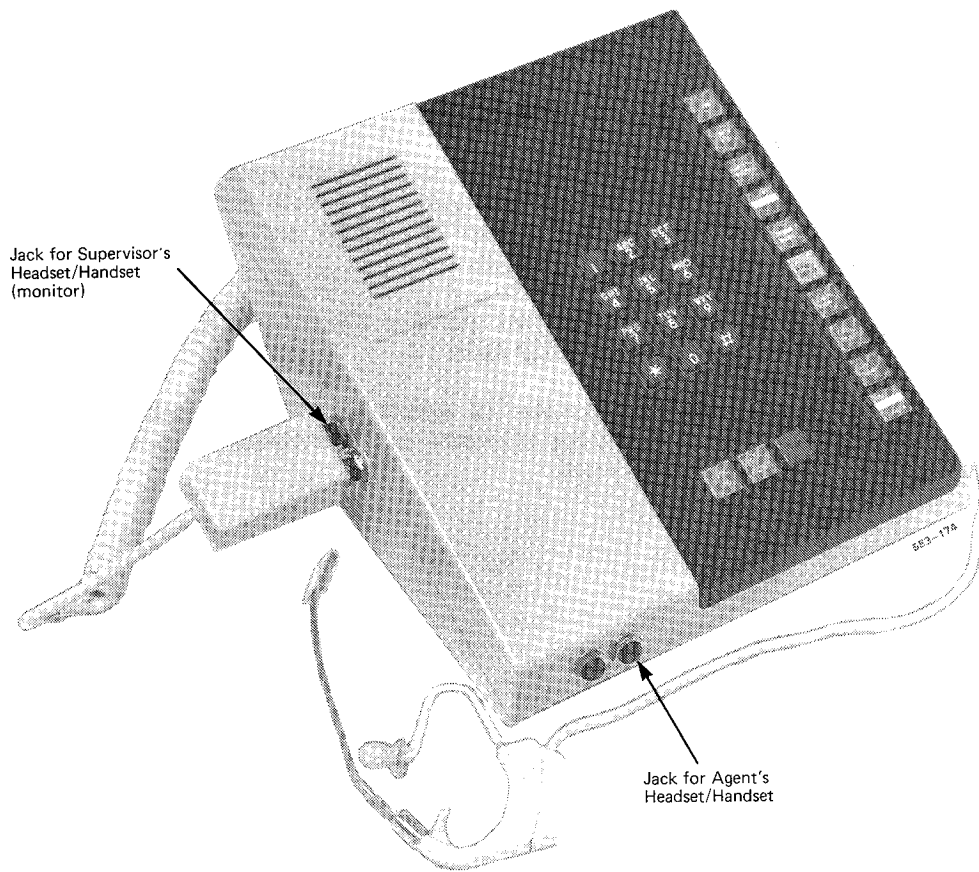


Fig. 2-1
Meridian SL-1 ACD Agent Set (QSU7)
Used at an ACD Agent Position

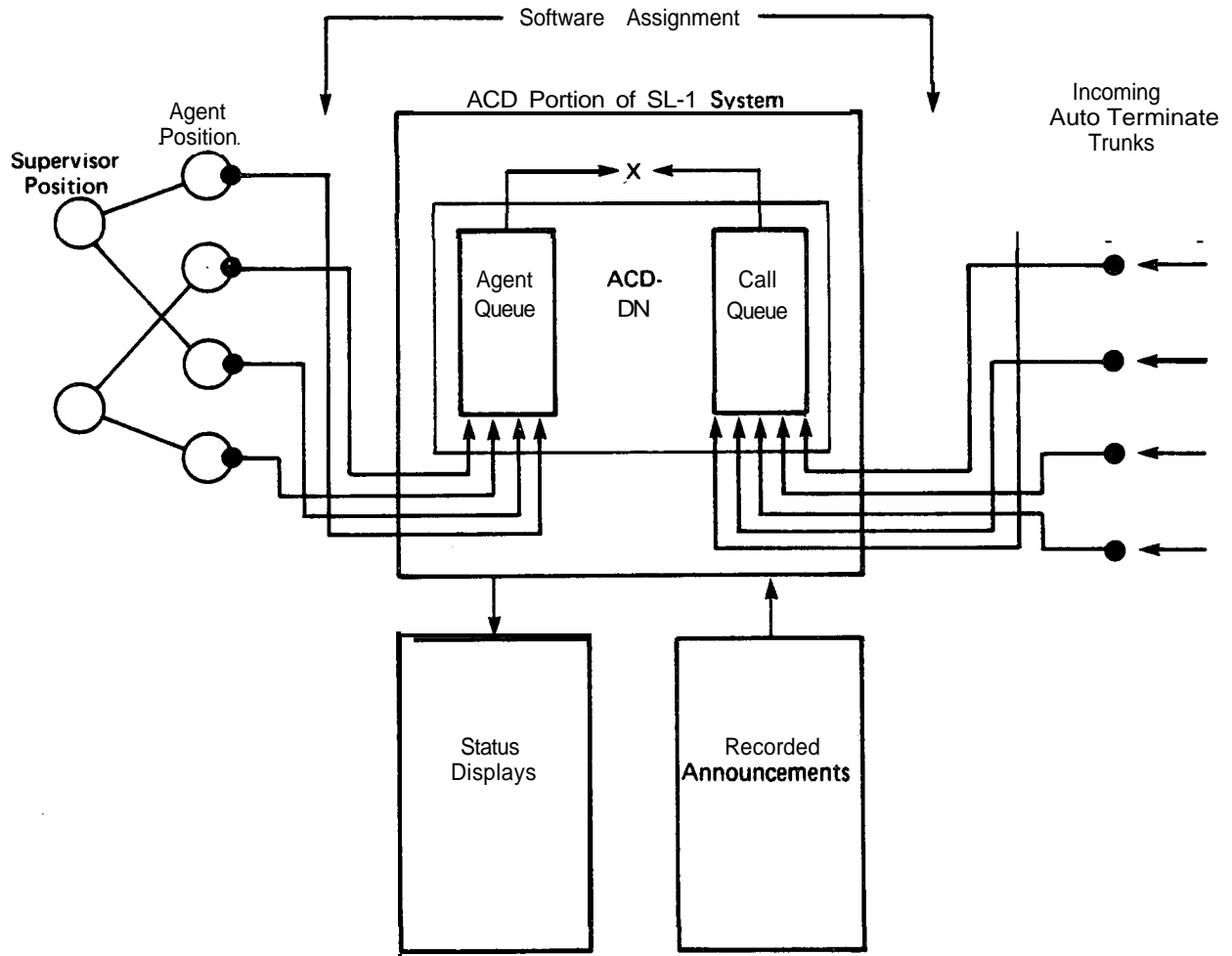


Fig. 2-2
Diagram of a Single Queue Meridian SL-1 ACD

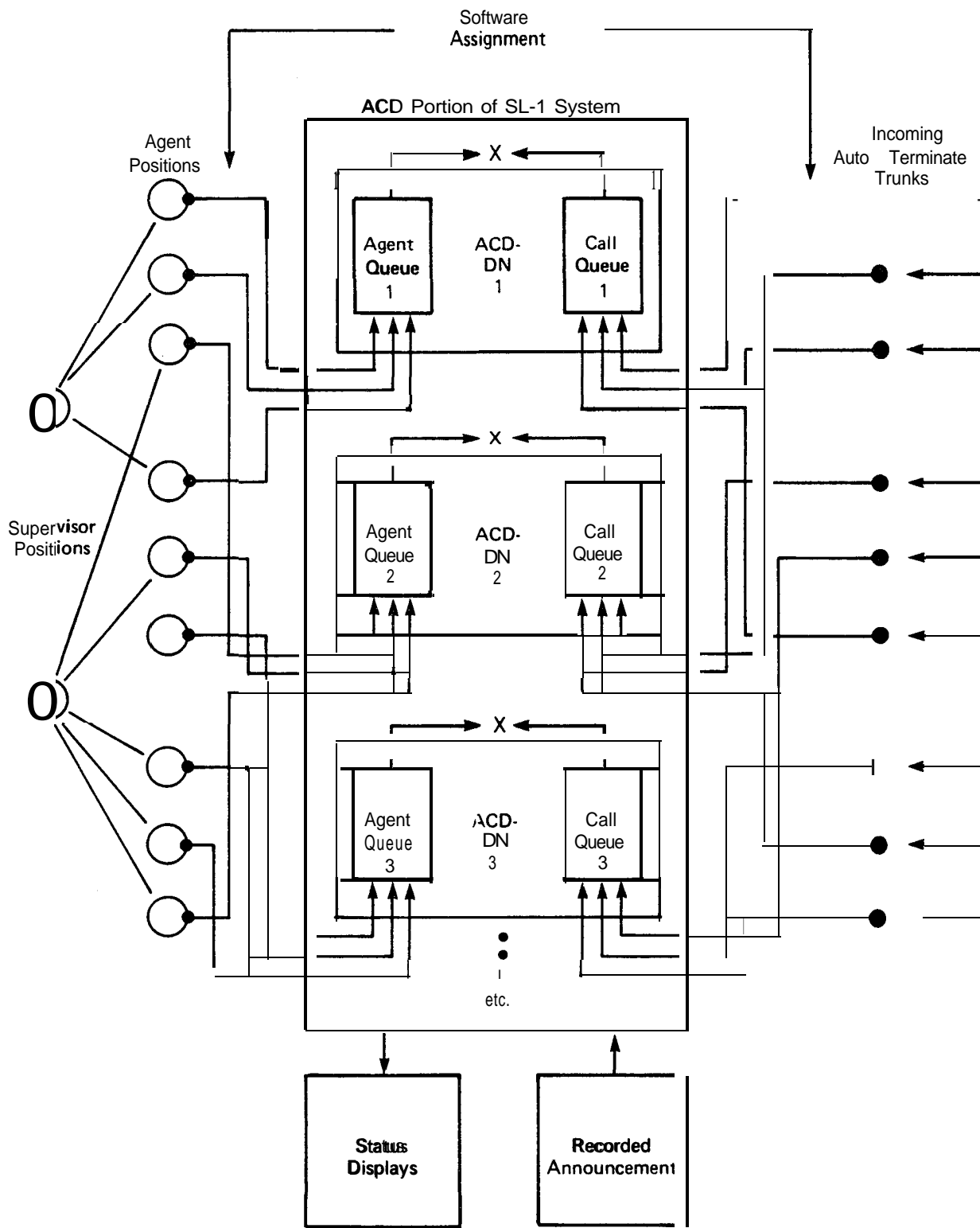


Fig. 2-3
Diagram of a Multiple Queue Meridian SL-1 ACD

3. AGENT FEATURES

IN-CALLS Key

3.01 ACD calls are presented on the IN-CALLS key at the agent position. Associated with the key is a number, the position identifier (POS ID) which is a unique number taken from the customer's numbering plan. This number identifies the agent's position for Automatic Number Identification, Automatic Identification of Outward Dialing (ANI/AIOD), and Call Detail Recording (CDR) purposes. This number cannot be the same as the ACD-DN, and cannot be used to call the ACD position. The IN-CALLS key is always located on key position 0 (bottom key) on the agent position.

3.02 When an incoming ACD call is presented to an agent position:

- | Tone ringing is heard, if the telephone is equipped with a built-in handset which is in its receptacle.
- | A 3-s burst of buzz tone is heard from the telephone loudspeaker, if the telephone is equipped with a headset or plug-in handset.
- IN-CALLS lamp flashes slowly (60 ipm).

3.03 When the agent answers the call by depressing the IN-CALLS key or by picking up the built-in handset:

- | the speaker is silenced
- the IN-CALLS lamp is steadily lit
- | a talking connection is established with the caller.

3.04 ACD calls can be terminated in one of two ways. If the caller disconnects first, the call is released and any subsequent call is presented. If the agent wishes to disconnect first, the IN-CALLS key is pressed. This releases the current call and presents the next call. The IN-CALLS lamp flashes quickly (120 ipm) if the call is placed on 'hold'. Note that calls extended by the attendant to the ACD-DN are not automatically returned to the attendant if they remain unanswered past the 'attendant recall' threshold.

Directory Number Key

3.05 Each agent position can be assigned one or more individual (or shared) DN to originate or receive non-ACD calls. These are normal PBX extensions which appear on one or more DN-type feature keys. The normal SL-1 classes of service apply to these DN, including single or multiple appearance arrangements. When an agent presses a DN key to make or receive a call, any other call in progress is automatically released (unless HOLD has been pressed). If the agent was waiting for an ACD call at the time of DN key activation, the position is automatically removed from the agent queue. When the call on the DN key is released, the position is automatically returned to whatever state it was in before the DN key was pressed (waiting for ACD call, 'not ready', etc.). Any call being presented on the IN-CALLS key (but not yet answered by the agent) when the DN key is pressed is moved back to the head of its priority grouping in the incoming call queue for the ACD-DN.

3.06 The DN lamp flashes slowly when an incoming call to the DN is presented to the telephone. It is lit steadily while the call is in progress, and flashes quickly, if the call is placed on 'hold'. Activating the MAKE BUSY key prevents DN calls from being presented to the agent's position.

NOT READY Key

3.07 An agent, on completion of an ACD call, may require time to perform tasks before accepting another call. (This is termed Post Call-Processing time.) Upon depression of the NOT READY key, the agent is removed from the agent queue, and the NOT BEADY lamp is lit steadily.

3.06 While the agent position is in the 'not ready' state, the agent may receive or originate calls using the DN key, but no ACD calls are presented on the IN-CALLS key. When the agents post-call tasks have been completed, the agent again depresses the NOT BEADY key. The 'not ready' lamp goes out, and the agent is placed in the agent queue. (It is also possible to press the IN-CALLS key to cancel 'not ready' and reenter the queue.) Any call being presented on the IN-CALLS key (but not yet answered by the agent) when the NOT BEADY key is pressed is moved back to the head of its priority grouping in the incoming call queue for the ACD-DN.

Release (RLS) Key

3.09 The optional **RLS** key may be assigned in place of (or as well as) the NOT BEADY key, if there is no requirement for post-call work. When an agent disconnects from a call by using the **RLS** key, the agent position is immediately placed in the agent queue.

MAKE BUSY Key

3.10 The MAKE BUSY key makes the agent position appear busy to the system, thereby preventing the agent from receiving any calls. When the key is operated, the MAKE BUSY lamp is steadily lit and the agent is removed from the agent queue. Any call being presented on the IN-CALLS key (but not yet answered by the agent) is moved back to the head of its priority grouping in the incoming call queue for the ACD-DN, i.e., pressing the MAKE BUSY key does not disconnect any active call. The agent will not receive any calls on the IN-CALLS key or DN keys, but may originate a call on a DN key.

3.11 If all the agent telephones assigned to one ACD-DN have the MAKE BUSY feature activated, the ACD-DN concerned goes into 'night' mode. ACD night mode is independent from the night service for the rest of the Meridian SL-1 and can be different for each ACD-DN (see Night Treatment).

Call Source Identification

3.12 This optional feature, using the digit display of the agent telephone (**QSU3** or **QSU7**), shows the agent where a call is coming from, allowing the agent to give an appropriate answer to the caller. For calls coming from incoming trunks, the trunk access code and the member number of the trunk are given. For calls originating from within the Meridian SL-1, the DN of the calling telephone is given.

3.13 The ACD agent position can be equipped with other features such as 'call transfer', 'date and time', etc. These features are described in **553-2YY1-105** and their operation defined in **553-2YY1-305** or **553-2YY1-315**. Features that can be used with a non-ACD SL-1 type telephone can also be used with the DN key of an SL-1 type telephone assigned to ACD.

4. SUPERVISOR FEATURES

4.01 The assignment of agents to supervisors is governed by the assignment of AGENT keys on the supervisor's telephone. (This is independent of the assignment of agents to **ACD-DN**.) An agent cannot be assigned to more than one supervisor at any one time.

AGENT Lamps

4.02 Up to 40 key/lamp pairs may be assigned to the AGENT function. (The key associated with each lamp is not used unless the ACD advanced features are equipped.) The lamps show the state of each individual agent in the supervisor's group and are updated whenever the status of the agent changes, whatever the supervisor position's mode of operation (agent/supervisor/make busy).

4.03 The lamps can assume the following states:

- 1 Dark. The agent position is not manned.
- 1 Steadily Lit. The agent is busy on an ACD call or is in 'not ready' mode.
- 1 Slow Flash (**60 ipm**). The agent is waiting for an ACD call.
- 1 Fast Flash (**120 ipm**). The agent is busy on a non-ACD call.

Note: If the system is equipped with ACD advanced features, read 553-2671-101 for a description of the changed/added AGENT key/lamp functions.

DISPLAY AGENTS Key

4.04 This key (the associated lamp is not used) when pressed, summarizes on the supervisor's digit display the status of all agents assigned to the supervisor. It gives a count of the number of *agents in* each of the four states at the moment the key was operated. The digit display is updated every time the DISPLAY AGENTS key is operated. The display takes the form:

AA-BB-CC-DD

where:

AA = the number of agents busy on ACD calls, or in 'not ready' mode

BB = the number of agents waiting for ACD calls

CC = the number of agents busy on non-ACD calls

DD = the number of agents positions not manned.

DISPLAY QUEUE Keys

4.05 A DISPLAY QUEUE key can be assigned to a supervisor position for each applicable ACD-DN. When the key is pressed, the calls-waiting status for the ACD-DN is displayed as follows:

AAA-BBB-CCC

where:

AAA = the number of calls currently waiting in the queue

BBB = the number of agent positions manned for the ACD-DN

ccc = the waiting time, in seconds, of the first call in the queue.

4.06 The lamp associated with the DISPLAY QUEUE key provides summary information on the calls waiting status for the ACD-DN. When steadily lit, it means there is more than one call waiting in the queue. A DISPLAY QUEUE key for a particular ACD-DN can be assigned to a maximum of 8 supervisor positions regardless of the number of supervisors.

5. SYSTEM FEATURES

incoming Trunk
Restrictions

5.01 Abandoned ACD calls are removed from incoming call queues and RAN unless the incoming trunk used for the call is of the loop-start type. Far-end disconnect on loop start trunks is only detected during ringing. If a call is routed to RAN, answer supervision is returned to the trunk and ringing is stopped. On completion of the announcement, the call is placed back in the queue on 'silent hold' (unless 'music on hold' has been specified for the ACD-DN concerned).

5.02 Incoming calls on trunks that do not have provision for disconnect supervision are not released by the system when the agent terminates a conversation by pressing the IN-CALLS or RLS keys. Thus, trunks without disconnect supervision should not be used for ACD systems.

Recorded
Announcement

5.03 When the number of calls in a call queue exceeds the number of available agents, delays in the answering of calls is encountered. A RAN can be used to advise of the delay. The Meridian SL-1 ACD feature allows a choice of two recorded announcements which operate independently of each other but can be used together.

5.04 RAN treatment is specified for each ACD-DN independently of the other ACD-DN's. Unlike the 'music' feature, the RAN feature does not use the conference circuit packs. When the system determines that an ACD caller is ready to receive RAN, the caller is connected to a recording trunk at the beginning of the RAN cycle, on a one-to-one basis. If an agent becomes available to serve a call currently receiving RAN, the RAN is disconnected and the call is presented to the agent. Calls from an attendant or transferred by an attendant to the ACD-DN also receive RAN.

- (a) First RAN Prior to X11 RLS 2. The system keeps track of how long each call receives ringback tone before being answered. Each incoming call is evaluated by the system on the basis of how long the most recently answered call on the ACD-DN had to wait for an answer. If the time expected to answer an incoming call exceeds a customer-specified time (t1), the call is diverted to a recorded announcement (subject to the beginning of the recorded announcement cycle) as soon as it is received by the system if Delay Start is specified in the Route data block. A call that arrives in the queue when the delay threshold (t1) was not exceeded, will receive the first RAN after the customer specified time of t2. After the announcement, the call is placed on 'silent hold, or receives 'music-on-delay' until answered or abandoned. Thus, a caller calling the ACD-DN hears an audible message describing a delay possibility, and can disconnect, decreasing the holding time on the trunk under busy conditions.
- (b) First RAN X11 RLS 2'. X11 RLS 2 provided a new prompt to allow customers to specify First RAN on arrival (FROA, service change overlay 23). If the response to FROA is NO, all calls must wait for the duration of the first RAN timer t1 (as specified) prior to receiving the First RAN. If the response is YES a call which is eligible for first RAN treatment will be provided First RAN immediately upon entry into the ACD queue.

- ↗
- (c) Second RAN. On completion of first RAN, a customer specified timer for a second recorded announcement (**t2**) is started. Each call that has been in the queue longer than **t2** seconds, gets second RAN. Second RAN is repeated every **t2** seconds until the call is answered or abandoned.
- ↘

5.05 In summary, first RAN is given either immediately upon being queued or at **t2** seconds later. Second RAN is presented, if necessary, **t2** seconds following first RAN and repeated at **t2** second intervals.

5.06 The two recorded announcements operate independently of each other. Both are optional and the customer can have either announcement alone or both. The two timers, **t1** and **t2**, bear no fixed relationship to each other. This allows the customer the flexibility to specify the RAN treatment to suit the requirements of the installation. Such factors as the time allowed for the announcement and the waiting time between announcements depends on the type of recorded announcement equipment used. Meridian SL-1 are compatible with Audichron, CODE-A-PHONE* Cook Electric and Interalia announcement machines. (Refer to '553-2641-200 for a description of the Recorded Announcement trunk.)

Night Treatment

5.07 This optional feature can be used to inform ACD callers that the ACD location is not in service (after-business-hours calls). There are three ways that such calls may be handled:

- (a) RAN may be provided as part of the night treatment to after-hours ACD calls indicating that the ACD location is closed.
- (b) The ACD call may be 'call forwarded' (maximum number of 'call forward digits is 23) to another ACD location or to a night service number. Only internal calls or calls from trunks that provide disconnect supervision can be 'call forwarded'.
- (c) No treatment' at all may be given. In this case, callers receive **ringback** tone only until the caller abandons the call, and no answer supervision is given to the central office from the Meridian **SL-1**.

5.08 ACD calls may be provided with both RAN and 'call forward'. This feature requires all agent positions to be equipped with a MAKE BUSY key. The feature is activated automatically when all agent positions assigned to an ACD-DN have the MARE BUSY key operated.

Priority Trunks

5.09 This optional feature allows the customer to designate certain ACD incoming trunks as priority. When implemented through service change programs, calls to ACD on priority trunks move directly to the front of any nonpriority calls in the call queue. Any nonpriority ACD calls in the queue maintain their position in relationship to each other, but are placed behind priority calls in the queue. Although ACD calls are not 'dropped' or 'lost' by the Meridian SL-1, long waiting times for nonpriority calls can be avoided by using 'automatic overflow' of the Advanced Features option (see **553-2671-101**).

* CODE-A-PHONE is a trademark of Northern Telecom Limited

Music on Hold

5.10 'Music-On-Hold' is provided from trunks (Fig. 5-1) which have been specified for 'music' to callers who have been connected with their destination (not necessarily the ACD-DN) and have been subsequently put on hold either by the HOLD key or as part of the operation of another feature.

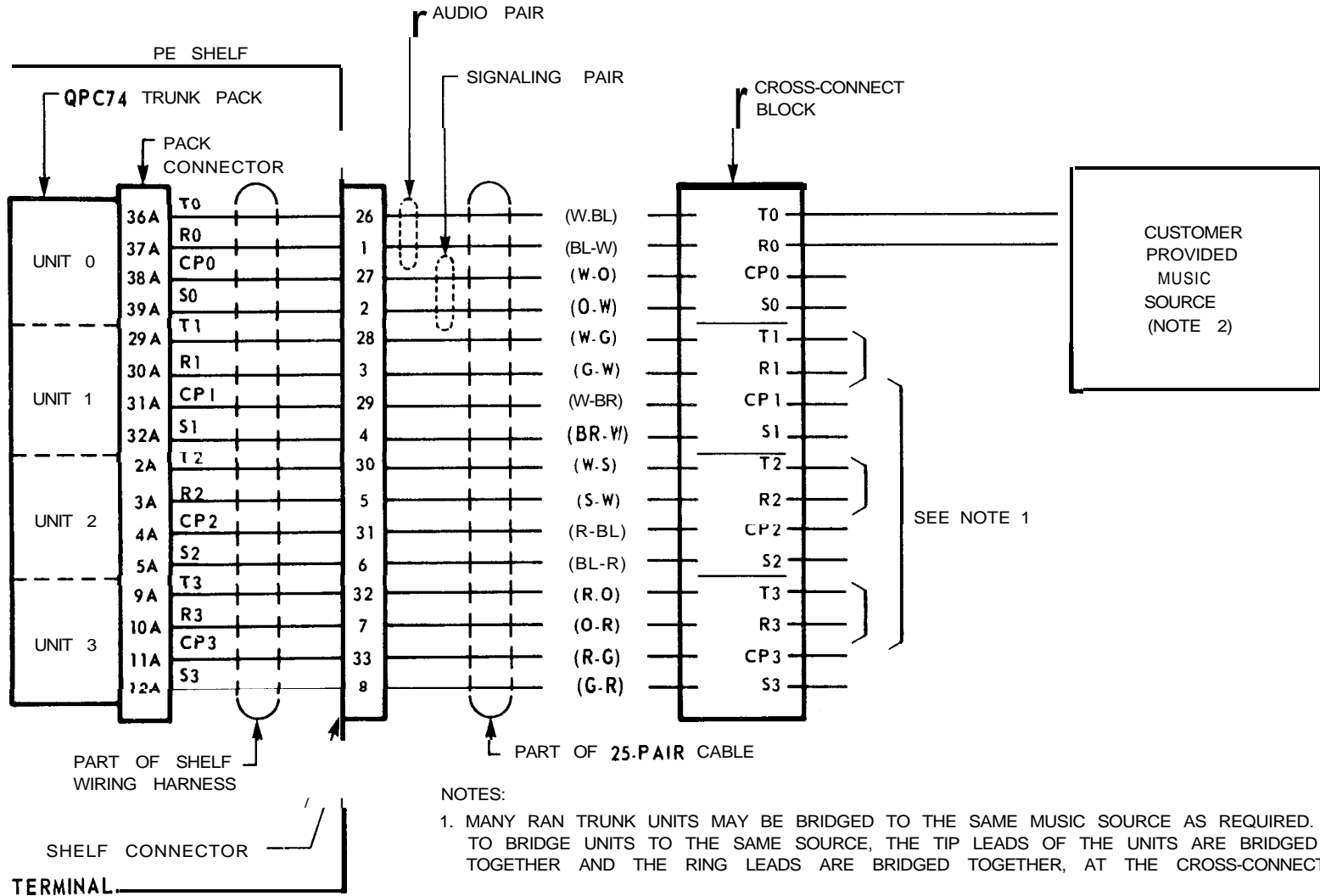
5.11 The music is taken from a music source via a 'music' trunk (specified in service change) and connected to a conference circuit pack. Callers being put on 'hold' are bridged into the conference circuit by software via a listen-only path. Each music trunk is assigned to a specific conference loop (which is not necessarily dedicated to music). Each ACD-DN can be programmed for a different source (if available). See 553-2671-151 for engineering information and 553-2001-220/221 or 553-2YY1-310/311 for service change information.

ACD/CDR Connection
Record Option

5.12 The Connection Record Option, introduced in Generic X11 Release 3, allows the customer to build and maintain a call profile that can be automatically transferred from one ACD agent to another.

5.13 A full description of the CDR Connection Record Option is given in 553-2631-100, Appendix 1.

Fig. 5-1
Music Trunk Connections



NOTES:

1. MANY RAN TRUNK UNITS MAY BE BRIDGED TO THE SAME MUSIC SOURCE AS REQUIRED. TO BRIDGE UNITS TO THE SAME SOURCE, THE TIP LEADS OF THE UNITS ARE BRIDGED TOGETHER AND THE RING LEADS ARE BRIDGED TOGETHER, AT THE CROSS-CONNECT
2. THE CUSTOMER PROVIDED MUSIC SOURCE MUST BE COMPATABLE WITH THE RAN TRUNK (i.e. 2-4 ohm IMPEDANCE OR 600 ohm IMPEDANCE MUSIC SOURCE).

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

AUTOMATIC CALL DISTRIBUTION
 ADVANCED FEATURES DESCRIPTION

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Reason for Reissue: Extended agent observe functions, included in Generic X11 Release 2, are added. Since this is a separate subpart within part 3, revision marks have not been provided.

* SL-1 is a trademark of Northern **Telecom** Limited

1. GENERAL

1.01 Automatic Call Distribution (**ACD**) is a feature of the SL-1 Business Communications System which allows a large number of incoming calls to the same directory number (called the **ACD-DN**) to be answered by *agents' at agent positions which share the influx of calls equally. The ACD system is more fully described in 553-2671-100, which also describes the ACD Basic Features package.

1.02 The ACD Advanced Features complement the Basic Features package. They allow ACD agents to communicate directly with the supervisor (for normal or 'emergency' assistance) and allow the supervisor to perform more operations, such as handling emergency calls, observing agents, etc.

1.03 The SL-1 ACD Advanced Features include the following

- Agent/Supervisor Communication
- EMERGENCY Key
 - ┆ Emergency Teletype Message
- Calls Waiting Indication
 - ┆ AGENT Key/Lamp
- DISPLAY QUEUE Lamp
 - ┆ NOT READY Key
 - ┆ OBSERVE AGENT Key
 - ┆ Supervisor/Agent Communication
 - ┆ Extended Agent Observe
 - ┆ INTERFLOW Key
 - ┆ Automatic Call Queue Overflow
 - ┆ Call Forcing
- Music On Delay.

1.04 Practices 553-2671-102 and -103 describe additional ACD packages and options that are available to provide Management Reporting and Load Management capabilities for the ACD operation.

2. AGENT FEATURES

Agent/Supervisor Communication

2.01 Two-Party Operation. When an ACD agent is not active on an ACD call, the SUPERVISOR key can be pressed to call the assigned supervisor. The associated lamp flashes and **ringback** tone is heard. When the supervisor answers, the SUPERVISOR lamp on the agent's set lights steadily, and two-way communication between the agent and the supervisor is established.

2.02 Three-Party Operation. If the ACD agent is active on an ACD call and presses the SUPERVISOR key, the calling party is put on hold (fast flashing IN-CALLS lamp), the lamp associated with the SUPERVISOR key flashes and **ringback** tone is heard. When the supervisor answers, the SUPERVISOR lamp lights steadily (at the agent's set) and two-way communication is established between the agent and the supervisor. Pressing the SUPERVISOR key a second time, adds the calling party (held on the IN-CALLS key) to the conversation. At this time the SUPERVISOR lamp on the agent's set goes dark, and the conference (agent/caller/supervisor) is established on the agent's IN-CALLS key. The agent can release from the conference, leaving the caller and supervisor connected, by pressing the RLS key. Alternatively, the supervisor can release, leaving the agent and caller connected.

2.03 When an agent calls the supervisor, tone ringing is heard from the speaker of the supervisor's terminal, and the ANSWER AGENT lamp flashes. The supervisor answers the call by pressing the ANSWER AGENT key, at which time the digit display shows the calling agent's ACD POS-ID, and the supervisor's NOT READY lamp lights steadily. If the NOT READY lamp is already on, it remains on.

EMERGENCY Key

2.04 In the event of threatening or abusive calls, an agent takes note of the trunk access code and trunk member number shown on the digit display, then presses the EMERGENCY key to conference with the supervisor (if one is assigned), bypassing all other agent-to-supervisor calls. The agent must remain in the conference for the duration of the call. At the supervisor's set, the ANSWER EMERGENCY lamp flashes and continuous buzzing is heard. The ACD POS-ID of the calling agent is displayed on the digit display of the supervisor's set. If the option is equipped (by the customer), a tape recorder can be used to record the call and a message typed on the maintenance TTY to facilitate call tracing. When the agent presses the EMERGENCY key, the associated lamp indicates the action being taken as follows:

- **Derk.** Neither supervisor nor recording trunk is available. The supervisor may be busy on another emergency call, unassigned, or not equipped with an ANSWER EMERGENCY key, and recording trunks are all busy or unassigned.
- ⌋ **Flashing.** The supervisor is available, but has not answered the emergency call yet.
- **Lit Steadily.** Either the supervisor, or a recording trunk, or both, are conferenced into the call.

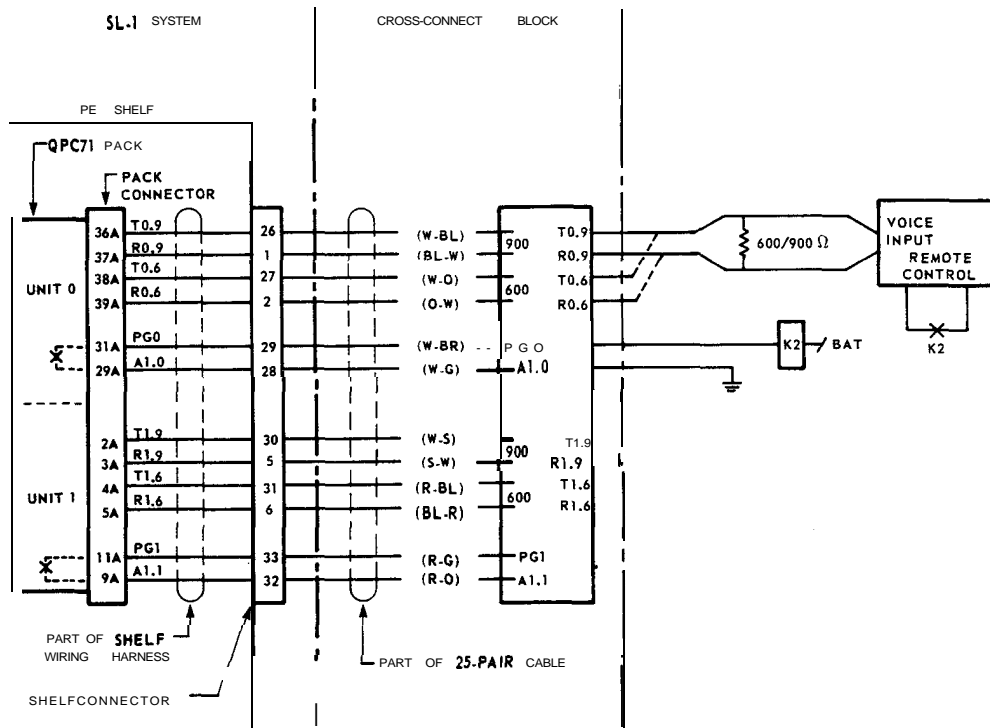


Fig. 2-1
Paging Trunk Connections

2.05 The supervisor, when answering an emergency call, can leave the handset on-hook (or headset unplugged), if desired, to initially 'listen-only' through the built-in loudspeaker. The supervisor can also put the emergency call on 'hold', without affecting the agent or caller, to make another call or take some other action and return to the emergency call by pressing the ANSWER EMERGENCY key again. The ANSWER EMERGENCY lamp flashes at 120 ipm while the call is on 'hold'.

2.06 The recording device (such as a tape recorder) can be connected to a paging trunk (QPC71) with 'recorder' class of service (defined in software). When the agent presses the EMERGENCY key, the recording device is started and the recording trunk is **conferenced** into the agent-incoming call conversation via the conference circuit pack.

2.07 Paging trunk Connections: The input impedance of a typical tape recorder is 47 kΩ, whereas the output impedance of the SL-1 paging trunk is switch-selectable for 600 or 900 Ω. It may, therefore, be necessary to add a 600 or 900 Ω resistor in parallel with the tape recorder input to properly terminate the paging trunk.

2.08 The paging trunk provides relay contacts (lead A1 and PG) which are closed when the paging trunk seizes the line. A separate relay, K2, should be provided by the customer to avoid any possible incompatibility between the recorder and the trunk. The system should be wired so that the paging trunk leads A1 and PG operate the relay, which, in turn, provides a contact closure to start the tape recorder (Fig. 2-1).

Emergency Teletype
Message

2.09 When the agent presses the EMERGENCY key, one or more of the following messages can be generated on the maintenance teletype:

EMRIOO RC L S C U RR MMM

EMRIOO AG XXXX YYYY

EMRIOO OR LSCU RR MMM

where:

RC denotes the recording trunk used for the emergency recorder, L S C U are the loop, shelf, card, and unit used for the recording trunk, and RR and MMM are the route and member numbers of the recording trunk.

AG denotes that an agent pressed the EMERGENCY key. XXXX denotes the ACD-DN to which the agent is assigned, and YYYY denotes the agent PGS-ID. The ACD-DN and the POS-ID digits, for software programming reasons, are given in reverse order, and the character "A" signifies a "0" and the digit "0" signifies the end of the number. For example, the ACD-DN "01A4" would denote "401" and POS-ID "5412" would denote "2145".

OR denotes the originator of the call. L S C U are the loop, shelf, card, and unit of the call origin. If the call is coming into the system, the RR and MMM parameters denote the route and member numbers of the trunk. If the call is originating from within the SL-1 system, these two fields are absent.

Calls Waiting
Indication

2.10 This lamp on the agent's set is intended to inform each agent assigned to the ACD-DN that the number of calls in the assigned ACD-DN call queue has exceeded a customer-specified threshold value, and that call handling should be speeded up. The lamp shows the status of the call queue and can assume any one of four states. The key associated with the lamp has no function. The four lamp states are:

- | Dark. There are no calls waiting.
- Steadily Lit. There are calls waiting. The ACD-DN can receive calls overflowed from another ACD-DN.
- Slow Flash. There are calls waiting. The ACD-DN cannot receive calls overflowed from another ACD-DN.
- | Fast Flash. There are calls waiting and some calls are being overflowed to another ACD-DN (if this option is equipped).

3. SUPERVISOR FEATURES

- 3.01 The ACD Advanced Features package provides the following supervisor features in addition to those in the Basic Features package.
- AGENT Key 3.02 The AGENT keys are used in conjunction with the OBSERVE AGENT and CALL AGENT keys to select the specific agent to be observed/called. The selection of the individual agent may be made anytime the OBSERVE AGENT lamp or CALL AGENT lamp is lit. Otherwise any operation of the AGENT key is ignored.
- AGENT Lamp 3.03 With the ACD basic features, a steadily lit lamp associated with an AGENT key means the agent is either handling an **ACD** call or is in the post-call processing mode (i.e., NOT READY lamp lit). The ACD advanced features provides an option for a Separate Post-Call Processing (**SPCP**) indication on the AGENT lamp. The SPCP option is enabled/disabled through the ACD overlay program (23). When the SPCP option is enabled, the AGENT lamp can assume any one of the following states:
- | Dark. The agent position is not manned.
 - | Steadily Lit. The agent is handling an ACD call.
 - | Slow Flash. The agent is waiting for an ACD call.
 - Fast Flash. The agent is busy with post-call work (NOT READY lamp lit) or is engaged in a non-ACD call.
- DISPLAY QUEUE Lamp 3.04 When the system is equipped with the 'advanced features', the DISPLAY QUEUE lamp can assume one of the following states:
- | Dark. No calls waiting for agents.
 - | Steadily Lit. Calls waiting for agents: ACD-DN can receive calls overflowed from another queue.
 - | Slow Flash. Calls waiting for agents; ACD-DN cannot accept calls overflowed from another queue.
 - Fast Flash. Calls waiting for agents; calls **are** being overflowed to another ACD-DN queue (if option is equipped).
- NOT READY Key 3.05 The following supervisor feature keys can be used only while the lamp associated with the NOT READY key is **lit**:
- . OBSERVE AGENT
 - . CALL AGENT
 - . ANSWER AGENT
 - | **ANSWER** EMERGENCY.
- 3.06 While the NOT READY lamp is lit on the supervisor's position, post call-processing time for the position is accumulated unless one of the key functions is invoked.

3.07 To receive ACD calls, the supervisor presses the NOT READY key again (NOT READY lamp goes dark) and is placed in the agent queue in the normal manner. In this mode (agent), operation of the OBSERVE AGENT, CALL AGENT, or ANSWER AGENT key is ignored. When a call is answered on the ANSWER EMERGENCY key, the supervisor position is automatically switched to the 'not ready' (i.e., supervisor) state.

OBSERVE AGENT Key

3.08 To observe the quality of service being offered to callers, the supervisor position can be used to monitor calls terminating on the IN-CALLS key of any agent in the supervisor's agent group. The supervisor operates the OBSERVE AGENT key, then the appropriate AGENT key to activate the feature. A periodic warning tone can be supplied to the agent being observed but not to the calling party, as a customer option. This warning tone lasts 256 ms, and is repeated every 16 seconds.

3.09 The supervisor cannot observe an agent who is in conference with three or more parties. Therefore, if the observed agent is in, or subsequently goes into a conference, at the time of observation, the observation is discontinued until the observed agent is in a simple call again. If the agent activates the 'call transfer' feature, any observation by the supervisor is suspended.

3.10 If the CALL AGENT key is operated by the supervisor while observing an agent, then the supervisor's voice path is enabled and a conference situation exists between the supervisor, agent, and calling party. Operation of another AGENT key at this stage, disconnects the supervisor's speech path and restores the observation mode. Reoperation of the OBSERVE key or operation of the IN-CALLS key by the supervisor terminates the OBSERVE function.

3.11 Operation of one AGENT key after another results in observation of each agent in turn. The agent being observed is identified (by POS-ID) on the supervisor's digit display. If any agent being monitored is found to need the supervisor's aid, the supervisor can operate the CALL AGENT key and be **conferenced** into the call. During service observation, any operation of the HOLD key on the supervisor's set is ignored. Operation of any other key causes the OBSERVE function to be terminated.

Supervisor/Agent Communication

3.12 The supervisor can call an agent by pressing the CALL AGENT key. The CALL AGENT lamp lights steadily, and the supervisor then presses the appropriate AGENT key to complete the call. The digit display shows the called agent's ACD POS-ID. If the supervisor wishes to call another agent, this can be done by pressing another AGENT key. The call to the previous agent is then terminated.

3.13 The CALL AGENT and ANSWER AGENT functions use separate keys on the supervisor terminal so that the supervisor can call an agent without first having to answer all the agents attempting to talk to the supervisor.

3.14 When the supervisor calls the agent, the agent's SUPERVISOR lamp flashes, and the agent hears tone ringing from the set speaker. (If busy or 'off-hook', the agent hears a 3-second buzz from the receiver.) The agent answers the call by pressing the SUPERVISOR key. If the agent is engaged in a call, the call must first be put on hold by pressing the HOLD key, before the SUPERVISOR key is pressed.

3.15 If the supervisor wants to conference with the agent and the customer, it can be done in two ways:

- (a) the agent can initiate a conference, or
- (b) the supervisor can force a conference as described under OBSERVE AGENT.

INTERFLOW Key

3.16 This feature gives the supervisor the ability to cope with excessive traffic by transferring incoming ACD calls to another predesignated DN either on the same SL-1 or, via the external switching network, to another machine. Each INTERFLOW key is associated with two 'target' numbers: an ACD-DN, and an Interflow DN (**IFDN**). The IFDN can consist of up to 23 digits and can include any required access code and asterisk (*) to indicate dialing pause. When the supervisor presses the INTERFLOW key, incoming calls to the source queue are redirected to the 'target' DN if the source queue is overloaded (using the overflow threshold set for automatic overflow). The INTERFLOW lamp flashes until the feature is de-activated.

3.17 The INTERFLOW key supplements the automatic overflow feature. After the INTERFLOW key is activated, interflow to the target DN does not occur until all the queues specified for automatic overflow have been filled past their busy thresholds (**BYTH**).

3.18 Since interflow should only occur to a lightly-loaded DN, and the status of the target DN cannot be determined automatically, this status must be determined verbally by the supervisor. After activating the interflow feature, the supervisor constantly monitors the status of the call queues to decide when to disable the interflow by pressing the INTERFLOW key again.

3.19 Sufficient numbers of outgoing trunks should be available to handle the expected volume of outgoing trunk traffic when the INTERFLOW key is activated.

Extended Agent Observe

3.20 Extended Agent Observe functions are available on Generic X11 Release 2 and later. This feature provides ACD supervisors with additional capabilities to control the quality of service offered to callers and to monitor the overall performances of ACD agents or supervisors.

3.21 The following capabilities are available with the Extended Agent Observe feature.

- A supervisor can observe an established ACD, DN, conference or supervisor call originated/answered at any ACD agent position in the customer's operation.
 - | A supervisor can observe/call any agent in the ACD operation by first operating the OBSERVE/CALL AGENT key, then dialing the Position-ID of the agent. (This is in addition to the existing observe/call agent capabilities.
 - | A supervisor (with an Allow Observation of Supervisor class of service) can observe another supervisor in the ACD operation by entering the observe mode and dialing the Position-ID of the supervisor to be observed.

3.22 Limitations. The following limitations apply to observation of an agent or supervisor:

- (a) Only ACD sets (i.e., with an IN-CALLS key defined) can be observed.
- (b) An ACD set cannot be observed if it is idle or if it is already being observed by another supervisor.
- (c) An ACD set cannot be observed while in the following call states (see Note): :
 - (1) the call is not yet established
 - (2) the call is not active (e.g., on hold)
 - (3) the call is connected to a release link trunk
 - (4) the call involves an attendant
 - (5) the call is private line
 - (6) the call is being transferred
 - (7) a conference is being set up.
- (d) A supervisor's set cannot be observed if it is in the observe mode.
- (e) A supervisor's set cannot be observed if the observing supervisor does not have an Allow Observation of Supervisor class of service.

Note: The observation connection is reattempted every 256 ms. When the call state changes such that observation is allowed, the observation conference is established.

3.23 If the observe warning tone is specified (ACD - overlay 231, the ACD position being observed will be presented with the intermittent tone.

3.24 Introduction of a dial-access method of observation enables a customer to eliminate the AGENT key/lamp appearances at a supervisor's set. If this is done, the effect on other ACD supervisor functions is as follows.

- (a) The agent status information that is presented by the AGENT lamps is no longer available.
- (b) The DISPLAY AGENTS key no longer functions.

3.25 In order to eliminate the AGENT key/lamps and maintain agent-to-supervisor assignments, the SL-1 set service change program (overlay 11) is modified to enable the agent's set to be associated with a supervisor position. In this way, agent key functions (EMERGENCY, SUPERVISOR) continue to function as normal.

3.26 The following intercept treatments apply when a supervisor is dialing in the OBSERVE or CALL AGENT mode.

- 1 If the ACD set is currently being observed by another supervisor, busy tone is presented.
- 1 If the dialed Position-ID is invalid, or if the set is not an ACD set, overflow tone is presented.
- 1 If a supervisor does not have an Allow Observation of Supervisor class-of -service, and attempts to observe another supervisor, overflow tone is presented.

4. OTHER ACD FEATURES

Automatic Overflow

4.01 This feature allows incoming ACD calls to be diverted from the call queue in which they would normally be placed (called the **source** queue) to another queue (the target queue) during busy periods. Up to three target queues may be designated for each source queue. The target queue that meets the requirements for overflow (i.e., is not itself handling a volume of calls which exceeds a predefined threshold) is selected as the queue to which incoming calls are redirected. Overflow does not occur if none of the overflow queues meets these requirements. The situation is evaluated for each new incoming call. Automatic overflow only applies to new calls entering a queue: calls already in the queue are not transferred to a target queue. Priority calls that are overflowed to another queue retain their priority status in the target queue, and the various treatments (RAN, music, etc.) specified for the source queue remain in effect for each call, even though it is answered in the target queue.

4.02 Three threshold levels must be established for each ACD-DN involved in 'automatic overflow':

- | Calls Waiting Threshold (**CWTH**)
- | Busy Threshold (**BYTH**)
- | Overflow Threshold (**OVTH**).

4.03 These thresholds correspond to the number of calls waiting and define the boundaries of four operating ranges for each ACD-DN queue. The four ranges, in ascending order, are light, normal, busy, and overflow. The threshold levels are set for each ACD-DN during installation and can be modified by service change or load **management**.

4.04 The first threshold (**CWTH**) sets the upper limit on the 'light' range. When this threshold is exceeded, the CALLS WAITING and DISPLAY QUEUE lamps light steadily, and the queue is in its 'normal' operating range. If a second threshold (**BYTH**) is exceeded, the CALL WAITING lamps flash slowly, and the queue is in its 'busy' operating range.

4.05 If a third threshold (**OVTH**) is exceeded, the CALLS WAITING lamps flash quickly, and the queue is put into its 'overflow' state. At this point, a target queue is selected from the list of overflow queues assigned to the source queue. The overflow queues are considered one at a time, and the first one which is operating in the 'light' or 'normal' range (i.e., BYTH or lower) is selected as the target queue. The call is then placed in the target queue and does not return to the source queue. Selection of a target queue is performed for each new call coming into the source queue. Thus, if a target queue goes beyond its 'normal' range (**BYTH** exceeded), then another queue may be selected as a target queue. If an available target queue is not found, the call is returned to its original position in the source queue.

4.06 Source and target queues must be in the same ACD machine. Overflow to a target queue outside the SL-1 is described under 'INTERFLOW Key', and is useful when automatic overflow mechanisms are no longer adequate.

4.07 The value that can be assigned to each threshold is independent of the value assigned to any of the other thresholds. This allows the user considerable flexibility in configuring queues to match the requirements of the installation.

4.08 For example, a customer may want to give recorded announcements and music to all calls except those coming in on WATS trunks, and yet only have one ACD-DN for answering all calls. This can be done by creating two queues. All incoming trunks, except WATS, can be assigned to auto-terminate on ACD-DN XXXX, which has agents assigned to it, as well as RAN and music, if RAN is defined in the original ACD queue. A WATS trunk can be assigned to auto-terminate on ACD-DN YYYY which has no agents, RAN, or music, but which has an OVTH set to 0, and has ACD-DN XXXX as the target queue. A physical set must exist and be in the not ready state for the WATS ACD-DN YYYY to overflow to ACD-DN XXXX. (Treatments for YYYY source queue remain, as the call goes to the target queue XXXX.) Thus, WATS calls are answered on the same ACD-DN as other calls, but receive no answer supervision until the call has actually been answered by an agent. Other calls receive RAN and music as required. Figure 4-1 illustrates the automatic overflow decision process.



Call Forcing

4.09 Call forcing is an optional alternative to standard manual answering. This feature is intended to speed up the processing of ACD calls by automatically presenting the call in an answered state. Thus, if the call forcing option is in effect, the IN-CALLS key need not be pressed to answer the presented call. If the agent waits for the caller to disconnect, the agent is then automatically returned to the agent queue.

4.10 An ACD call answered under call forcing can be terminated in one of the following ways:

- 1 If the caller releases, and disconnect supervision is provided, the agent is returned to the agent queue automatically after a **2-second** delay and a **500-ms** tone burst (to permit the agent to press another key if desired).
- 1 If post-call processing is not required, the agent operates the IN-CALLS key to force a disconnect and is immediately inserted into the agent queue.
- 1 If post-call processing is required, the NOT READY key is operated. When post-call work is done, the IN-CALLS key is operated, and the agent is inserted into the agent queue.

4.11 In all cases, if there is an ACD call in the call queue, it is presented to the agent immediately after a **500-ms** tone in the headset. No ringing tone is heard from the set speaker.

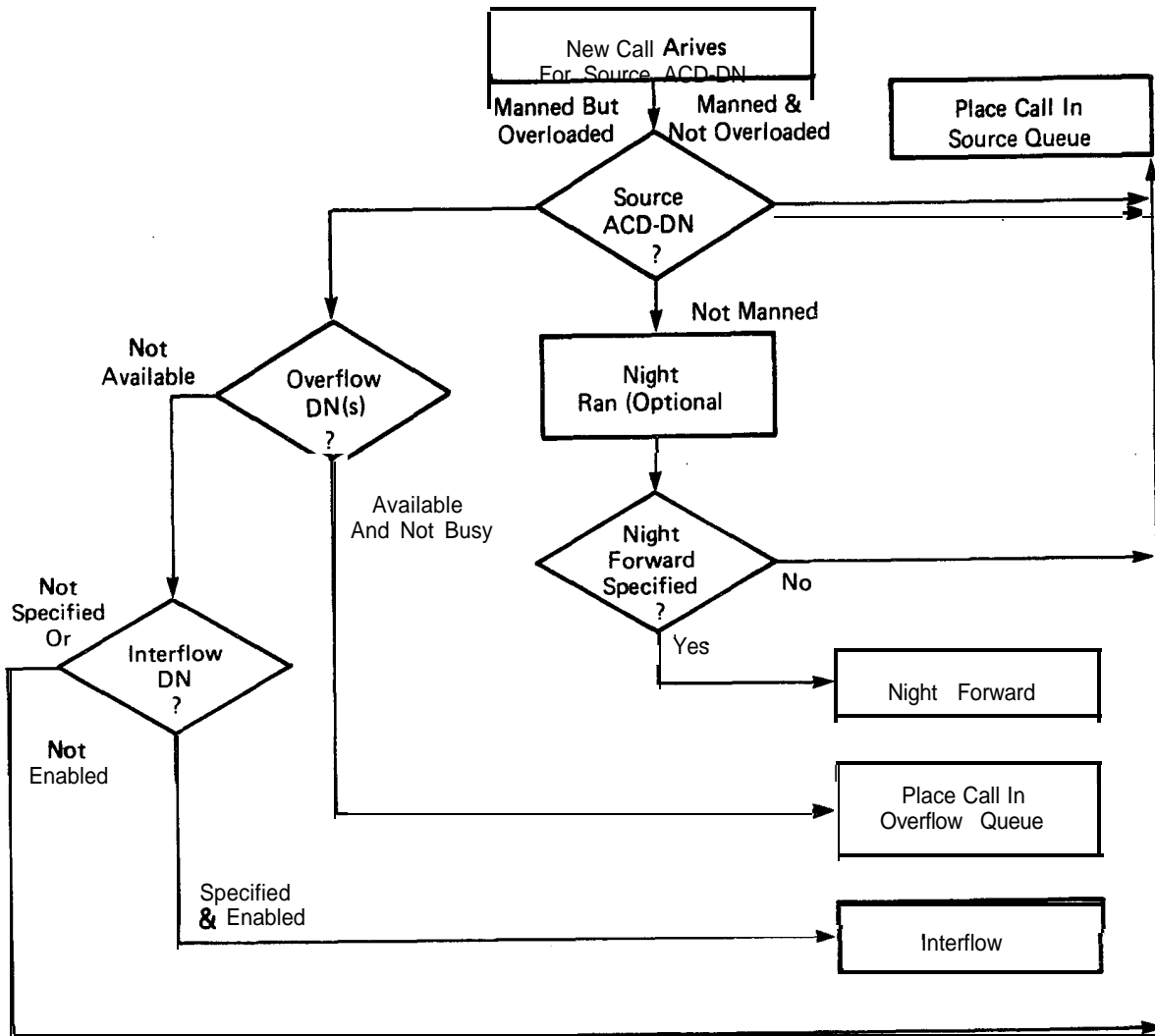


Fig. 4-1
Automatic Overflow Decision Process

4.12 Call forcing can be used with agent sets equipped with either a headset or a handset:

- **Headset Operation.** If an agent set is equipped with a headset or plug-in handset, the set remains in the off-hook state as long as the headset or handset is plugged into the lower jack pair on the left-hand side of the set, and call forcing operates as just described. If the agent's headset is disabled or removed, the agent set is put into the 'make busy' mode.
- **Built-in Handset Operation.** As long as the handset remains off-hook, call forcing operates as just described. If the handset is placed on-hook, the agent set is put in the 'make busy' mode, and the MAKE BUSY lamp lights. If the MAKE BUSY key is subsequently depressed, the MAKE BUSY lamp goes out, the IN-CALLS lamp flashes, and tone ringing is heard when the next ACD call arrives. The agent picks up the handset, and answers the call. If the handset is picked up while the MAKE BUSY key is active, MAKE BUSY is de-activated, and the call is presented after a **2-second** delay and a **500-ms** tone.

4.13 If the agent wishes to initiate some action during an ACD call, yet does not want to release the call, then the call can be put on hold by operating the HOLD key. The IN-CALLS lamp flashes quickly while the call is on hold. To reestablish the original call, the IN-CALLS key is operated and the associated lamp becomes steadily lit. Other keys which put the incoming call on hold are CONFERENCE, TRANSFER, and SUPERVISOR.

Music On Delay

4.14 Music on delay is heard by callers in an ACD queue who are not hearing Recorded Announcement (RAN) or ringback tone, but are waiting in the queue for service. Music on delay is triggered by the end of each RAN heard by the caller. The music continues until a subsequent RAN is provided, or the call is answered or abandoned. ACD calls do not receive music on delay if RAN is not also specified, or the call was extended to the ACD by an attendant. Music on delay is provided optionally after the first or second RAN, and between subsequent RAN, until the call is either answered or abandoned.

4.15 The music for music on delay is obtained from a music source via a music trunk (specified in service change), and connected to a conference circuit pack. Callers experiencing ACD delay are bridged into the conference circuit via a listen-only path. Each music trunk is assigned to a specific conference loop (not necessarily dedicated to music), and each ACD-DN can be programmed for a different music source (if available).

INTEGRATED SERVICE!3 NETWORK**MERIDIAN SL-1*****AUTOMATIC CALL DISTRIBUTION****MANAGEMENT REPORTING:****DESCRIPTION AND INPUT/OUTPUT FORMAT**

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Reason for Reissue: To revise information on **Logout** and **DN** key restrictions. Changes are indicated by arrows in the Margin.

1. GENERAL

1.01 The Meridian SL-1 Automatic Call Distribution (**ACD**) feature provides a means of distributing a large number of incoming calls among a number of telephones (called ACD 'agent position'). The incoming calls are served on a **first-in** first-out basis, and are distributed to the available agent positions in such a manner that the agent position that has been idle the longest is presented with the first call. Read the following practices for more detailed information on the ACD feature:

553-2671-100	ACD Basic Features
553-2671-101	ACD Advanced Features
553-2671-103	ACD Load Management
553-2671-103	Appendix 1 ACD Load Management with the Auxiliary Data System (ADS).

1.02 The ACD Management Reporting feature provides the ACD customer with timely and accurate statistics relevant to the ACD operation. These statistics, in the form of periodic printed reports or ongoing status displays, enable the customer to monitor changing ACD traffic loads, levels of service being provided, etc., and implement corrective action, where required.

1.03 This practice describes the types of management reports and displays that are available, and provides the inputs required to request or schedule these reports.

AGENT-ID OPTION
(Generic **X11 R2+**)

1.04 This feature provides the ACD customer with the option of operating in either an Agent-ID mode or Position-ID mode. If the Agent-ID option is selected, ACD agents must enter a four-digit Agent-ID code (range 0001 to 9999) before access is allowed to ACD features (this is part of the ACD set '**Login**' procedure that must be performed before access to ACD features is allowed). Statistical performance data continues to be accumulated on a Position-ID basis, however, the Agent-ID code of the agent that used the particular position is reflected in the management reports.

ACD SET **LOGIN**
(Generic **X11 R2+**)

1.05 An SL-1 set equipped at each agent or supervisor position is prevented from receiving any ACD calls until the set is logged in by the individual occupying that position. Access to regular Private Branch Exchange (**PBX**) features, however, is allowed once a headset/handset is plugged into the set; e.g., to make an outgoing call with the DN key.

1.06 The IN-CALLS key on the set is pressed to start the **login** process. (Operation of the key is ignored by the system if a headset/handset is not plugged into the key.) Depending on whether or not the Agent-ID option is defined, two results are possible.

- (a) If the Agent-ID option is not defined, the LED associated with the NOT-READY key lights, and the ACD-DN and supervisor position ID to which the set is assigned, are shown in the digit display of the set. ACD features are now active on the set, and position manned time is being accumulated against the position.

- (b) If the Agent-ID option is defined then, after the IN-CALLS key is pressed, a special (interrupted) dial tone is heard. The 4-digit Agent-ID code must then be keyed in on the dial pad of the set. (The digits appear on the digit display of the set, if equipped.) The Meridian SL-1 validates the digits and,
 - | sends overflow tone to the agent, if the code is not inputted before normal digit timeout, or the code is invalid or out of range
 - | sends busy tone to the agent if another individual is already logged in with that Agent-ID code.

Note: The configuration record specifies the maximum number of agents that can be logged into the SYSTEM at any one time (prompt DAGT in overlay 17). If an agent, by attempting to log in, exceeds this threshold, **login** is not allowed and overflow tone is presented.

↗ **ACD SET LOGOUT**
(Generic X11 R2+)

1.07 Removing the headset/handset from a set that is currently logged in to the ACD operation terminates the access of that set to the ACD features. This assumes that all agents (on a queue basis) have been given the option of using headset/handset removals, or activating the MARE BUSY key. When there is no option, the MARE BUSY key must be used to **logout**, unless there is no MARE BUSY key available for the individual agent(s). This also assumes no calls are active or 'on hold when the headset/handset is unplugged or the handset is placed on-hook (see 'Walkaway' Feature). The moment an agent has logged out the following events occur:

- | the agent position is removed from the ACD agent queue
- | all timing against that position stops.



1.08 If the ACD set is equipped with a MARE BUSY key, pressing the MARE BUSY key will cause a **logout**. In this case, the **logout** occurs immediately if there is no active call on the IN-CALLS key, or occurs as soon as the active call on the IN-CALLS key is disconnected.

WALKAWAY/RETURN
(Generic X11 R2+)

1.09 This feature involves the use of the HOLD key in conjunction with the 'log-in' feature. it permits an agent who is logged in to leave the position (**walkaway**) for a period of time, then return to the position and resume normal operation without going through the **login** procedures again; i.e., the headset/handset can be unplugged from the set without initiating a **logout**.

1.10 The 'Walkaway' feature is in effect under the following conditions:

- (a) Direct Call-Processing (**DCP**) phase (IN-CALLS LED lit). In this phase of operation, voice communication is established between the agent and a customer. **Prior** to removal of the headset/handset, the agent operates the HOLD key to prevent release of the customer from the connection. This causes the IN-CALLS LED to flash at 120 ipm.
- (b) Post Call-Processing (**PCP**) phase (NOT READY LED lit). In this phase of operation, the customer has already been released from the agent position. Operation of the HOLD key causes the NOT READY LED to flash at 120 ipm.

- (c) Non-ACD call-handling phase (**DN LED lit**). In this condition, a non-ACD call is connected to the position. Operation of the HOLD key retains the call connection, and causes the DN LED to flash at 120 ipm.

1.11 In (a), (b), and (c), timing is continued during the **walkaway** period in the same category as existed before the **walkaway** occurred. Removal of the headset/handset under any condition other than (a), (b), or (c) causes a **logout** to be initiated for the position.

Note: If the incoming caller disconnects from a held IN-CALLS or DN key, the set reverts automatically to the NOT READY state (i.e., Not Ready LED flashes), timing for the prior state stops, and PCP timing commences.

1.12 When an agent/supervisor returns to a position which is in the 'Walkaway' mode, and plugs the headset/handset into the set, normal operation of ACD features resumes.

AGENT USE OF DN KEYS

1.13 Activation of any DN key will light the associated LED **indicator** ↖ lamp and give dial tone, while an incoming call to that DN will cause the LED indicator to flash along with audible ringing on the agents **set**. This assumes that any and all agents on a queue basis are permitted to use DN keys or the agents is/are already logged in. If the agent is not permitted to use DN keys when not logged in, the LED for incoming DN calls will flash. However, the agent will not be able to answer the calls until logged in to the queue. ↙

2. MANAGEMENT REPORTING

**MANAGEMENT
REPORT TERMINALS**

2.01 Statistical data on the ACD operation can be retrieved from the Meridian SL-1 via either a Cathode-Ray Tube (CRT) terminal or Teletypewriter (TTY) at an ACD supervisor or senior supervisor position. (The type of terminal used must be **EIA RS-232-C** compatible, and support the standard ASCII character set) Typically, the senior supervisor position and supervisor positions are equipped with CRT terminals, and a hard-copy printer is provided for the printing of the periodic management reports.

2.02 ACD Supervisor Terminal. Any ACD supervisor position equipped with a CRT terminal (or TTY) is presented with an ACD ongoing status display. This display is updated at specified (through service change) intervals of either 30 or 60 s, and reflects the current status of any combination of ACD-DN (queues). In addition to the ongoing status display, a supervisor can query the schedules and printing options (defined by the senior supervisor) for the periodic management reports. A customer can equip more than one **supervisor** position with a CRT terminal.

2.03 ACD Senior Supervisor Terminal. An ACD senior supervisor position equipped with a CRT terminal can, in addition to performing any supervisor function, query, define, or change the printing schedules and options for the periodic management reports. (The periodic report schedules and options can also be defined through service change overlay program 23.) Only one ACD senior supervisor position is allowed per ACD customer.

**PERIODIC
MANAGEMENT
REPORTS****Scheduling**

2.04 The periodic management reports can be scheduled to be printed regularly on the hour, on the half-hour (or on the quarter hour (report 3 for Generic X11 Release 3 or later). When printed, the reports contain statistics accumulated since the last report printing. The periodic management **reports** are:

- | Agent Group Report (Report **1**)
- Queue Report (Report **2**)
- Trunk Routes Report (Report **3**)
- | Agent Position Report (Report **4**).

Warning Messages

2.05 Each printed periodic report (Fig. **2-1**) contains a heading (ACD customer number, date, time), and warning messages (if any), followed by the reports that have been specified by the senior supervisor (or through service change). One or more of the following warning messages may appear after the report heading:

- (a) ****SCHED CHG****. Schedule Change. This message is printed if the periodic reporting schedule is changed (by the senior supervisor or by service change) since the last reporting period.

- (b) ****INIT****. Initialize. This message is printed if an initialization occurred since the last reporting period. The periodic reports are not printed because all data was obliterated by the initialization. The accumulation areas are zeroed (cleared) to ensure accurate statistics at the next reporting period.
- (c) ****PER GT HR****. This message is printed if the time of data accumulation since the last reporting period exceeds one hour (e.g., if the reporting schedule is set at **08:00** to 16:00 daily then each **08:00** report carries this warning message). This message is important because it indicates that data in some of the reported fields may be misleading, e.g., the 'average agents available' field is calculated assuming a maximum interval of 60 min. Similarly, other fields which involve the accumulation of elapsed time may have overflowed and, thus, are reported inaccurately. Fields which are affected are 'busy time', position manned time', and 'CCS usage'.

Note: If the calculation of a field requires division, and the resulting number is greater than five digits, asterisks(*) are printed instead. This also occurs when the calculation requires division by zero.

Report Data

2.06 Tables 2-A through 2-D provide a description of the data contained in each field of the Agent Group (report 1), Queue (report 2), Trunk Routes (report 3), and Agent Position (report 4) reports respectively as shown in Fig 2-1.

Note: Data shown in the 'calls accepted' field is based on the following

- | If a call is Night Forwarded, it is counted in the 'interflow' field for the source ACD-DN in the Queue report.
- If the Night Forwarded number is an ACD-DN, then call accepted, answered, abandoned, etc., is reflected in the count for the destination ACD-DN. The call is not counted as call accepted, etc., against the source ACD-DN.
- | If a call is not Night Forwarded (whether or not Night **RAN** is given), it counts as call accepted, answered, etc., against the source ACD-DN. It will not count under 'interflow' in this case.

Calls per ACD DN

2.07 The total number of calls per ACD DN equals the total number of INC CALLS for all trunk routes terminating on the ACD DN. If there is more than one trunk route the INC CALLS for all trunk routes must be added together. See Fig 2-1 for INC CALLS per trunk route.

Calls Delayed per ACD DN

2.08 The number of calls delayed is determined from the ANSWERED AFT-T field (Fig 2-1). When the value of T is set to a minimal value such as 1, all calls encountering any delay are counted as having waited. The definition of calls that waited can be adjusted by changing the value of T.

2.09 The ABANDONED AFT-T field (Fig. 2-1) indicates the number of calls that were abandoned because of the delay.

Total Trunk Usage per ACD DN

2.10 The total trunk usage is indicated by the INC CCS field. See Fig. 2-1.

ACD 000 1980 02 01 15:00

REPT 1

ACD	AVG CALLS	AVG DCP	AVG PCP	AVG WORK	AVG WAIT	AVG CALLS	DN AVG TIME	AVG-TIME-POSN
DN	AGTS	ANSWD	ASA	DCP	PCP	WORK	WAIT	BUSY MANNED
2222	1	11	7	41	5	46	127	0 **** 152 431
5555	1	14	12	25	1	26	77	0 **** 180 396

REPT 2

ACD	CALLS	ABANDONED	TSF	OVER	INTER	DELAY-ANN	THRESHOLDS
DN	ACCPD	NO.	AVG.WT	TSF	T FLOWS	FLOWS	1ST 2ND CWTH BYTH OVTH
2222	11	0	****	100	100	0	0 0 2 3 4
5555	15	1	68	100	100	0	4 0 3 4 5

REPT 3

ROUTE	NO-OF-TRUNKS	INC	INC	OUT	OUT	ABANDONED	ANSWERED
CODE	ASSN	WORK	HPR	CALLS	CCS	CALLS	CCS
c0-9	5	5	2	0	0	0	0

REPT 4

POS	AGT	CALLS	AVG	AVG	AVG.	DN	INC	DN	OUT	BUSY	MANNED
ID	ID	ANSWD	DCP	PCP	WAIT	INC	TIME	OUT	TIME	TIME	TIME
506	1111	2	28	0	67	0	0	0	0	278	412
507	1112	2	67	0	165	0	0	0	0	138	436
508	1113	2	24	9	65	0	0	0	0	298	436
509	1114	3	35	7	147	0	0	0	0	126	436
510	1115	2	57	5	179	0	0	0	0	126	436
501	1116	0	****	****	****	0	0	0	0	236	238
502	1117	4	13	0	37	0	0	0	0	260	436
503	1118	4	17	1	25	0	0	0	0	310	436
504	1119	2	52	2	211	0	0	0	0	110	436
505	1120	4	30	1	102	0	0	0	0	132	436

Fig. 2-1
Format of a Typical Periodic Management Report

Table 2-A
 AGENT GROUP REPORT **FIELD** DESCRIPTIONS

FIELD TITLE	DESCRIPTION
ACD DN	Identification. The ACD-DN (up to four digits) is given to identify the queue.
AVG AGTS	Average Agents Available. The sum of all the position manned times, divided by the length of the reporting interval. This statistic is meaningless in reporting periods that last longer than one hour.
CALLS ANSWD	Incoming calls Answered. Peg count of the number of calls entered in the queue and actually answered by an agent (see Note in 2.06).
ASA	Average Speed of Answer. This is the sum of all waiting times for all calls coming into the queue (and eventually answered), divided by the number of incoming calls answered on this queue.
AVG DCP	Average Direct Call-Processing (DCP) Time. Average time (in seconds) that each agent spent handling ACD calls. Handling time is measured as the time from initial answer of the call to final release of the call. Average DCP time is the sum of all handling times, divided by the number of incoming calls answered by the group.
AVG PCP	Average Post Call-Processing (PCP) Time. Average time (in seconds) that each agent spent 'not ready' per incoming call. PCP time is measured from the time at which the agent goes into 'not ready' from DCP, until any event which removes the agent from 'not ready'; e.g., pressing the IN-CALLS key. The average PCP time is the total time accumulated against all NOT READY keys, divided by the total number of ACD calls answered on this queue.
AVG WORK	Average Work Time. Average time (in seconds) that each agent required to serve incoming calls. This includes direct call-processing time.
AVG WAIT	Average Waiting Time. This is the sum of all time that an agent was available to receive an ACD call, divided by the number of incoming calls answered. This statistic is meaningless if the number of calls answered does not exceed the average number of agents available.
DN CALLS	Number of DN Calls A peg count of the number of times that agents initiated or received a call on their individual DN keys. It should be noted that if an agent uses the conference key to initiate an outside call while active on the IN-CALLS key, it is not reported as an outside call.
AVG TIME	Average Outside Call Time. This is the sum of all times from the initial selection of the individual DN key to final release of the call, divided by the number of outside calls. The system only accumulates call time for one outside call per agent position at a time. This means that if an agent position has more than one DN key and the agent uses both at once, the reported DN call time will not be accurate. Agent sets should be configured with only one DN key unless the customer is willing to forego the accuracy of DN call statistics.

Table 2-A Continued
AGENT GROUP REPORT FIELD DESCRIPTIONS

FIELD TITLE	DESCRIPTION
AVG-TIME-POSN BUSY MANNED	<p data-bbox="521 443 1455 583">Average Position Busy and Manned Time. The average position busy time is the sum of all position manned times, minus the sum of all waiting times, divided by the number of positions which had any position manned time accumulated against them. A position is considered manned whenever one of the following conditions exist:</p> <ul data-bbox="521 611 1455 762" style="list-style-type: none"><li data-bbox="521 611 927 642"> The MAKE BUSY lamp is dark<li data-bbox="521 667 1360 699"> The MAKE BUSY lamp is lit and the position is active on any call<li data-bbox="521 724 1024 756">● The position is in the 'not ready' state.

Table 2-B
 QUEUE REPORT FIELD DESCRIPTIONS

FIELD TITLE	DESCRIPTION
ACD DN	Identification. Shows the ACD-DN (up to four digits) of the queue being reported.
CALLS ACCPD	Total Number of Calls Accepted. This is the total number of calls which were placed in this ACD-DN call queue, including any overflows from another queue (see Note in 2.06).
ABANDONED NO. AVG.WT	Number of Abandons and Average Waiting Time of Each. A peg count of the total number of calls for this queue in which the caller disconnected before being answered by an agent. The average waiting time of each abandoned call is the sum of all waiting times for abandoned calls, divided by the number of calls abandoned in this queue.
TSF	Telephone Service Factor. The TSF is a measure of how quickly incoming calls are being answered. A time ('T' seconds) is specified by the customer either in service change or by using 'load management'. The system keeps track of the percentage of incoming calls that are answered or abandoned before 'T' seconds. This figure is the TSF. A value of 100 means that all calls were answered or abandoned within 'T' seconds.
TSF T	Value of 'T' (in seconds) used for the Telephone Service Factor calculation.
OVER FLOWS	Calls Overflowed to Another Queue. A peg count of the total number of calls which were re-directed to another ACD-DN by using 'automatic overflow'.
INTER FLOWS	Calls Interflowed to Another Customer. A peg count of the total number of calls which were removed from this queue, and directed to another customer via the interflow mechanism (see Note in 2.06).
DELAY-ANN 1ST 2ND	First and Second RAN Count. Peg counts of the number of calls which received first and second recorded announcement treatment during the reporting period.
THRESHOLDS CWTH BYTH OVTH	Automatic Queue Overflow Thresholds. These are the three thresholds associated with 'automatic call queue overflow' (see 553-2671-101): Call-Waiting threshold, Busy threshold, and Overflow threshold.

Table 2 - C
TRUNK ROUTES REPORT FIELD DESCRIPTIONS

FIELD TITLE	DESCRIPTION
ROUTE CODE	Trunk Route Identification. Give the trunk route access code and the trunk route type: i.e., CO,FX, WATS * all generics. (DID, CCSA, TIE Generic X11 Release 3 and later.)
NO-OF-TRUNKS ASSN WORK HPR	<p>Number of Trunks. This statistic shows a snapshot of the numbers of trunks assigned, working, and set as priority at the time of report generation:</p> <ul style="list-style-type: none"> ASSN. The number of trunks (including non-ACD trunks, if any) currently assigned to the trunk route WORK. The number of trunks (including non-ACD trunks, if any) that are currently enabled HPR. The number of trunks currently given 'priority'.
INC CALLS	Total Number of Incoming Calls Offered. This is the total number of calls that came in on this trunk route (including non-ACD calls, if any) during the report period.
INC CCS	Total Incoming Traffic. This statistic gives the total incoming trunk traffic for the trunk route (in CCS) between seizure and disconnect (including non-ACD calls, if any).
OUT CALLS	Total Outgoing Calls. The total number of calls outgoing on this route during the report period. By definition, these are non-ACD calls, but could include outgoing calls made from the DN keys of the ACD agent positions.
OUT CCS	Total Outgoing Traffic. This figure shows the outgoing call traffic (in CCS) on the trunk route.
ABANDONED BEF-T AFT-T	Calls Abandoned. Two peg counts of calls abandoned showing how many were abandoned before and after the TSF time 'T'. The value used for 'T' depends on the trunk involved and on which queue that trunk terminates.
ANSWERED BEF-T AFT-T	Number of Calls Answered. Two peg counts of the number of calls answered before and after the time 'T' used for TSF. The value used for 'T' depends on the trunk involved and on which queue that trunk terminates.

Table 2-D
 POSITION REPORT FIELD DESCRIPTIONS

FIELD TITLE	DESCRIPTION
POS ID	Identification. Shows the ACD PGS-ID of the agent being reported.
CALLS ANSWD	Incoming Calls Answered at this Position. A peg count of the number of calls answered at this position in the reporting period.
AVG DCP	Average Direct Call - Processing (DCP) Time. Average time, in seconds, that this position spent handling incoming ACD calls. Handling time is measured as the time from initial answer of the call to final release from an agent. The average DCP time is the total of all the handling times, divided by the number of calls answered at this position.
AVG →PCP	Average Post Call-Processing Time. This is the average time, in seconds, that this position spent in the 'not ready', or ' login ', state related to each ACD call answered by that position.
AVG WAIT	Average Waiting Time. This is the sum of all times that the position was available to receive ACD calls, divided by the number of ACD calls answered at this position.
DN INC	Total number of incoming calls on the agent's DN key(s) during the report period.
INC TIME	Total duration (in seconds) of all incoming calls on the agent's DN key(s) during the report period, timed from answer to final release of the call.
DN OUT	Total number of outgoing calls on the DN key(s) during the report period.
OUT TIME	Total time (in seconds) of all outgoing calls on DN key(s) during the report period, timed from the initial seizure of the DN to final release for each call.
BUSY TIME	Total Position Busy Time. This is the total position manned time, minus the total waiting time (in seconds).
MANNED TIME	Total Position Manned Time. This figure is the total of all the following times for the position in the reporting period. A position is considered manned when one of the following conditions exist: <ul style="list-style-type: none"> The MARE MARE BUSY lamp is dark The MARE MARE BUSY lamp is lit and the position is active on any call The position is in the 'Not Ready' state.

**ONGOING STATUS
DISPLAY**

2.11 The ongoing status display (Fig. 2-2) is presented on the CRT display, and updated either every 30 s or every 60 s (as specified through ACD service change). Table 2-E describes the fields shown in the display.

ACD			#CALLS	#POS	#POS	#POS	#POS	#POS
DN	TSF	ASA	IN QUEUE	MANNED	DCP	PCP	WTG	NON-ACD
2222	100	2	0	1	0		1 0	0
5555	46	120	8	1	1 0		0	0

ACD			#CALLS	#POS	#POS	#POS	#POS	#POS
DN	TSF	ASA	IN QUEUE	MANNED	DCP	PCP	WTG	NON-ACD
2222	100	2	0	1	0	1	0	0
5555	45	184	7	2	2	0	0	0

ACD			#CALLS	#POS	#POS	#POS	#POS	#POS
DN	TSF	ASA	IN QUEUE	MANNED	DCP	PCP	WTG	NON-ACD
2222	100	2	0	1	0	1	0	0
5555	44	242	6	3	3	0	0	0

ACD			#CALLS	#POS	#POS	#POS	#POS	#POS
DN	TSF	ASA	IN QUEUE	MANNED	DCP	PCP	WTG	NON-ACD
2222	100	2	0	3	0	1	2	0
5555	44	296	5	4	4	0	0	0

ACD			#CALLS	#POS	#POS	#POS	#POS	#POS
DN	TSF	ASA	IN QUEUE	MANNED	DCP	PCP	WTG	NON-ACD
2222	100	2	0	- 3	1	1	1	0
5555	44	296	5	4	4	0	0	0

ACD			#CALLS	#POS	#POS	#POS	#POS	#POS
DN	TSF	ASA	IN QUEUE	MANNED	DCP	PCP	WTG	NON-ACD
2222	100	3	0	3	2	1	0	0
5555	44	296	5	4	4	0	0	0

Fig. 2-2
Format of a Typical Ongoing Status Display

Table 2-E
ONGOING STATUS DISPLAY FIELD DESCRIPTIONS

FIELD TITLE	DESCRIPTION
ACD DN	Identification. Shows the ACD-DN of the queue being reported.
TSF	Telephone Service Factor. This is the percentage of calls answered or abandoned before the time 'T' (in seconds) programmed for this queue. It is obtained from the total peg counts accumulated since the last periodic report time.
ASA	Average Speed of Answer. This is the average time, in seconds, that each answered ACD call had to wait for an answer. It is obtained from the total (cumulative) counts since the last periodic report time.
#CALLS IN-QUEUE	Number of Calls in Queue. This is a snapshot of the number of calls awaiting service in this queue, but not yet presented to an agent position.
#POS MANNED	Number of Manned Positions. The number of agent positions associated with this queue for which the MARE BUSY lamp is dark.
#POS DCP	Agents Direct Call-Processing. The number of agent positions currently active on their IN-CALLS key (including those which have put their ACD calls on 'hold').
#POS PCP	Agents Post Call-Processing. Snapshot of the number of agent positions currently in 'not ready' state.
#POS WTG	Positions Waiting. Snapshot of the number of agent positions currently available to receive an incoming ACD call.
#POS NON-ACD	Positions on Non-ACD Calls. Snapshot of the number of agents active on any key other than the IN-CALLS key.

Agent-ID Reporting
(Generic **X11 R2+**)

2.12 If the Short Report and Agent-ID options are defined, the hourly/half-hourly agent position report (Report 4) shows the Agent-ID of the agent that occupied the position while the statistics were accumulated. A new field heading (AGT ID) is added to Report 4 to accommodate this information. The Agent-ID is followed by an asterisk (*) if an agent with a different Agent-ID logs in to the position during the reporting interval.

2.13 If, during a reporting interval, a different agent logs in to a position that was manned previously, a one-line report, termed a Short Report, is outputted (if so specified in the ACD data block, overlay 23). This one-line of statistics reflects data accumulated for the previously logged in agent at that position. Statistics reported for that position in the hourly/half-hourly position report reflect only data accumulated for the currently logged on agent. Headings of the Short Report are identical to those for Report 4.

System Totals (Generic
X11 **R2+**)

2.14 A new line of statistics is printed at the end of each hourly/half-hourly report for Reports 1, 2, 3 and 4. This line of statistics represents a system total of all data contained in each report. A line of dashes separates these statistics from the rest of the report.

2.15 Data shown for system totals represent totals of all agents, all times, all calls, etc., for all **ACD-DN/Pos-ID** in the customer's ACD operation. Rather than a single line of statistics for each **ACD-DN/Pos-ID**, the system total line of statistics reflects the total number of **ACD-DN/Pos-ID** under the **ACD-DN/Pos-ID** field heading.

2.16 Certain data are not reflected in the system total line of statistics, too, some data are expressed differently (e.g., minutes rather than seconds). Following is a summary of those statistics not reported, and those statistics expressed differently.

- (a) Report 1 - Agent Group. The number shown in the ACD DN field reflects the total number of ACD-DN. Data for the AVG AGTS field is not reported.
- (b) Report 2 - Call Queue. The number shown in the ACD DN field reflects the total number of ACD-DN. Data for the TSF T and THRESHOLDS CWTH **BYTH** OVTH fields is not reported.
- (c) Report 3 - Trunk Routes. The number shown in the ROUTE CODE field reflect the total number of trunks that auto-terminate on all ACD-DN in the customer's operation. Data for the NO OF TRUNKS ASSN WORK HPR field is not reported.
- (d) Report 4 - Agent Position. The number shown in the POS ID field represents the total number of Pos-ID in the customer's operation. Data for the INC TIME, OUT TIME, BUSY TIME, and MANNED TIME fields is expressed in minutes. The AGT ID field is not reported.

Daily Totals (Generic
X11 **R2+**)

2.17 The customer can schedule Reports 1, 2, and 3 to be outputted at the end of each daily period Report printout format is the same as that for the hourly/half-hourly reports. Data shown in the daily reports reflects data accumulated over the daily reporting period for both totals and averages. The daily reporting period is defined as the time between the specified start time and end time of the reports. To accommodate 24-hr reporting, start and end times of 0 hours are supported (i.e., daily reports will be printed at midnight). The system total line of statistics is included as part of each daily report.

2.16 Refer to 553-2671-103 for new/changed load management commands that are available to control these reports.

3. MANAGEMENT REPORTING COMMANDS

ACCESSING THE
COMMAND MODE

3.01 The periodic management reports **can** be selected and scheduled for regular printing by the senior supervisor. (Supervisors can query the printing schedules and options, but cannot alter them.)

3.02 Access to the management reporting command mode is accomplished by inputting '**\$L**' on the CRT keyboard followed by a carriage return. The system responds with the prompt '>' to indicate it is ready to receive management reporting commands.

3.03 When a management reporting command is inputted, the system responds by printing the current parameters (if any) associated with the command, followed by a double dash '- -'. The double dash is an indication that input will be accepted to change the existing parameters or add new parameters associated with the command. (Supervisors do not receive the double dash after inputting a management reporting command.)

3.04 If new parameters are to be associated with a command, they are inputted after, and on the same line as the double dash, followed by a carriage return. If one or more of the current parameters associated with a command are to be changed, the current parameters that are unchanged must be inputted as well as the changed parameter(s), after the double dash. Example: if current parameters are 1, 2, 4, and parameters 1, 2, 3 are desired, then 1, 2, 3 must be inputted.

Note: The management reporting commands that can be inputted at a CRT (or **TTY**) by the senior supervisor are shown in this practice in uppercase characters (e.g., **XXXX**); system responses are shown in lowercase characters (e.g., **xxxx**).

Scheduling Periodic
Reports

3.05 The Select Schedule (**SSCH**) command enables the senior supervisor to define a printing schedule for the periodic management reports. The format for the command is:

```
SSCH sd sm ed em - SDSMEDEM
sh eh s - SHEH S
d d d d d d - D D D D D D
```

where,

- sd** = the starting day (1-31)
- sm** = the starting month (1-12)
- ed** = the ending day (1-31)
- em** = the ending month (1-12)
- sh** = the starting hour (0-23)
- eh** = the ending hour (0-23)
- s** = the schedule code:
 - 0, means no reports are printed.
 - 1, means reports are printed hourly on the hour.
 - 2, means reports are printed hourly on the half-hour.
 - 3, means reports are printed on the half-hour.
 - 4, means Report 3 is printed every quarter hour. No other reports are printed.
 - 5, means Report 3 is printed every quarter hour. Other reports are printed hourly on the hour.
 - 6, means Report 3 is printed every quarter hour. Other reports are printed hourly on the half hour.
 - 7, means Report 3 is printed every quarter hour. Other reports are printed on the half hour.
- d** = the **day(s)** of the week on which the reports are to be printed:
 - 1, is Sunday
 - 2, is Monday
 - 3, is Tuesday
 - 4, is Wednesday
 - 5, is Thursday
 - 6, is Friday
 - 7, is Saturday

3.06 When the 15 min Report 3 option is selected, the data contained in this report reflects only the previous 15 min interval, regardless of the interval selected for other reports.

3.07 The 15 min Report 3 option should not be used in environments where the averagelength of calls approaches or exceeds 15 min.

3.08 Periodic Reports can also be scheduled by using service change overlay program 23.

Selecting Print Options

3.09 The Select Print (**SPRT**) commands enables the senior supervisor to specify which of the management reports appear in the scheduled printing The format for this command is

SPRT w x y z - W X Y Z

where,

w,x,y, and **z** are the reports to be printed:

- 1, for the Agent Group Report
- 2, for the Queue Report
- 3, for the Trunk Routes Report
- 4, for the Agent Position Report

Error Codes

3.10 Table 3-A lists the error codes that can appear as a result of incorrect management reporting commands or system faults.

Table 3-A
MANAGEMENT REPORTING ERROR CODES

CODE	MEANING
ACD001	noisy ACD terminal: disabled since noise threshold exceeded
ACD002	too many invalid characters entered on ACD terminal;
ACD003	disabled since invalid character threshold exceeded
ACD010	invalid character entered
ACD011	invalid input character: only blanks allowed
ACD012	input buffer size exceeded; field too long
ACD013	unknown command
ACD014	invalid day: valid range 1 to 31
ACD015	invalid month, valid range 1 to 12
ACD016	invalid day-month combination
ACD017	invalid hour; valid range 0 to 23
ACD018	invalid schedule; valid range 0 to 3
ACD019	invalid day of week: valid range 1 to 7
ACD021	same day cannot be re entered
ACD022	invalid digit; valid digits are 0 to 9 and *
ACD029	invalid digit; valid digits are 0 to 9
ACD030	same report option cannot be reentered
	invalid report option

INTEGRATED SERVICES NETWORK

MERIDIAN SL-1.

AUTOMATIC CALL DISTRIBUTION

LOAD MANAGEMENT:

DESCRIPTION AND INPUT/OUTPUT FORMAT

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Reason for Reissue: To move Note 2: from the Second RAN Route Assignment to the the correct location which is, Select Agent to Supervisor Assignment (**SATS**). Change arrows in the margin indicate the changes.

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1. GENERAL

1.01 The Meridian SL-1 Automatic Call Distribution (**ACD**) feature provides a means of sharing a large volume of incoming calls among several ACD agent positions. Incoming ACD calls are served on a first-in first-out basis, and are distributed among the available agent positions in such a manner that the agent position that has been idle the longest is presented with the first call. The ACD features are more fully described in the following practices:

553-2671-100	Description of ACD Basic Features
553-2671-101	Description of ACD Advanced Features
553-2671-102	Description of ACD Management Reports
553-2671-104	Description of ACD with an Auxiliary Data System (ADS).

1.02 The ACD Load Management feature provides customers having large ACD operations the capability to alter the ACD configuration (e.g., assignment of agents to ACD-DN) in response to varying ACD traffic loads. These changes would normally be made on the basis of statistics obtained from the supervisor's digit display, or an analysis of the printed Management Reports (where this option is chosen).

1.03 Each ACD customer can have one supervisor position designated as a 'senior supervisor' position. This position can be equipped with a Teletypewriter (TTY) or Video Display Terminal (**VDT**) for the purpose of Management Report functions (**553-2671-102**) and/or Load Management functions. The TTY or VDT is connected to the Meridian SL-1 through a standard Serial Data Interface (**SDI**) circuit pack.

1.04 When the TTY or VDT is used for Load Management functions, the senior supervisor can:

- | reassign auto-terminate ACD trunk routes to a different ACD-DN
- | reassign an ACD agent position(s) to another ACD-DN
- | redefine the ACD night forwarding number
- | reassign an ACD agent position to another supervisor
- | assign priority or non priority status to ACD trunks
- | set the timers and routes for first and second delay Recorded Announcements (**RAN**)
- | specify a night RAN route
- | set the target answer time 'T' used to calculate the Telephone Service Factor (**TSF**)
- | define the 'target' queues for automatic overflow
- | define the overflow thresholds

- | define the interflow digits
- | query the existing parameters of any of the above.

1.05 This practice addresses the input/output commands and system responses that are possible with the ACD Load Management feature.

Note: If the ACD-ADS feature is equipped, additional Load Management functions are possible. These additional functions are described in Appendix 1 to this practice.

2. LOAD MANAGEMENT COMMANDS

Accessing Command Mode

2.01 Access to the Load Management functions from a VDT or TTY by a senior supervisor is accomplished by inputting '\$L' on the keyboard, followed by a carriage return. The system responds with the prompt '>' to indicate it is ready to receive further commands. (If the ACD is equipped with the Management Reports feature, any current report display is aborted when '\$L' is inputted.) Once the '>' is received, the appropriate Load Management command can be inputted.

2.02 When a Load Management command is inputted, the system responds by printing/displaying the current data related to the inputted command, followed by a double dash (- -). If the senior supervisor wishes to change the existing data, the new data is inputted (after the double dash and on the same line), followed by a carriage return. If no change is required to the existing data, a carriage return leaves the current data unchanged and responds with the prompt '>' for further commands. To exit from the Load Management command mode, '\$L' is inputted after the prompt '>', followed by a carriage return.

Note: When changing a single data item in a line containing several data items, all data items preceding the data item to be changed must be 'changed' to their same value before changing the selected data item.

2.03 Any supervisor position equipped with a VDT or TTY can enter the Load Management command mode (by inputting '\$L', carriage return), and query existing parameters (by inputting a Load Management command), but cannot alter them. The system prints/displays the existing parameters, but does not print/display the double dash (-) to allow further data input. Exit from the mode is via the command '\$L'.

2.04 The following conventions are used in the Load Management commands shown here:

- | System outputs are shown in upper-case characters (XXXX); senior supervisor (and supervisor) inputs are shown in lower-case characters (xxxx)
- Agent positions are identified by their ACD Position-ID (POS-ID)
- | Trunk routes are identified by access code and member number
- | Queues are identified by ACD-DN.

Query Current Options

2.05 This command (POPT) enables the senior supervisor (or supervisors) to determine the current options that are in effect for each (or all) ACD-DN. For each ACD-DN specified, the following information is given:

- the First RAN Route Number (FRRT)
- the First RAN Route Time (FRTT)
- the Second RAN Route Number (SRRT)

- | the Second RAN Route Time (**SRTO**)
- | the Music Route (**MURT**)
- whether or not ‘call forcing’ (**FORC**) is in effect
 - | whether or not ‘observation tone’ (**OBTN**) is given to an agent when being observed by a supervisor
 - | whether or not the Separate Post Call-Processing (**SPCP**) option is in effect
- | the Night RAN Route (**NRRT**)
- | the Night Forwarding Number (**NITE**), if defined.

2.06 The format for this command is:

POPT xxxx xxxx xxxx

where,

xxxx = an up to four-digit ACD-DN. Up to six ACD-DN can be specified, or ALL can be specified to determine the options applicable to all ACD-DN in the customer’s operation.

Query Current Parameters

2.07 This command (**PPAR**) enables the senior supervisor or supervisors to query the current parameters associated with each (or all) ACD-DN. For each ACD-DN specified, the following information is given:

- | the maximum number of positions assigned (**MAXP**)
- | the Telephone Service Factor Time (**TSFT**), in seconds
- | the Calls Waiting Threshold (**TLDA**)
- | the Busy Threshold (**TLDB**)
- | the Overflow Threshold (**TLDC**)
- | the first overflow ‘target’ queue (**SQ01**)
- | the second overflow ‘target’ queue (**SQ02**)
- | the third overflow ‘target’ queue (**SQ03**)
- | the Interflow number (**IFDN**).

2.08 The format for this command is:

PPAR xxxx xxxx xxxx

where,

xxxx = an ACD-DN of up to four digits. Up to six ACD-DN can be specified, or ALL can be specified to determine the parameters in effect for all ACD-DN in the customer’s operation.

Select Route and Trunk Assignment

2.09 This command (**SRTA**) is used to assign the terminating ACD-DN for an auto-terminate ACD trunk. Changing the terminating ACD-DN for a trunk, affects only those calls that seize the trunk after the ACD-DN is changed. The input format for this command is:

SRTA RAC MEM xxxx - XXXX

where,

RAC = the route access code of up to 4 digits

MEM = the trunk member number (1-1261)

xxxx = the current ACD-DN to which the trunk is assigned

XxXx = the new ACD-DN (up to 4 digits).

Select Trunk Priority Assignment

2.10 The senior supervisor can assign individual ACD trunks to priority or non priority status with the following command:

SPRI RAC MEM x - X

where,

RAC = the trunk route access code (up to 4 digits)

MEM = the trunk route member number (1-1261)

x = the current priority assignment (0 = no priority, 1 = priority)

X = the new priority assignment (0 or 1).

Select Agent Position Assignment

2.11 The Select Agent Position Assignment (**SAPA**) command is used to change the ACD-DN (queue) to which an individual agent position is assigned. The agent position must go into or be in Make Busy mode before the new assignment becomes effective.

2.12 If execution of the command is delayed (e.g., because the agent position is not in Make-Busy mode), and at the time of execution there is no room in the ACD position list, an error message is printed/displayed and the transfer does not take place. More than one **SAPA** command can be outstanding at any one time. If more than one **SAPA** command is outstanding for a particular agent position, only the final command is effective. The format for this command is:

SAPA XXXX yyyy - YYYY

where,

XxXx = the applicable agent Position-ID
 YYYY = The ACD-DN to which the position is currently assigned
 YYYY = the new ACD-DN assignment.

Select Agent to Supervisor Assignment



2.13 This command (**SATS**) allows the senior supervisor to reassign an agent position from an AGENT key on one supervisor position to an AGENT key on another supervisor position. (AGENT keys are assigned through the SL-1 set overlay program, 11.) The format for this command is:

SATS XXXX yyyy zz - YYYY ZZ

where,

XXXX = the Position-ID of the agent position to be reassigned
YYYY = the supervisor Position-ID to which the agent position is currently assigned
zz = the AGENT key on the supervisor's telephone to which the agent position is currently assigned
YYYY = the new supervisor Position-ID (Note)
ZZ = the AGENT key on the new supervisor's telephone.

Note 1: Specifying 'X' for the new supervisor Position-ID, removes the agent from the current supervisor without assigning a new supervisor.

 Note 2: If a supervisor's telephone is not equipped with AGENT keys (option available on Generic XII Release 2 and later), an asterisk (*) must be input in place of the key number 77, whenever the SATS command is used to assign/reassign an agent to a supervisor.


First RAN Route Assignment

2.14 The First RAN Route Assignment (**FRRT**) command is used to specify (for each ACD-DN) the trunk route access code for the source of the first Recorded Announcement (**RAN**). The format for this command is:

FRRT XXXX yyyy - YYYY

where,

xxxx = the applicable ACD-DN (up to 4 digits)
 yyyy = the current first RAN route access code
 YYYY = the new first RAN route access code of up to 4 digits (Note).

Note: Specifying 'X' for the new RAN route access code, removes the first RAN feature from the ACD-DN.

First RAN Route Time

2.15 The First RAN Route Time (**FRTT**) command allows the senior supervisor to specify how long (in seconds) an incoming ACD call can remain unanswered in the queue before being given first RAN. The format for this command is:

FRTT XXXX yyyy - YYYY

where,

xxxx = the applicable ACD-DN (up to 4 digits)
 yyyy = the current first RAN time
 YYYY = the new first RAN time, in seconds (0-2044).

Second RAN Route Assignment

2.16 The Second RAN Route Assignment (**SRRT**) command is used to specify (for each ACD-DN) the trunk route access code for the source of the second RAN. The format for this command is:

SRRT XXXX yyyy - YYYY

where,

xxxx = the applicable ACD-DN (up to 4 digits)
 yyyy = the current second RAN route access code
 YYYY = the new second RAN route access code (Note).

Note: Specifying 'X' for the second RAN route access code, removes the second RAN feature from the ACD-DN. ←

Second RAN Route Time

2.17 The Second RAN Route Time (**SRTO**) command allows the senior supervisor to specify how long (in seconds), after being given first RAN, a call can remain unanswered before being given second RAN. The command format is:

SRTO XXXX yyyy - YYYY

where,

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XXXX = the applicable ACD-DN
 YYYY = the current second RAN time
 YYYY = the new second RAN time, in seconds (0-2044).

Night RAN Route
Assignment

2.18 The Night RAN Route Assignment (**NRRT**) command allows the senior supervisor to specify the trunk route access code for the source of night RAN. The format for this command is:

NRRT XXXX yyyy - YYYY

where,

XxXx = the applicable ACD-DN (up to 4 digits)
 YYYY = the current night RAN access code
 YYYY = the new night RAN access code of up to 4 digits
 (Note).

Note: Specifying 'X' for the new night RAN access code, removes the night RAN feature from the ACD-DN.

Night Forwarding
Number Assignment

2.19 The NITE command allows the senior supervisor to specify an up to 23 digit number to which ACD calls are to be forwarded when the ACD is in night service (i.e., all ACD telephones in Make Busy mode). The format is:

NITE XXXX yyyy...y - YYY...Y

where,

XXXX = the applicable ACD-DN (up to 4 digits)
 yyyy...y = the current night forward number
 YYY...Y = the new night forward number of up to 23 digits including the asterisk (*) to indicate a dialing pause, where required (Note).

Note: Specifying 'X' for the night forward number, disables night forwarding for the ACD-DN.

Automatic Overflow
Target DN

2.20 The following commands allow the senior supervisor to define or change (for each ACD-DN) the up to three target DN for the automatic overflow feature. The formats of the commands are:

SQ01 wwwww xxxx-XxXx
SQ02 wwwww yyyy-YYYY
SQ03 wwwww zzzz- ZZZZ

where,

WWWW	=	the ACD-DN of the source queue (up to 4 digits)
xxxx	=	the current first target ACD-DN
XXXX	=	the new first target ACD-DN (Note)
yyyy	=	the current second target ACD-DN
YYYY	=	the new second target ACD-DN (Note)
zzzz	=	the current third target ACD-DN
ZZZZ	=	the new third target ACD-DN (Note).

Note: Specifying 'X' for a new target ACD-DN, removes it and moves the next choice forward, if applicable (e.g., if 'X' is specified for the target ACD-DN for **SQ02**, the target ACD-DN specified for SQ03 becomes the SQ02 target ACD-DN).

Automatic Overflow Thresholds

2.21 The TLDA, TLDB, and TLDC commands allow the senior supervisor to adjust the calls waiting, busy, and overflow thresholds respectively. Note the following:

- 1 Specifying 'X' as the new threshold for TLDA, reduces the threshold to 1, thus lighting the CALLS WAITING lamp when any call is waiting.
- 1 Specifying 'X' as the new threshold for TLDB, reduces the threshold to 0, thus denying acceptance of any calls overflowed from another ACD-DN.
- 1 Specifying 'X' as the new threshold for TLDC, increases the threshold to its maximum value (**2047**) and prevents any calls from overflowing out of the ACD-DN.

2.22 The overflow threshold commands are formatted as follows:

TLDA XXXX yyyy - YYYY

TLDB XXXX yyyy - YYYY

TLDC XXXX yyyy - YYYY

where,

XXXX = the applicable ACD-DN (up to 4 digits)

YYYY = the current threshold value

YYYY = the new threshold value (0 to 2047).

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Setting Interflow DN

2.23 This command enables the senior supervisor to specify (for each ACD-DN) the DN to which ACD calls are to be routed when the interflow feature is activated. The command format is:

IFDN XXXX yyy...y - YYY...Y

where,

XxXx = the applicable ACD-DN (up to 4 digits)

yyy...y = the current interflow DN

YYY...Y = the new interflow DN of up to 23 digits including the asterisk (*) to indicate a dialing pause (Note).

Note: Specifying 'X' for the new interflow DN, disables the interflow feature for the ACD-DN, even though an INT'ERFLOW key might still be assigned on a supervisor position.

Telephone Service
Factor Time

2.24 This command allows the senior supervisor to set/change the value of 'T' (in seconds) for the TSF. The command format is:

TSFT XXXX yyy - YYY

where,

XxXx = the applicable ACD-DN (up to 4 digits)

YYYY = the current value of 'T'

YYY = the new value of 'T' (1 to 510 seconds).

Daily-System Totals

2.25 Two additional commands are available with Generic X11 Release 2. They are available to a senior supervisor which enable statistics for daily and system totals (553-2671-102) to be examined. The commands are DTOT (daily totals) and STOT (system totals).

2.26 DTOT. This command, followed by the desired report number (1, 2, and/or 3), presents a daily totals report on the senior supervisor's terminal. Statistics shown in the report(s) are those accumulated since the beginning of the current daily period to the current time, and includes system totals for each requested report. The Position Report (Report 4) is not accessible via this command. If a Meridian SL-1 initialization occurred in the reporting period, an IN100 message is presented preceding the DTOT report.

The command format is:

DTOT X

where,

X = report number 1, 2 or 3

2.27 STOT. This command, followed by the desired report number (1, 2, 3, or 4), presents a system totals report on the senior supervisor's terminal. Statistics in the report(s) are those accumulated since the last hourly/half-hourly report. If a Meridian SL-1 initialization occurred in the reporting period, a message so indicating is presented preceding the STOT report.

The command format is:

STOT X

where,

X = report number 1, 2, 3 or 4.

3. ERROR CODES

3.01 Table 3-A lists the error codes that can result from incorrect Load Management commands or system faults. They may be displayed while load management functions are being performed by a senior supervisor.

Table 3-A
ERROR CODES

CODE	MEANING
ACD001	noisy ACD terminal: disabled since noise threshold exceeded
ACD002	too many invalid characters entered on ACD terminal: disabled since invalid character threshold exceeded
ACD003	invalid character entered
ACD004	cannot transfer the reported agent to another queue
ACD010	invalid input character; only blanks allowed
ACD011	input buffer size exceeded; field too long
ACD012	unknown command
ACD013	invalid day; valid range 1 to 31
ACD014	invalid month; valid range 1 to 12
ACD015	invalid day-month combination
ACD016	invalid hour; valid range 0 to 23
ACD017	invalid schedule: valid range 0 to 3
ACD018	invalid day of week; valid range 1 to 7
ACD019	same day cannot be re-entered
ACD020	too many digits supplied for DN
ACD021	invalid digits; valid digits are 0 to 9 and *
ACD022	invalid digit; valid digits are 0 to 9
ACD023	DN supplied is not a valid ACD-DN
ACD024	invalid RAN time threshold or timeout
ACD025	invalid RAN route
ACD026	invalid threshold values
ACD027	queue overflow destination cannot be itself
ACD028	invalid value of 'T' in TSF
ACD029	same report option cannot be reentered
ACD030	invalid report option
ACD031	DN supplied is not a valid trunk route access code
ACD032	route does not auto terminate (for SRTA command, or no members for this route (for SRTA & SPRI commands)
ACD033	member number out of range
ACD034	invalid member number; for SRTA command it could also mean that the member selected did not have an auto-terminate ACD-DN defined
ACD035	invalid priority value
ACD036	no empty position for agent in new queue
ACD037	DN supplied is not a valid ACD Position-ID
ACD038	position selected must be an agent, not a supervisor
ACD039	an agent position can only be moved to a supervisor position, not another agent position
ACD040	invalid key number
ACD041	key number not reserved for agent position on supervisor's telephone
ACD042	cannot transfer an agent position, currently being observed by its supervisor, to another supervisor
ACD047	too many parameters for storage
AcDo48	too many ACD-DN; maximum is 6

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Table 3-A Continued
ERROR CODES

CODE	MEANING
CODES ACD049–ACD055 APPLY TO GENERIC X11 RELEASE 2 AND LATER.	
ACD049	Unable to fulfill request - - system error
ACD050	Data not available because of initialization
ACD051	Request for DTOT is outside of current daily reporting period.
ACD052	Unable to fulfill request - - another report is running
ACD053	Invalid input for report number
ACD054	No reports requested for DTOT/STOT
ACD055	DTOT/STOT only valid for senior supervisor
→ ACD056	High speed link does not exist
→ ACD057	High speed link is down

INTEGRATED SERVICES NETWORK**MERIDIAN SL-1.****AUTOMATIC CALL DISTRIBUTION
FEATURE ENGINEERING**

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1. GENERAL

1.01 The Automatic Call Distribution (**ACD**) feature allows a Meridian SL-1 to be arranged so that a number of SL-1 sets ■ called agent terminals ■ share equally in the answering of incoming calls made to one or more specified **ACD** Directory Numbers (**ACD-DN**). The incoming **ACD** calls are placed in an **ACD-DN** queue, and presented to agent terminals on a first-in, first-out basis. Similarly, agent terminals are placed in an agent terminal queue such that the agent terminal that has been available the longest is the first removed from the queue and presented with an **ACD** call. Further information on **ACD** can be found in the following practices:

- 553-2671-100 for **ACD** Basic Features
- 553-2671-101 for **ACD** Advanced Features
- 553-2671-102 for **ACD** Management Reporting
- ┆ 553-2671-103 for **ACD** Load Management
- 553-2671-104 for **ACD** with an Auxiliary Data System (**ADS**).
- 553-2671-105 for **ACD** Management Reports

1.02 The Meridian SL-1 can be arranged to support customers with only the **ACD** feature, or arranged to support both **ACD** and Private Branch Exchange (**PBX**) features. When the Meridian SL-1 is to serve both **ACD** and **PBX** features, the **ACD** requirements should be calculated first. Any remaining system capacity is then used to serve the **PBX** customers.

1.03 This practice is used to determine how a customer to be equipped with the **ACD** feature can be added to an existing Meridian SL-1, or to provision an initial Meridian SL-1 dedicated totally to the **ACD** feature. This practice gives information on how to calculate and assign:

- ┆ the required number of agent terminals
- the required number of supervisor terminals
- the **ACD** trunk traffic
- the required number of recorded announcement (**RAN**) trunks
- the required number of music trunks
- the required number of call registers.

1.04 Because the information given here applies only to the **ACD** feature, 553-2001-151 or 553-2201-151 should be used in conjunction with this practice to determine the Meridian SL-1 requirements as a whole.

2. ENGINEERING

Traffic **Estimates** Required

2.01 Initial engineering of the AW feature requires the following information, which can be obtained either from an estimate provided by the customer, or estimated based on a comparison of historical records of other systems equipped with the AW feature and serving similar applications. If estimates are not available, use the suggested values shown.

- (a) The average number of AW calls offered (**ACO**) at Average Busy Season Busy Hour (**ABSBH**).
- (b) The agent average Direct Call-Processing (**DCP**) time (suggest 180 s per AW call for enquiry/reservation applications; 30 s per AW call for credit verification or directory assistance applications).
- (c) The agent average Post Call-Processing (**PCP**) time (suggest 0 s or 30 s per AW call).
- (d) The agent total DN (**TDN**) time (total time spent by all agents assigned to an AW-DN on calls other than AW calls) per manned hour (suggest 360 s).
- (e) The High-Day Busy-Hour (**HDBH**) to ABSBH ratio (suggest 1.3).
- (f) The agent occupancy (suggest 92 percent); e.g., if the traffic load requires 92 agents, 100 should be assumed (allows for pauses, rest periods, etc.).
- (g) The Average Speed of Answer (**ASA**) objective (suggest 10 s).
- (h) The queue time for first RAN, if provided (suggest 20 s).
- (i) The percentage of connections to second RAN as a function of first RAN connections (suggest 30 percent, when required).
- (j) The basis for provisioning supervisor positions (suggest one supervisor per 25 agents). The system maximum is 40 agents per supervisor.
- (k) The proportion of AW calls coming from trunks rather than internal calls (suggest 100 percent).

Meridian SL-1 Type

2.02 If an initial Meridian SL-1 is being provisioned and the total number of AW calls to be offered is known, then the type of Meridian SL-1 required can be determined by comparing the estimated total number of AW calls offered with the maximum values applicable for each type of Meridian SL-1 (**LE**, **MS**, **N**, **N(QCA96)**, **S**, **XN**, **XL**, **XN(QCA97)***). Consult the appropriate appendix to 553-2001-151 or 553-2201-151, to determine the most up to date maximum data.

* LE, MS, N, S, VLE, XL and XN are trademarks of Northern **Telecom** Limited.

2.03 For the purpose of showing the calculations required for determining the type of Meridian SL-1 required, this practice will compare a hypothetical ACD system with the data available at the time of printing for the following four Meridian SL-1 types:

- an LE can process **6000/3000** ACD calls per hour at **0.35/0.7** s per call, or terminate up to 360,000 call-seconds per hour.
- a VLE can process **16000/6400** ACD calls per hour at **0.13/0.33** s per call, or terminate up to **1,800,000** call-seconds per hour.
- an XL or XN can process **35000/14000** ACD calls per hour at **0.06/0.15** s per call, or terminate up to **1,800,000** call-seconds per hour.

Note: Values shown depend on queuing, recorded announcement, and answered/abandoned calls assumptions. Note that specific limits may apply to the Auxiliary Data Link/Auxiliary Processor configuration, where applicable (**553-2671-104**).

Calculating Meridian
SL-1 Type
Requirements

2.04 This part outlines the procedures required to calculate the type of Meridian SE-1 required for the ACD feature. To clarify the concepts involved, a hypothetical ACD system with the following requirements is shown in the procedures. Note that calculations which result in a decimal number are rounded up to the next higher whole number.

- (1) Two ACD-DN are required, with **ACD-DN1** handling 2000 calls per hour, and ACD-DN2 handling 1500 calls per hour.
- (2) For **ACD-DN1**:
 - agent average DCP time is 120 s
 - agent average PCP time is 30 s
 - agent total DN time is 360 s per hour
 - percentage of trunk calls is 100 percent.
- (3) For **ACD-DN2**:
 - agent average **DCP** time is 200 s
 - agent average PCP time is 0 s
 - agent total DN time is 1000 s per hour
 - percentage of trunk calls is 50 percent.
- (4) Agent occupancy is equal to or less than 92 percent at ABSBH.
- (5) The **ASA** objective is less than or equal to 10s. This comprises 4s average wait in queue plus 6 s to ring the agent set and answer the call (reduces to 1 s when the 'call forcing' option is used).
- (6) Ratio of HDBH to ABSBH traffic is 1.3.
- (7) ACD calls delayed 20 s or more receive a 12 s recorded announcement.

- (8) **Thirty percent** of callers who receive first BAN, also receive **second RAN**.
- (9) Each supervisor monitors 35 agents and both call queues, and each agent is associated with only one supervisor. Supervisor positions generate an estimated 10 CCS per position.
- (10) A Meridian **SL-1** type VLE is required.
- (11) An average of 0.22 s (real time) per ACD call is assumed.

Note: Actual real time per ACD call is dependent on the type of Meridian **SL-1**.

Calculating Agent
Traffic and Agents
Required

2.05 Use the following formula to determine the agent traffic for each ACD-DN:

$$AT = (PCO/100) \times (DCP + PCP) \times F$$

where:

AT = agent traffic in CCS

PCO = peak calls offered, i.e., **ACO** × HDBH to ABSBH ratio

DCP = direct call-processing time per call

PCP = post call-processing time per call

F = the agent DN time factor; i.e., 3600 divided by (3600 minus the total agent DN (**TDN**) time per hour).

2.06 Example.

$$AT1 = (2000 \times 1.3 / 100) \times (120 + 30) \times (3600 / (3600 - 360))$$

$$= 4290 \text{ CCS}$$

$$AT2 = (1500 \times 1.3 / 100) \times (200 + 0) \times (3600 / (3600 - 1000))$$

$$= 5382 \text{ CCS}$$

2.07 Total agent traffic is , therefore, $4290 + 5382 = 9672$ CCS.

2.08 Once the agent traffic has been calculated, refer to Table 2-A to ascertain the number of agents required. If the agent traffic is not an exact multiple of 600 CCS (**660** CCS for Enhanced Meridian SL-1 N or XN), then use the following formula to determine how many agents are needed:

$$NA = AT \times B/A$$

where:

NA = the number of ACD agents required

AT = the agent traffic

A = value of agent CCS from Table 2-A which is closest to AT

B = the number of agents required for the value A in the table.

2.09 Example.

$$NA1 = 4290 \times (127/4200)$$

$$= 130$$

$$NA2 = 5382 \times (164/5400)$$

$$= 163$$

2.10 Then, **the** total number of agents required is $130 + 163 = 293$, and the average traffic per agent is $9672/293 = 33$ CCS

Note: If automatic overflow is used between the two queues, then the total agent traffic can be used to determine the combined number of agents required; e.g.,

$$NA \text{ (with overflow)} = (AT1 + AT2) \times (B/A)$$

$$= 9672 \times (290/9600)$$

$$= 292$$

Although there is little difference in this example, a significant savings in total agents can result from the availability of a larger (combined) number of answering agents for the combined traffic in certain circumstances.

Table 2-A
NUMBER OF ACD AGENTS REWIRED

AGENT TRAFFIC (CCS)	BUSY TIME/CALL (seconds)							
	20	40	80	120	150	200	300	500
	NUMBER OF ACD AGENTS REQUIRED							
600	20	21	22	22	23	23	24	25
1200	37			40			42	
1800	55	38	39		41	41		43
2400	73	55 ⁷³	57 ⁷⁴	58 ⁷⁵	58 ⁷⁶	59 ⁷⁷	78	62 ⁷⁸
3000	91	91	91	92	93	94	95	97
3600	109	109	109	109	110	111	113	115
4200	127	127	127	127	127	128	130	132
4800	145	145	145	164	145	145	147	149
5400	164	164	164	182	164	164	164	167
6000	182	182	182	182	182	182	182	184
6600	200	200	200	200	200	200	200	201
7200	218	218	218	218	218	218	218	218
7800	236	236	236	236	236	236	236	236
8400	254	254	254	254	254	254	254	254
9000	272	272	272	272	272	272	272	272
9600	290	290	290	290	290	290	290	290
10200	308	308	308	308	308	308	308	308
10800	327	327	327	327	327	327	327	327

Calculating ACD Trunk Traffic

2.11 Use the following formula to determine trunk traffic:

$$TT = THT \times (B/A) \times (TF/100)$$

where:

TT = trunk traffic in **CCS**

THT = trunk holding traffic $(PCO \times [DCP + ASA]) / 100$

PCO = peak calls offered $(ACO \times HDBH \text{ to } ABSBH \text{ ratio})$

DCP = direct call-processing time per call

ASA = average speed of answer

ACO = average calls offered

A = agent **CCS** from Table 2-B closest to **THT**

B = trunk traffic for value **A** from Table 2-B

TF = percentage of ACD calls from trunks

2.12 Example.

$$TT1 = ([2000 \times 1.3 \times (120+10)] / 100) \times (3240/3000) \times (100/100)$$

$$= 3650 \text{ CCS}$$

$$TT2 = ([1500 \times 1.3 \times (200+10)] / 100) \times (3773/3600) \times (50 \times 100)$$

$$= 2150 \text{ CCS}$$

2.13 Then the total trunk traffic (**TT**) is $3650 + 2150 = 5800 \text{ CCS}$.

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**Table 2-B
ACD TRUNK TRAFFIC (CCS)**

AGENT TRAFFIC (CCS)	TRUNK HOLDING TIME/CALL (seconds)							
	20	40	60	120	150	200	300	500
TOTAL TRUNK TRAFFIC (TT)								
600	841	722	662	647	633	627	617	610
1200	1702	1466	1344	1294	1276	1258	1238	1224
1800	2579	2209	1998	1934	1916	1884	1857	1832
2400	3251	2890	2673	2587	2544	2507	2474	2447
3000	4022	3571	3346	3240	3187	3140	3098	3057
3600	4792	4252	3982	3892	3892	3773	3711	3666
4200	5564	4933	4618	4513	4471	4406	4333	4281
4800	6336	5615	5255	5135	5087	5039	4956	4880
5400	7091	6281	5876	5741	5687	5633	5579	5502
6000	7866	6966	6516	6366	6306	6246	6186	6116
6600	8641	7651	7156	6991	6925	6859	6793	6736
7200	9417	8337	7797	7617	7545	7473	7401	7344
7800	10193	9023	8438	8243	8165	8087	8009	7947
8400	10970	9710	9080	8870	8786	8702	8618	8551
9000	11746	10300	9721	9496	9406	9316	9226	9154
9600	12523	11083	10363	10123	10027	9931	9835	9758
10200	13300	11770	11005	10750	10648	10546	10444	10363
10800	14072	12452	11642	11372	11264	11156	11048	10962

Calculating RAN Traffic and RAN Trunks Required

2.14 The 'recorded announcement' (**RAN**) feature does not make use of conference cards. Each **RAN** trunk is connected to only one ACD call at a time (but only for the duration of the **RAN** message). The **RAN** machine is wired to one or more **RAN** trunks which are grouped in software into trunk routes. ACD calls are connected to these **RAN** trunks in a manner directed by the Meridian SL-1 software.

2.15 A continuous **RAN** source can serve more than one trunk route and more than one trunk in a route. However, different **RAN** sources require different **RAN** trunk routes. (Stop/start **RAN** machines can only be assigned to one customer and one **RAN** trunk route.) For example, if the first **RAN** is different from the second **RAN**, they would have to use different **RAN** trunk routes. If a **RAN** is used for night treatment, and is different from the two delay announcements, it requires its own **RAN** trunk route. If, on the other hand, it is decided to use the same **RAN** trunk message for second **RAN** as for first **RAN**, then the same **RAN** trunk route can be used.

2.46 Use Table 2-C and the trunk holding traffic (**THT**) **CCS** from ACD trunk traffic calculations to determine the number of first RAN trunks required. If the trunk holding traffic is not an exact multiple of 600 **CCS**, then use this formula to determine the number of first RAN trunks required:

$$\text{NRT} = \text{THT} \times (\text{B}/\text{A})$$

where:

NRT = the number of first RAN trunks required

THT = the trunk holding traffic **CCS** as determined in the ACD trunk traffic calculations

A = the value of agent **CCS** from the table which is closest to THT

B = the number of first RAN trunks required for value A in the table

2.17 Example.

$$\text{NRT1} = 3380 \times (12/3000)$$

$$= 14 \text{ first RAN trunks}$$

$$\text{NRT2} = 4095 \times (10/3600)$$

$$= 12 \text{ first RAN trunks.}$$

2.18 The number of trunks for the second RAN can be determined by multiplying the number of first RAN trunks by the specified (or assumed) factor (e.g., 0.3). This serves as a value to start with, and may have to be modified as a result of ACD Management Reports data when available.

2.19 Example.

$$\text{NRT1} \text{ requires } 14 \times 0.3 = 4 \text{ second RAN trunks}$$

$$\text{NRT2} \text{ requires } 12 \times 0.3 = 3 \text{ second RAN trunks}$$

2.20 Therefore, the total number of **RAN** trunks required is $14 + 4 + 12 + 3 = 33$.

2.21 Assume that the traffic on each RAN trunk is the same as the individual agent **CCS**; i.e., 33 **CCS**. Then, the total **RAN** trunk traffic (**RT**) is $33 \times 33 \text{ CCS} = 1089 \text{ CCS}$.

Table 2-C
NUMBER OF FIRST RAN TRUNKS REQUIRED

TRUNK TRAFFIC (CCS)	TRUNK HOLDING TIME (seconds)							
	20	40	80	120	550	200	300	500
RAN TRUNKS REQUIRED								
600	15	11	6	5	5	4	3	2
1200	27	16	10	8	7	6	5	3
1800	39	22	14	10	9	7	6	4
2400		28	17	12	11	9		5
3000			19	14	12	10	4	6
3600			22	15	13	11	8	7
4200			24	15	13	11	9	7
4800			28	18	16	13	9	8
5400			31	21	18	15	11	8
6000				23	20	17	12	9
6600				26	19	18	13	10
7200				28	24	19	14	10
7800				30	25	19	15	10
8400				32	26	20	15	11
9000				33	27	21	16	12
9600				34	28	22	17	13
10200				35	29	23	18	13
10800				36	30	24	18	14

Calculating Network Loops Required For ACD Trunks

2.22 Each network loop can handle a maximum of 600 **CCS** (660 CCS for Enhanced Meridian SL-1 N or **XN**) of ACD trunk traf **fic**. Use the following formula to determine the number of network loops required:

$$L = TT/600 \text{ or } IT/660$$

where:

L = the number of network loops required

TT = the total ACD trunk traf **fic**.

2.23 Example.

$$L = 5800/600$$

= 10 network loops.

Supervisor Positions Required

2.24 In theory, one ACD supervisor can monitor 40 ACD agents. However, if the supervisor is required to handle some ACD **calls**, or perform normal administrative functions, then to assign that supervisor to 40 agent positions is not practical. Therefore, the number of supervisors to be provisioned is left to the discretion of the customer.

2.25 For our hypothetical **ACD** system, we have arbitrarily assigned 35 agents to each supervisor. Our calculations show that **ACD-DN1** requires 130 agents and **ACD-DN2** requires 163 agents. Based on 35 agents per supervisor, **ACD-DN1** would be assigned four supervisors and **ACD-DN2** would be assigned five supervisors.

Note: Agent-to-supervisor assignments need not be related to Agent-to-ACD-DN assignments. The customer can choose any combination or configuration to meet operational needs.

Calculating Network Loops Required for Agents, Supervisors and RAN Trunks

2.26 Use the following formula to determine the number of network loops required to serve the agents and RAN trunks:

$$L = (AT + RT)/750$$

where:

L = the number of network loops required

AT = the total agent traffic in CCS

RT = the total RAN trunk traffic in CCS (including second RAN).

Note 1: If (AT + RT) is less than 2000, divide by 600 (or 660 for Enhanced Meridian SL-1 N or XN) rather than 750.

Example:

$$L = (9672 + 1089)/750$$

$$= 15 \text{ network loops.}$$

Note 2: If agent positions require multiple appearance Directory Number (DN) keys, the 'same loop' restriction (see Assignment Guidelines) must be observed, which can result in an increase in the total number of agent loops required.

2.27 In the majority of cases, one network loop is more than sufficient to handle any traffic generated by the supervisor positions. Agents, ACD trunks, and RAN trunks should not be assigned to a loop which has supervisor positions assigned to it. Other PBX traffic can be used to utilize any remaining capacity of a loop which has supervisors assigned to it. Therefore, a total of 16 network loops are required to serve the agents, RAN trunks, and supervisors of our example.

Assignment Guidelines

2.26 Once the quantities of agents, RAN trunks, etc., have been determined, they can be assigned to the required network loops. Use the following guidelines to ensure an equal distribution of traffic:

- (a) If the Meridian SL-1 is to serve PBX lines and trunks as well as the ACD feature, assign the ACD feature requirements first. The remaining capacity is then assigned to serve PBX lines and trunks.
- (b) Agents and RAN trunks should be assigned together on loops separate from all other traffic. Distribute the agents and RAN trunks as evenly as possible over all such loops. Assign supervisor positions to a loop which is not associated with ACD agent traffic.
- (c) Assign ACD trunks to dedicated loops (600 CCS per loop maximum, or 660 for Enhanced Meridian SL-1 N or XN).
- (d) Intermix agent positions for the various ACD-DN on each loop, if applicable.
- (e) For multiple network group configurations, assign the ACD agent and trunk loops over as few network groups as possible to minimize the possibility of junctor blocking.

- (f) Music and RAN trunks must not be assigned to the same trunk circuit pack (QPC74).
- (g) When assigning music trunks, remember that these trunks cannot be accessed between network groups; i.e., a call in network group 1 cannot receive music from a music trunk in network group 3. A music trunk must be provided for each network group which requires music-on-hold or music-on-delay for callers from that group. These trunks can, however, be assigned to the same music route and/or music source if desired.

2.29 Table 2-D shows how the requirements calculated in our example would typically be assigned.

Table 2-D
 ASSIGNING EXAMPLE FOR AGENTS
 RAN TRUNKS AND SUPERVISORS

LOOP	NA1	NA2	NRT1	NRT2	SUPV1	SUPV2
1	9	10	3			
2	9	10	2	1		
3	9	11	1	1		
4	9	11	1	1		
5	9	11	1	1		
	9	11	1	1		
4	9	11	1	1		
8	9	11	1	1		
9	9	11	1	1		
10	9	11	1	1		
11	8	11	1	1		
12	8	11	1	1		
13	8	11	1	2		
14	8	11	1	2		
15	8	11	1	4		
16					4	5
TOTAL	130	163	18	19	4	5

Note: Loops 1-7 would be in one network group, and loops 8-15 would be in another network group in a multiple network group configuration. The loop for supervisors, ACD trunks, and music conference would also be split between the two network groups.

Calculating Call
Register Requirements

2.30 Use the following formula to determine the number of call registers required to serve the ACD feature:

$$\text{NCR} = ((\text{TT} \times \text{R}) + \text{Y}) / \text{Z}$$

where:

NCR = the number of call registers

TT = the total ACD trunk CCS

R = the ratio of HDBH traf **fic** to ABSBH traffic (1.3)

Y = 128 when ACD trunk **CCS** is 1400 or less

= 815 when ACD trunk **CCS** exceeds 1400

Z = 25.5 when ACD trunk **CCS** is 1400 or less

= 33.8 when ACD trunk **CCS** exceeds 1400.

2.31 Example.

$$\text{NCR} = ((5418 \times 1.3) + 815) / 33.8$$

$$= 233 \text{ call registers}$$

Note: Each call register occupies space in the unprotected data store portion of the Meridian SL-1 memory. Refer to 553-2001-151 or 553-2201-151 and the appropriate appendix to determine how much unprotected memory is required to serve the call registers, and determine additional call register needs for other PBX purposes, when applicable.

Conference Packs for
'Music on Delay'

2.32 Music (via conference) can be provided to calls waiting in an ACD queue after first and second RAN. The music comes from a music source via a **RAN** trunk dedicated to music, and is **connected** to the delayed calls via a conference circuit pack in a broadcast mode. The music trunk constitutes one of the conference parties, and the remaining parties are calls eligible to receive music-on-delay. The total conference traf **fic** for the Meridian **SL-1** depends on:

- | the number, size, and holding time of multi-party conferences
- the temporary use of the conference feature by other features: e.g., call transfer.

2.33 Because these factors can vary greatly, the total conference traffic is difficult to calculate exactly. In general, one conference loop is provided for each half-network group to handle maximum conference loads and to provide enhanced reliability (see 553-2001-151 or 553-2201-151).

2.34 For ACD, the total expected music traffic depends on:

- | the number of calls/hour arriving at the ACD
- | the percentage of these calls for which first RAN is anticipated
- | the average time that each ACD call connected to music has to wait for an agent (minus an allowance for repeat connections to second RAN, where applicable).

2.35 Unexpected increases in ACD traffic or answering delay, can cause great increases in the percentage of calls connected to RAN. This, in turn, can cause a sharp increase in the number of calls being conferenced into 'music on delay'. For this reason, the music feature is designed on a 'worst-case' basis. **One** additional conference loop (per network group) assigned to 'music on delay' is generally sufficient even for a large ACD system. (Note that this loop can still be used for non-music conferences, if ports are available at any time.) This conference loop, in turn, requires one music trunk. Since the music conference operates in a broadcast mode only, up to 29 calls can be conferenced with music simultaneously. Note that music cannot be accessed across network groups.

2.36 Thus, each network group equipped for music should be provided with at least two conference cards. Each network group containing trunks or stations liable to receive music must have a music conference loop with a music trunk assigned to it. The music trunks can share the same music source and be members of the same trunk route. Music trunks cannot be shared between customers or network groups. Each music source requires a separate trunk route. All RAN trunks on a RAN circuit card must terminate on the same RAN machine or music source; music trunks cannot be on the same card as RAN trunks.

2.37 The conference loop for 'music on delay' should be selected from the lower network loops of a network group. This is because the Meridian SL-1, when searching for regular conference ports, begins the search with the higher-numbered conference card, and will, therefore, minimize conflicts with the music conference(s).

Output Buffer and
Memory Requirements

2.38 Refer to 553-2001-151 or 553-2201-151 and the appropriate appendix to determine the impact of the ACD feature on Meridian SL-1 output buffers and memory.

INTEGRATED SERVICES NETWORK

MERIDIAN SL-1*

AUTOMATIC CALL **DISTRIBUTION** WITH
AN AUXILIARY DATA SYSTEM:
ENGINEERING, ORDERING AND
CONFIGURATION INFORMATION

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Reason for Reissue: To add Release 5.5 information. Release 5.5 introduces the **MicroPDP-11/73** processor in Part 2. Information applicable prior to RLS 5.5 is now contained in Part 3. Changes are marked by arrows in the margin. ↘

* Meridian and SL-1 are trademarks of Northern Telecom Limited

1. GENERAL

1.01 The Meridian SL-1 system is connected to an auxiliary processor (e.g., minicomputer) when the Automatic Call Distribution-Auxiliary Data System (ACD-ADS) feature is equipped. (The ACD-ADS feature is described in 553-2671-104.) The auxiliary processor used for ACD-ADS applications is the **PDP-11** type, manufactured, installed and maintained by Digital Equipment Corporation (**DEC**). Figure I-1 shows a typical ACD-ADS Configuration. The following PDP-11 processor types can be used with the ACD-ADS feature:

- **MicroPDP-11/73** ↗

- ┆ **PDP-11/23+** (see Part 3)

- ┆ **PDP-11/44** (see Part 3). ↘

1.02 The purpose of this appendix is to:

- provide ordering information for each processor
- provide ordering information for peripheral devices that connect to the processors
- provide ordering information for DEC support documentation
- detail the required modifications to DEC hardware that must be performed by DEC personnel either at the time of manufacture or installation of the processor
- provide the logical unit configuration and cabling configuration for each processor and the associated peripheral devices.

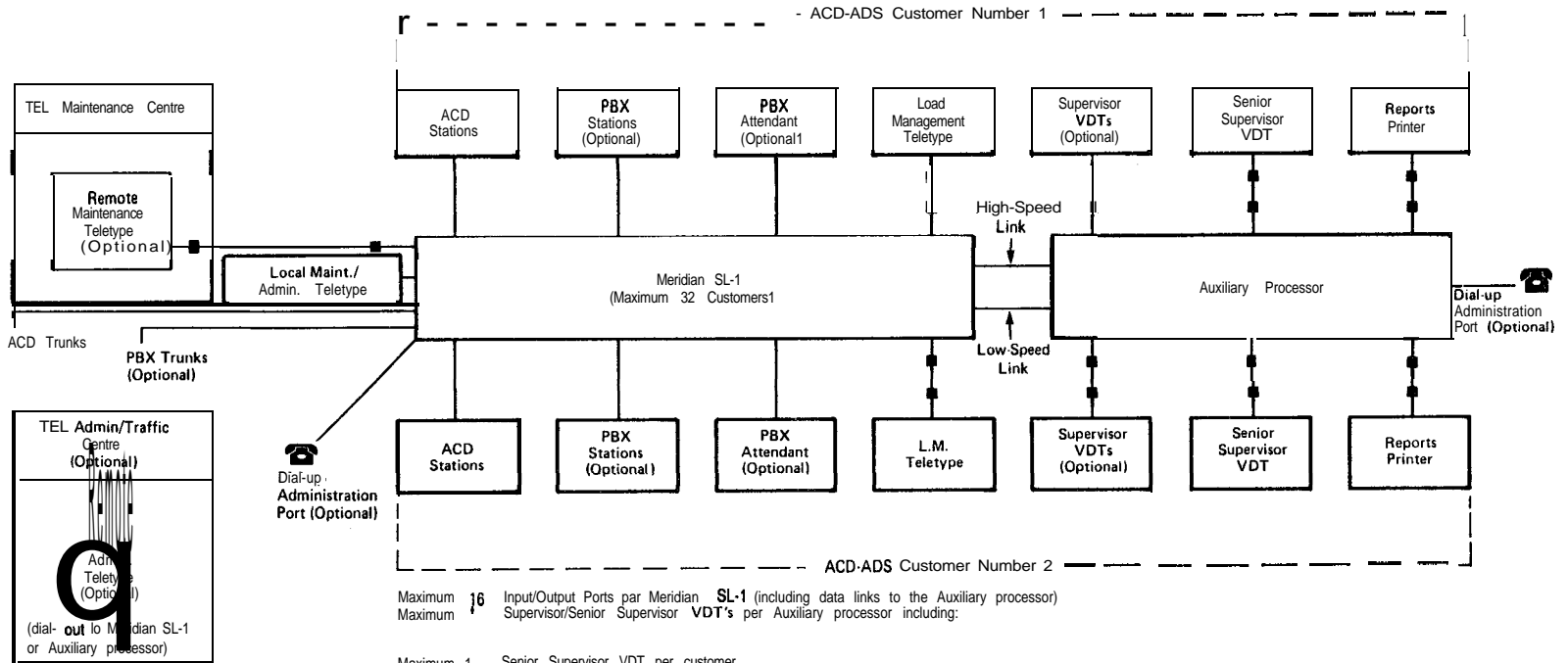
Note 1: The auxiliary processor, **VDT's** and printers are **ordered** ↗ from Northern Telecom in CANADA and in the USA from Northern Telecom by Direct Sales Offices, or from DEC by Northern Telecom Distributors.

Note 2: The customer may contact the local DEC office to request coordination of delivery, installation, environmental requirements, warranty and maintenance contract terms for a proposed installation as **applicable**. ↘

CAPACITY AND
ENGINEERING

1.03 Reference Appendix 2 to 553-2671-151 for capacity information and associated engineering requirements.

Fig. 1-1
A Typical ACD-ADS Configuration



Maximum 16 Input/Output Ports per Meridian SL-1 (including data links to the Auxiliary processor)
 Maximum 16 Supervisor/Senior Supervisor VDT's per Auxiliary processor including:

Maximum 1 Senior Supervisor VDT per customer.
 Maximum 1 Hard-copy Reports Printer per customer

16 for 11/23+
 16 for 11/44 and 11/73


Indicates modem or other data equipment which may be required if the distance exceeds 50 feet (15m).

(III. 0290)



2. ENGINEERING RLS 5.5 AND LATER

ORDERING
INFORMATION

2.01 Ordering information and associated documentation is detailed in  the following Tables and Charts:

ORDERING
MicroPDP-11/73

Table 2-A

Support documentation

Table 2-B

CONFIGURATION
Logical unit and cabling
for the **MicroPDP-11/73**Fig. 2-1 and
Fig. 2-2


2.02 Figures 2-1 and 2-2 should be referred to, when ordering a processor to determine the required number of interface cards, cables, **VDI's** and printers. 

Table 2-A
ORDERING INFORMATION FOR THE MICRODP-11/73

Northern Telecom **CPC** numbers (**A032nnnn** etc.) are listed directly below the DEC ordering code in the **CODE** column. These are for items that can be ordered separate from the Basic System (**A0326785**). NT Distributors must **use** the **A032nnnn** type number when ordering. Quantities marked with an asterisk are ordered as required.

QUANTITY	CODE	PROVIDES
1 (Note 1)	A0326785	2MB MicroDP-11/73 Basic System, which includes one each (except TK50 tape) of the following: <ul style="list-style-type: none"> • 11/73 processor (2MB memory) • Disk Controller • 31MB Winchester disk & drive • Tape Controller • 95MB cartridge tape drive • 8-Line Serial MUX card (5 VDT's max.) • TK50 cartridge tape (3 are supplied) • 24-Port distribution panel • General Operating System Software Licence (Note 2) • Integration, Testing, Documentation, Diagnostics (Note3) (see Table D)
* (Note 4)	DHV11-M A0326786	8-Line Serial MUX cards (see Fig. 2-1) for ordering requirements
1	LA210-AA A0326790	Reports Printer (r/o version)
*	TK50-K A0326789	TK50 cartridge tape
		VDT
*	VT220-A2 A0320897	- white screen
	or	
*	VT220-B2 A0320895	- green screen
	or	
*	VT220-C2 A0320896	- amber screen

Table 2-A Continued
 ORDERING INFORMATION FOR THE MICRO PDP-11/73



QUANTITY	CODE	PROVIDES
		Language Kit
*	VT22K-AA A0324082	- English (1 per VDT)
*	or	
	VT22K-AC A0326788	- French (1 per VDT)
		Cables
*	BC22D-25 A0326791	Printer/VDT cable 25 ft (7.62 m) see Fig. 2-2
*	BC22D-50 A0326792	Printer/VDT cable 50 ft (15.24 m) see Fig. 2-2
*	BC22D-A0 A0326793	Printer/VDT cable 100 ft (30.48 m) see Fig. 2-2
*	BC22E-25 A0326794	HSL/LSL cable 25 ft (7.62 m) see Fig. 2-2
*	BC22E-50 A0326795	HSL/LSL cable 50 ft (15.24 m) see Fig. 2-2
*	BC22E-A0 A0326796	HSL/LSL cable 100 ft (30.48 m) see Fig. 2-2

Note 1: This **MicroPDP-11/73** configuration has been carefully designed by both Northern Telecom and DEC. It is designed to serve all ACD-ADS (**ACD-D**) applications, regardless of size. The **MicroPDP-11/73** processor has all the capabilities of the **PDP-11/44**

Note 2: This is the Right-To-Use (**RTU**) fee paid to DEC for the use of **DEC's** proprietary operating system software (**RSX11M+**).

Note 3: This **MicroPDP-11/73** system is completely assembled, integrated and factory tested prior to shipment.

Documentation provided is listed in Table 2-B. The diagnostics provided include all the required programs to successfully install and troubleshoot the **MicroPDP-11/73** system to the board level. The **MicroPDP-11/73** is a board swap device; either the customer or the DEC Field Service can diagnose a faulty system and determine the failing part from the supplied diagnostics.



2.03 The customer will specify that DEC must configure the device and vector addresses on the **DHV11** cards. The addresses and the associated switch settings are as follows:

Note 1: All switch settings not shown as ON for each individual address are set to OFF.

Note 2: All vector address switch settings are located on the E43 switch.

Note 3: All device address switch settings are located on the **E58** switch, except the **TT11-20** setting of switch 1 which is located on the E43 switch.

DEVICE	PORT	VECTOR ADDRESS	VECTOR SWITCH SETTINGS ON	DEVICE ADDRESS	DEVICE SWITCH SETTINGS ON
DHV11-M	TT11-20	300	4 5	160440	5 8
DHV11-M	TT21-30	320	4 5 8	160500	5 8 1
RQDX3		Note 4	4 5 7	172150	Note 5
TQK50		Note 4		174500	Note 5

Note 4: The interrupt vector address for this card is set under factory control.

Note 5: The device addresses for the **RQDX3** (tape control) and the **TQK50** (disk control) cards are set at the factory by the use of jumpers installed on locations A2 through A12 on both cards. These device address jumpers are as follows:

RQDX3 - jumpers are on **A12,A10,A6,A5,&A3**, all others are removed.

TQD50 - jumpers are on **A12,A11,A8,&A6**, all others are removed.



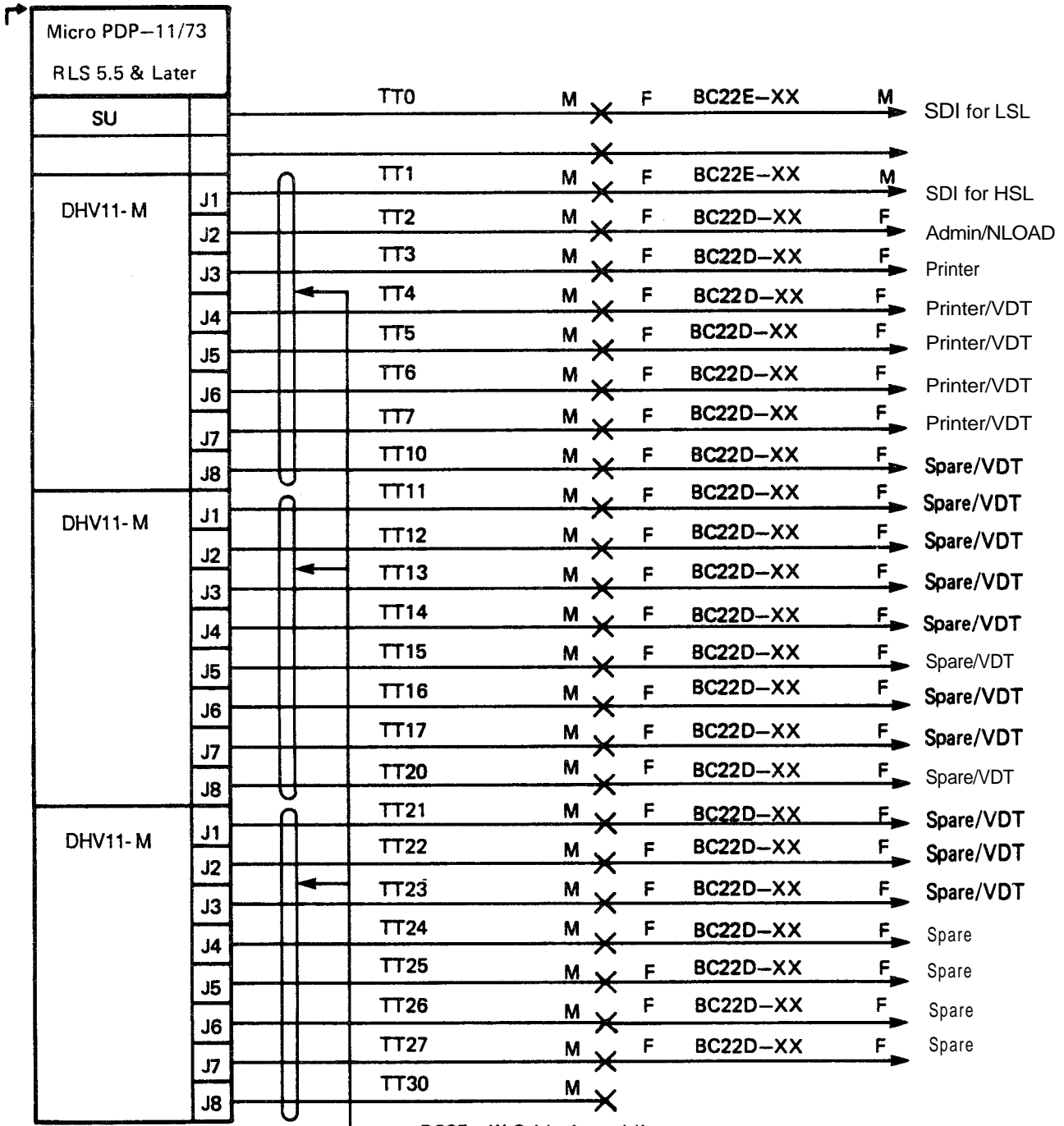
Micro PDP-11/73 RLS 5.5 and later				
MUX Card	Jack	Logical Unit Designation	Speed (Baud)	Function
SU	Note 1	TT0	300	Low-Speed Link (LSL)
DHV11-M	J1	TT1	9600	High-Speed Link (HSL)
	J2	TT2	300	Admin Port/N LOAD
	J3	TT3	1200	Printer # 1
	J4	TT4	1200/4800	Printer # 2/VDT
	J5	TT5	1200/4800	Printer # 3/VDT
	J6	TT6	1200/4800	Printer # 4/VDT
	J7	TT7	1200/4800	Printer # 5/VDT
	J8	TT10	4800	Spare/VDT
DHV11-M	J1	TT11	4800	Spare/VDT
	J2	TT12	4800	Spare/VDT
	J3	TT13	4800	Spare/VDT
	J4	TT14	4800	Spare/VDT
	J5	TT15	4800	Spare/VDT
	J6	TT16	4800	Spare/VDT
	J7	TT17	4600	Spare/VDT
	J6	l-r20	4800	Spare/VDT
DHV11-M	J1	TT21	4800	Spare/VDT
	J2	l-r22	4800	Spare/VDT
	J3	TT23	4800	Spare/VDT
	J4	TT24	4800	Spare
	J5	TT25	4800	Spare
	J6	TT26	4800	Spere
	J7	TT27	4800	Spare
	J8	TT30		Spare

Notes:

1. This jack is located on the System Unit (SU).
2. Printers are assigned on a one per customer basis starting at TT3.
VDTs are assigned starting at the first free port after the last assigned printer.
3. With more than one printer defined in a multi-customer environment, all the **VDTs** will have to move down the appropriate amount.
 e.g. With 5 printers, **TT10 = VDT #1, TT27 = VDT #16.**

(III. 553-1007)

Fig. 2-1
MicroPDP-11/73 Logical Unit Configuration RLS 5.5 and later



BC05-W Cable Assemblies

Notes: 1. Replace BC22D-X null-modem cables with BC22E-XX cables when modems are required.

- NE-A25MQ-type standard EIA-RS232C interface cables may be used in place of BC22E-XX cables. These cables may be ordered from Northern Telecom in lengths as required.
- With more than one printer defined in a multi-customer environment, all the VDTs will have to move down the appropriate amount. e.g. With 5 printers, TT10=VDT #1, TT27=VDT # 16.
- M = male connector; F = female connector.

(III. 553-1008)

Fig. 2-2
MicroPDP-11/73 Cabling Configuration RLS 5.5 and later

Table 2-B
PDP-11 SUPPORT DOCUMENTATION SUMMARY



TITLE	ORDERING NUMBER
MicroPDP-11/73 System Manuals	Note 1
-Owners Manual	*
-Users Manual	*
-Technical Manual	*
-Diagnostic (TK-50)	*
LA-210 User Guide	Note 2
VT-220 User Guide	Note 2

Note: 1 One each of the Manuals marked with an asterisk are supplied as a package with each MicroPDP-11/73 system. Additional 3 item **MicroPDP-11/73 Manual packages** can be ordered from DEC using Part number **ZYAAA-P5**. The package manuals are of a generic flavour and include information on the supported options of the **MicroPDP-11/73** i.e., disks, tapes, common devices, etc. and also the required diagnostics to successfully install and troubleshoot the **MicroPDP-11/73** to the board level.

Note 2: This documentation is supplied with the associated equipment.



PERIPHERAL
EQUIPMENT

2.04 The following peripheral equipment devices can be ordered for use with the ACD-ADS feature:

- | ACD supervisor Video Display (VDT) terminals (connect to the PDP-11 processor)
- | ACD Management Reports printers (connect to the PDP-11 processor)
- | ACD Load Management terminals (connect to the Meridian **SL-1**)
- | modems or other data equipment
- | connecting cables.

2.05 Supervisor VDT. ACD-ADS can support a maximum of:

- | 16 Video Display terminals. The terminal recommended for this application is the VT-220 (**see** Table 2-C). The terminal must be configured to operate at 4800 baud in full-duplex mode.



Table 2-C
VT220 TYPE VIDEO DISPLAY TERMINAL

Characteristics:

- | | |
|--|--|
| <ul style="list-style-type: none"> Baud rate: 50 to 19,200 b/s Format: 24 lines x 80 characters Characters: 7 x 9 dot matrix with descenders ● Character set: 94 displayable-character ASCII set and 32-character special line-drawing graphics set Double width/double height characters Standard numeric and function keypads Bidirectional vertical scrolling Selectable smooth or jump scrolling Split-screen capability Format: 14 lines x 132 characters, selectable | <ul style="list-style-type: none"> ● Normal or reverse screen image Adjustable tabs ● Full-duplex operation Keyboard selectable features Non-volatile set-up memory Cursor control keys ANSI/VT52 command modes 20 character answerback message Selectable XON/XOFF buffer control Self-test diagnostics |
|--|--|

Special features

- | | |
|--|--|
| <ul style="list-style-type: none"> Monitor tilt button 12 in (305 mm) non-glare screen | <ul style="list-style-type: none"> plain language set-up menu ● low profile detachable keyboard. |
|--|--|

Ordering Information:

Video Terminals:

- VT220-A White phosphor non-glare screen
- VT220-B Green phosphor non-glare screen
- VT220-C Amber phosphor non-glare screen

Country Kits:

- VT22K-AA USA and English-speaking
Canada
- VT22K-AC French-speaking Canada

Note: Cables are not provided and must be ordered separately. To utilize the printer port the **BCC05** cable is recommended. The recommended cables for terminal to host connections are as follows: BCC04 EIA cable for connection to a modem, and **BC22D** null modem cable for connection to a line unit.

→ 2.06 Reports Printer. ACD-ADS supports a maximum of 1 hard-copy printer per ACD-ADS defined customer (see Table 2-E). The LA210 printer is not included in the system package and must be ordered separately.

↗ **Table 2-E**
REPORTS PRINTER

MODEL	KEYBOARD & QUALITY	PRINT SPEED	PAPER FEED	SPECIAL FEATURES
LA210-	(AC)French Canada AQ English Canada AA USA only	240 cps Draft 40 cps letter quality	Friction	Plug-in fonts PC-compatible

→ 2.07 Load Management Terminal. Each ACD-ADS customer can be equipped with one Load Management terminal (connected to the Meridian SL-1). The LA210 is preferred but, either the LA100 or LA120 can be used (see Part 3 Table 3-E). Alternatively, any 20 mA, RS232-D and ASCII-compatible device with keyboard may be used.

→ 2.08 The Load Management function can, if desired, be performed from a VDT that is equipped as a senior supervisor position. This requires that a suitable customer-provided switching device be equipped to enable the senior supervisor to select between the ACD status display function of the auxiliary processor and the Load Management function of the Meridian SL-1. (Alternatively, the Meridian SL-1 Lanstar Data Service feature can be used to provide this switching function.) A cabling arrangement to support this option is shown in Fig 2-3. Use of this option means that a hard-copy record of Load Management transactions is not maintained, and ACD status displays are not available while in the Load Management mode.

→ 2.09 Similarly, the Management Reports printer can be used to perform a Load Management function as well as being used to print reports. This option also requires that a suitable customer-provided switching device be equipped to alternate the functions. Use of this option means that some management reports may be lost while the printer is switched to Load Management mode. The lost reports would have to be requested a second time when the printer is restored to the Management Reports mode.

→ 2.10 If on-site cabling between VDTs and the PDP-11 processor exceeds 50 ft (15 m), the use of Limited Distant Data Sets (LDDS) is recommended. If the VDTs are located in separate buildings from the PDP-11, an appropriate asynchronous modem and telephone facilities must be used. Modems are also required if the auxiliary processor's maintenance and administration port is to be accessed from a remote location.

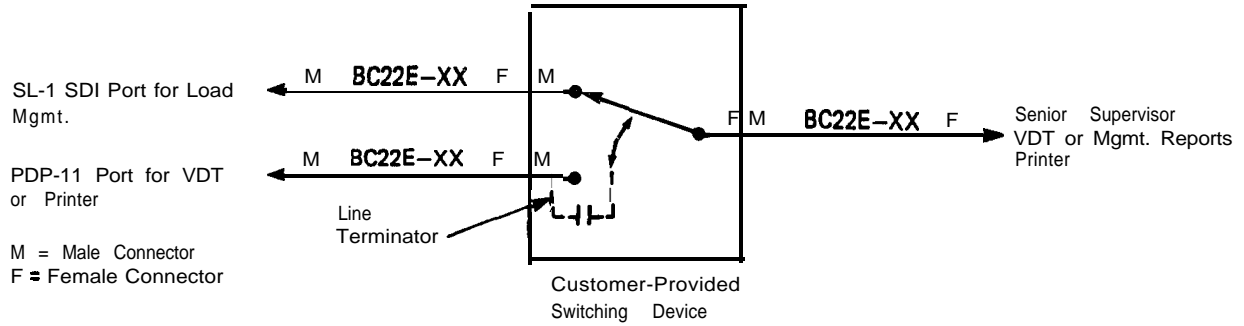


Fig. 2-3
Arrangement to Share Load Management with **VDT/Printer** Functions

DEC ACCESS TO
PROCESSOR

2.11 Digital Equipment Corporation personnel periodically require **TTY** access to the auxiliary processor for preventive maintenance purposes. The Meridian SL-1 maintenance TTY may be used (if a customer-provided switching device) configured and cabled as shown in Fig 2-4, is installed. ←

CABLING
CONFIGURATION

2.12 Fig. 2-2, shows a typical cabling configuration for the auxiliary+ processor and related peripheral equipment. Follow DEC recommendations for cable lengths, modem requirements, etc. Ensure that at the time of processor installation, DEC personnel clearly designate the processor logical unit number (**TT0-TT27**) on all cables. Ensure that DEC personnel clearly identify logical unit numbers on the distribution panel of the **DHV11-** cards for the **MicroPDP-11/73** ←

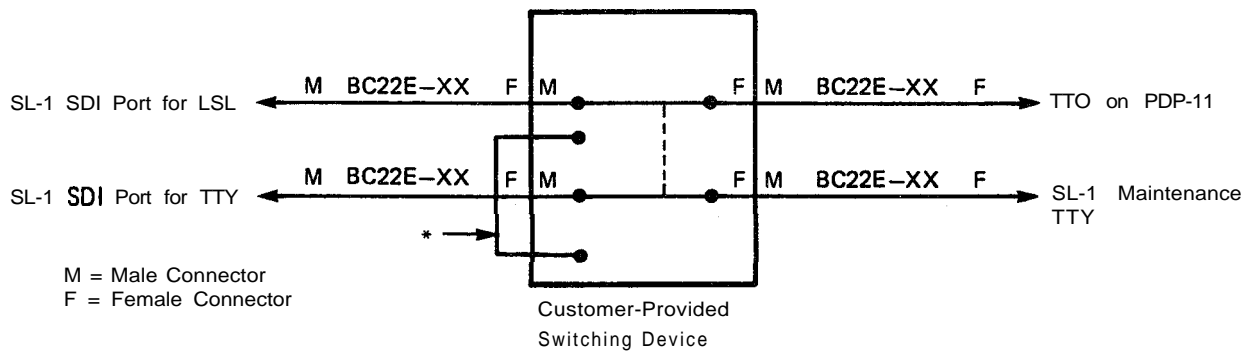


Fig. 2-4
Arrangement for DEC to Use the Meridian SL-1 TTY for PDP-11 Maintenance
(bypassing the **LSL**)

3. ENGINEERING RLS 5.0 AND EARLIER

CAPACITY AND
ENGINEERING

3.01 Reference Appendix 2 to 553-2671-151 for capacity information and associated engineering **requirements**.

ORDERING
INFORMATION

3.02 Ordering information and associated documentation is detailed in the following Tables and Charts:

ORDERING		↖
PDP-11/23+	Table 3-A	
PDP-11/44	Table 3-B	
Support documentation	Table 3-C	

CONFIGURATION		
Logical unit and cabling for the PDP-11/23+	Fig. 3-1 through Fig. 3-4	
Logical unit and cabling for the PDP-11/44 .	Fig. 3-5 through Fig. 3-8	

3.03 Figures 3-1 through 3-8 should be referred to, when ordering a processor to determine the required number of interface cards (e.g., **DZV11-DP, DZ11-DP**), cables, VDT and printers. ↙

Table 3-A
ORDERING INFORMATION FOR THE **PDP-11/23+**

QUANTITY	CODE	PROVIDES
1	SX-RXMMA-EK*	PDP-11/23-1- RL-02 based system which includes: <ul style="list-style-type: none"> • 11/23+ CPU and power supply • 512 kb PARITY MOS memory (MSV11-PL). • Bootstrap module with diagnostics. See Note 2. ┆ Two single-line asynchronous interfaces: one for the LA120 console terminal (TT0) and one for LA100 (TT1). • Two RL-02 10.4 Mb removable cartridge disk drives and controller. See Note 1. ┆ CPU cabinet (H9642) and power controller. ┆ BC22D-25 cable for console terminal. ┆ Input/output connection panel (H349). ┆ General Operating System Software Licence.
1	MSV11-PL*	512 kb parity MOS memory.
As req'd →	DZV11-DP	4-line asynchronous MUX (with modem control) for logical units. one card each - See Fig. 3-1 (Fig. 3-3 for RLS 4.5 and later):
6	RL02K-DC*	10.4 Mb cartridge disks (for ACD-ADS applications).
2 →	BC22E-XX*	Cable for connection to the SDI port for the HSL and LSL. where XX is the cable length. Specify length as required - see Fig. 3-2 (Fig. 3-4 for RLS 4.5 and later).
As req'd →	VT100-AA	Video display terminal (VDT) for supervisor positions and senior supervisor position (Table 3-D).
As req'd →	VT1XX-AB	Advanced video option for each VT 100 (provides highlighting and reverse video capabilities, Table 3-D).
As req'd	VT1XX-AE	Formed contrast filter for each VT 100.
	and/or	
As req'd →	VT220-A/B/C	VDT screen for supervisor positions and senior supervisor position (Table 2-C).
As req'd →	VT22K-AA	North American keyboard and documentation for each VT220 (Table 2-C).
As req'd	LA100-CA	LETTERWRITER 100 Hard-copy desk top printer for Load Management or Maintenance Terminal.

Table 3-A Continued
 ORDERING INFORMATION FOR THE **PDP-11/23+**

QUANTITY	CODE	PROVIDES
As req'd	LA100-ZA	LETTERPRINTER 100 - hard copy management reports printer (receive only - no keyboard).
As req'd	LA10X-SL	LA100 terminal stand (optional).
As req'd	BC22D-XX	Null-modem cable for connection to VDT and printers. Specify lengths as required.

* denotes items that are always required.

Note 1: Specify that DEC must configure the **PDP-11/23+** with DLO: as the top drive and **DL1:** as the bottom drive.

Note 2: Specify that DEC must set the Bootstrap Card to 'auto-boot' on power-up.

Note 3: Specify that DEC must configure the device and vector addresses on the **DZV11** cards as follows:

DEVICE	PORT	VECTOR ADDRESS	DEVICE ADDRESS
s u	TT1	60	177560
s u	TT2-5	300	176500
DZV11-DP		310	160100
DZV11-DP	TT6-11	320	160110
DZV11-DP	TT12-15	330	160120

Note 4: If only the vector addresses are set incorrectly the system will attempt to startup, but will stop generating LL:NNNNNN* followed by the Executive Debugging Tool prompt XDT . IF the response to XDT is X the system will generate a register dump. XDT will not accept a wide variety of inputs - see DEC reference manual. The total failure of one of the **DZ** cards would probably cause this situation to occur on reboot.

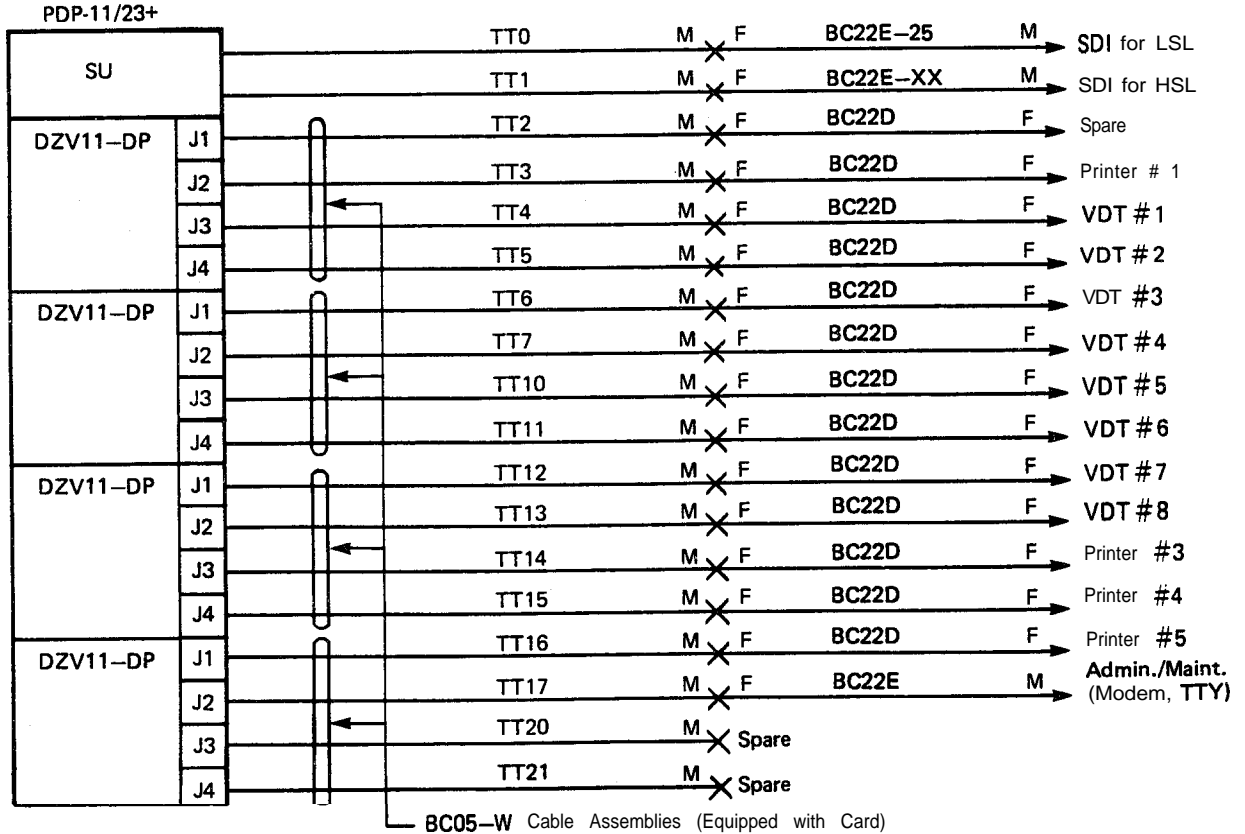
* L = letter N = number

PDP-11/23+				
MUX Card	Jack	Logical Unit Designation	Speed (Baud)	Function
SU	Note 1	TT0	300	Low-Speed Link (LSL)
		TT1	1200	High-Speed Link (HSL)
DZV11-DP	J1	TT2	1200	Spare
	J2	TT3	1200	Printer # 1
	J3	TT4	4800	VDT # 1
	J4	TT5	4800	VDT # 2
DZV11 -DP	J1	TT6	4800	VDT # 3
	J2	TT7	4800	VDT # 4
	J3	TT10	4800	VDT # 5
	J4	TT11	4800	VDT # 6
DZV11-DP	J1	TT12	4800	VDT # 7
	J2	TT13	4800	VDT # 8
	J3	TT14	1200	Printer # 3
	J4	TT15	1200	Printer # 4
DZV11 -DP	J1	TT16	1200	Printer # 5
	J2	TT17	300	Admin Port (Modem, TTY)
	J3	TT20	9600	Spare
	J4	I-I-21	9600	Spare

Notes:

1. These two jacks are located on the System Unit (SU).

Fig. 3-1
PDP-11123-t Logical Unit Configuration RLS 4.0 and earlier



Notes:

1. Replace BC22D-XX null-modem cables with BC22E-XX cables when modems are required. BC22E-XX and BC22D-XX cables are ordered from DEC in lengths as required.
2. NE-A25MQ-type standard EIA-RS232C interface cables may be used in place of BC22E-XX cables. These cables may be ordered from Northern Telecom in lengths as required.
3. M = male connector; F = female connector

(III. 01140)

Fig. 3-2
PDP-11/23+ Cabling Configuration RLS 4.0 and earlier

PDP-11/23+		RLS 4.5 & Later		
MUX Card	Jack	Logical Unit Designation	Speed (Baud)	Function
SU	Note 1	TT0	300	Low-Speed Link (LSL)
		TT1	1200	High-Speed Link (HSL)
DZV11 -DP	J1	TT2	300	Admin Port/NLOAD
	J2	TT3	1200	Printer
	J3	TT4	4800	VDT #1
	J4	TT5	4800	VDT # 2
DZV11 -DP	J1	TT6	4800	VDT # 3
	J2	m	4800	VDT # 4
	J3	TT10	4800	VDT # 5
	J4	TT11	4800	VDT # 6
DZV11 -DP	J1	TT12	4800	VDT # 7
	J2	TT13	4800	VDT # 8
	J3	TT14	4800	VDT # 9
	J4	TT15	4800	VDT # 10
DZV1 I-DP	J1	TT16	4800	VDT # 11
	J2	TT17	4800	VDT # 12

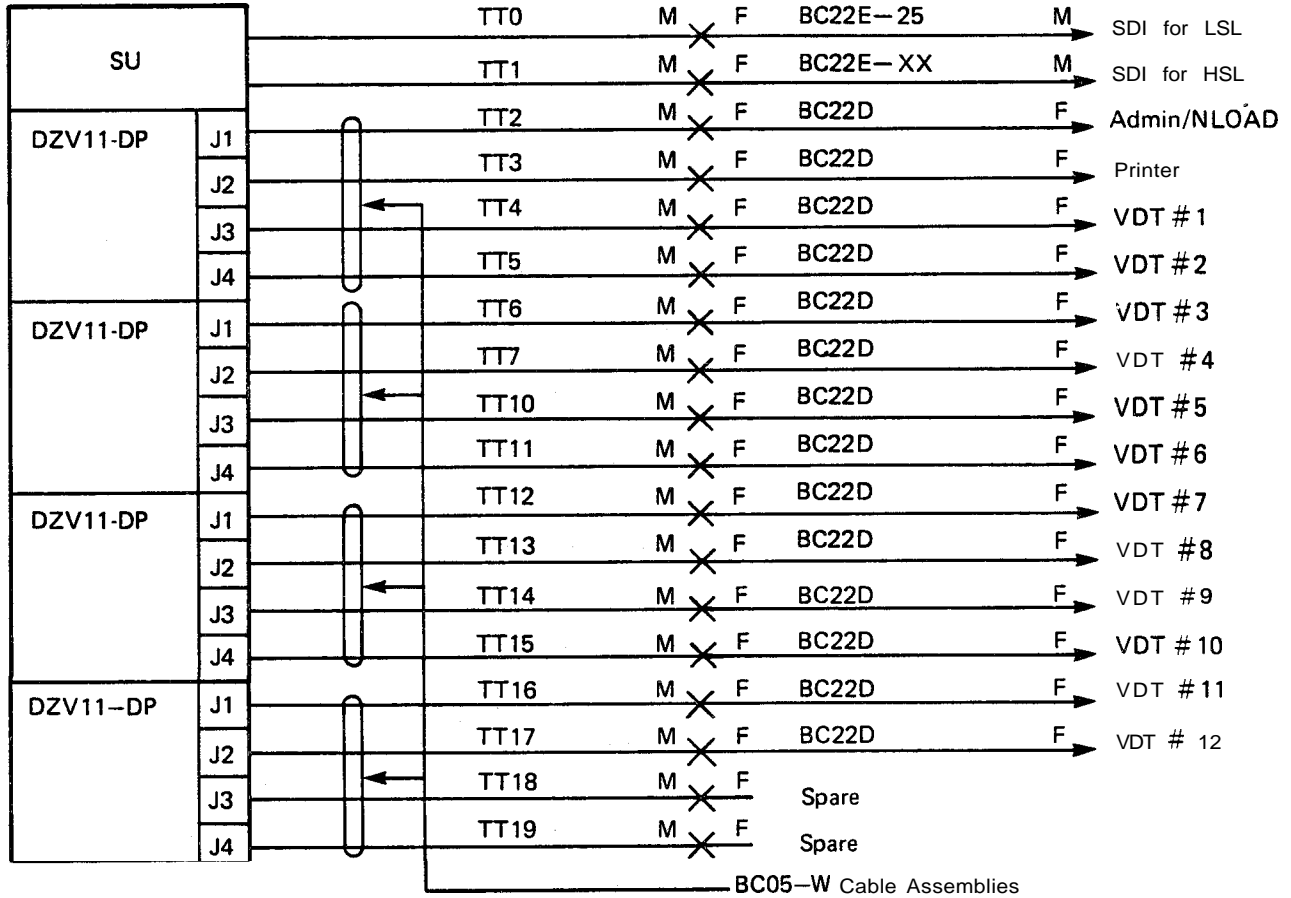
Note :

1. These two jacks are located on the System Unit (**SU**).

(III. 6375)

Fig. 3-3
PDP-11/23+ Logical Unit Configuration RLS 4.5 and later

PDP-11/23+
 RLS 4.5 & Later



Notes:

1. Replace **BC22D-XX** null-modem cables with **BC22E-XX** cables when modems are required. **BC22E-XX** and **BC22D-XX** cables are ordered in lengths as required.
2. **NE-A25MQ-type** standard **EIA-RS232C** interface cables may be used in place of **BC22E-XX** cables. These cables may be ordered from Northern Telecom in lengths as required.
3. M = male connector; F = female connector.

(III. 06376)

Fig. 3-4
PDP-11/23+ Cabling Configuration RLS 4.5 and later

Table 3-B
ORDERING INFORMATION FOR THE PDP -11/44

QUANTITY	CODE	PROVIDES
1	SX-40MMB-EK*	<p>PDP-11/44 RL-02 based system which includes:</p> <ul style="list-style-type: none"> 11/44 CPU and power supply. • Bootstrap module with diagnostics. See Note 3. MS11-PB 1.0 MB ECC/MOS Memory. Two single-line asynchronous interfaces: one for the LA120/LA100 console terminal (TT0); one for expansion. • One BC22D-25 E/A cable for console terminal. I/O connection panel. • Two RL-02 10.4 Mb removable cartridge disk drives and controller. See Note 2. Wide CPU cabinet (H9645-EA) with power controller. • General Operating System Software Licence.
1	MS11-PB*	1.0 MB ECC/MOS Memory.
As req'd	DZ11-DP	4-line asynchronous MUX (with modem control) for logical units, one card each - See Fig. 3-5 and Note 1 (see Fig: 3-7 for RLS 4.5 and later).
→		
6	RL02K-DC*	10.4 Mb cartridge disks (for ACD-ADS applications).
1	BC05C-XX*	Cable for connection to the SDI port for the HSL. Specify length as required - Fig. 3-6 (see Fig. 3-8 for RLS 4.5 and later). If distance is less than 25 ft (7.5 m), the cable provided with the DL11-AP card may be used.
→		
As req'd	VT100-AA	Video display terminal (VDT) for supervisor positions and senior supervisor position (Table 3-D).
As req'd	VT1XX-AB	Advanced video option for each VT-100. (req'd for Bold, Link etc.
As req'd	VT1XX-AE	Formed contrast filter for each VT-100.
	and/or	
As req'd	VT220-A/B/C	VDT screen for supervisor positions and senior supervisor position (Table 2-C).
→		
As req'd	VT22K-AA	North American keyboard and documentation for each VT220 (Table 2-C).
→		
As req'd	LA100-CA	LETTERWRITER 100 hardcopy desk top printer for load management or maintenance terminal.

Table 3-B Continued
ORDERING INFORMATION FOR THE PDP-11144

QUANTITY	CODE	PROVIDES
As req'd	LA100-ZA	LETTERPRINTER 100 - hard copy management reports printer (receive only - no keyboard).
As req'd	LA10X-SL	LA100 terminal stand (optional).
As req'd	BC22E-XX	Cable for connection to the SDI port for the LSL, or to a modem. Specify lengths as required.
As req'd	BC22D-XX	Null-modem cable for connection to VDT and printers. Specify lengths as required.

* denotes items that are always required.

Note 1: Specify that DEC must configure **DZ11** card device and vector address switches as follows:

DZ11-DP

- | Device Address: (160020) switch 2 ON; all the rest OFF
- | Vector Address: (310) switches 1, 3, 4, 7, 8 ON; all the rest OFF

DZ11-DP

- | Device Address: (160030) switches 1, 2 ON, all the rest OFF
- | Vector Address: (320) switches 1, 2, 4, 7, 8 ON; all the rest OFF.

Note 2: Specify that DEC must configure the **PDP-11/44** with **DL0:** as the top drive and **DL1:** as the bottom drive.

Note 3: Specify that DEC must set the Bootstrap Card to 'auto-boot' on power-up.

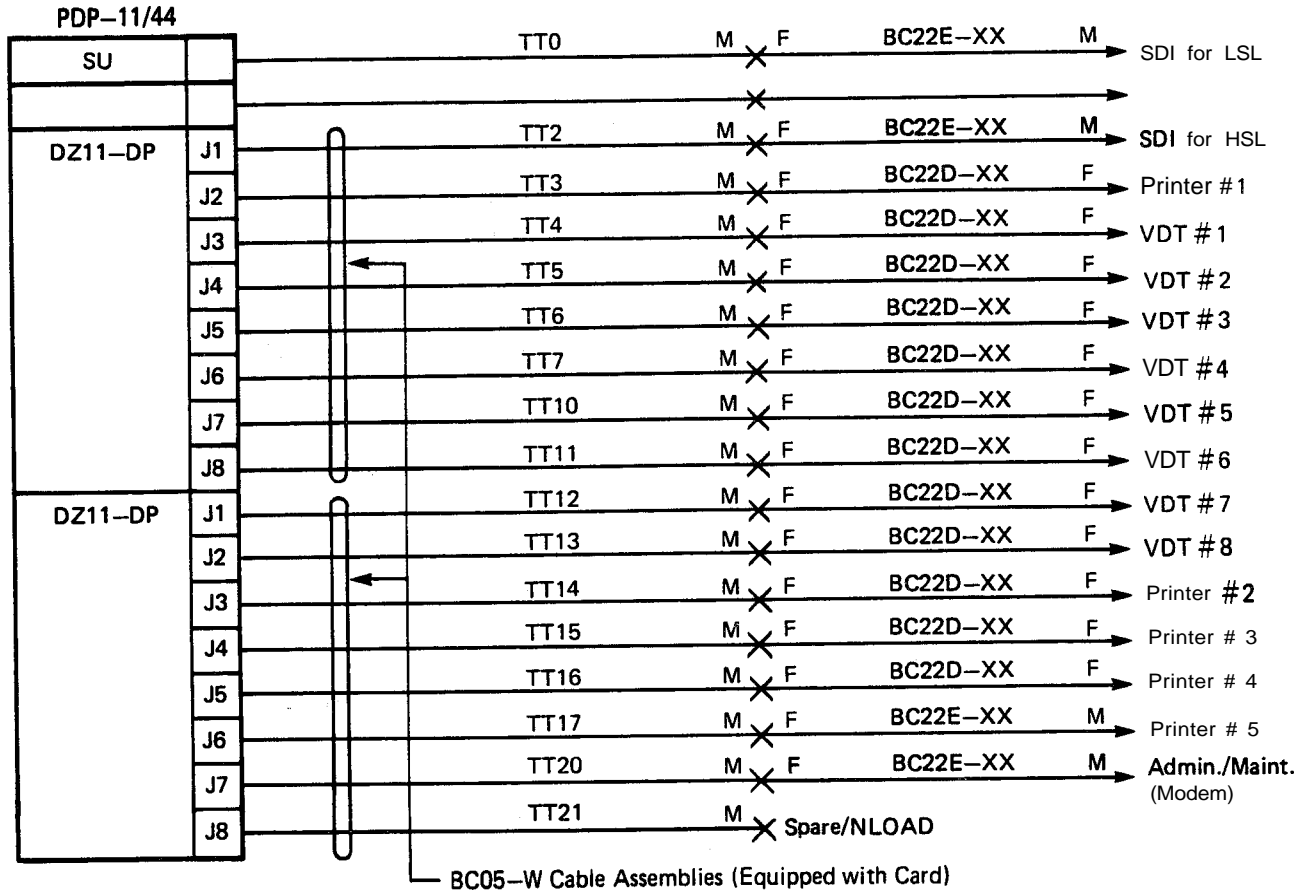
POP-1 1/44				
MUX Card	Jack	Logical Unit Designation	Speed (Baud)	Function
SU	Note 1	TTO	300	Low-Speed Link (LSL)
DL11-AP	Note 2	TT1	9600	HSL see Note 3
DZ11-DP	J1	TT2	9600	HSL
	J2	TT3	1200	Printer # 1
	J3	TT4	4800	VDT # 1
	J4	TT5	4800	VDT # 2
	J5	TT6	4800	VDT # 3
	J6	TT7	4800	VDT # 4
	J7	TT10	4800	VDT # 5
	J8	TT11	4800	VDT # 6
DZ11-DP	J1	TT12	4800	VDT # 7
	J2	TT13	4800	VDT # 8
	J3	n-14	1200	Printer # 2
	J4	TT15	1200	Printer # 3
	J5	TT16	1200	Printer # 4
	J6	TT17	1200	Printer # 5
	J7	TT20	300	Admin Port (Modem)
	J8	TT21	9600	Spare/N LOAD

Notes:

1. This jack is located on the System Unit (SU).
2. DL11-AP previously designated DL11 -E.
3. Not used for RLS 3.7X and later.

(III. 01141)

Fig. 3-5
PDP-11/44 Logical Unit Configuration RLS 4.0 and earlier



Notes:

1. Replace **BC22D-X** null-modem cables with **BC22E-XX** cables when modems are required. **BC22D-XX** and **BC22E-XX** cables are ordered from DEC in lengths as required.
2. **NE-A25MQ-type** standard EIA-RS232C interface cables may be used in place of **BC22E-XX** cables. These cables may be ordered from Northern Telecom in lengths as required.
3. M = male connector; F = female connector.

Fig. 3-6
PDP-11/44 Cabling Configuration RLS 4.0 and earlier

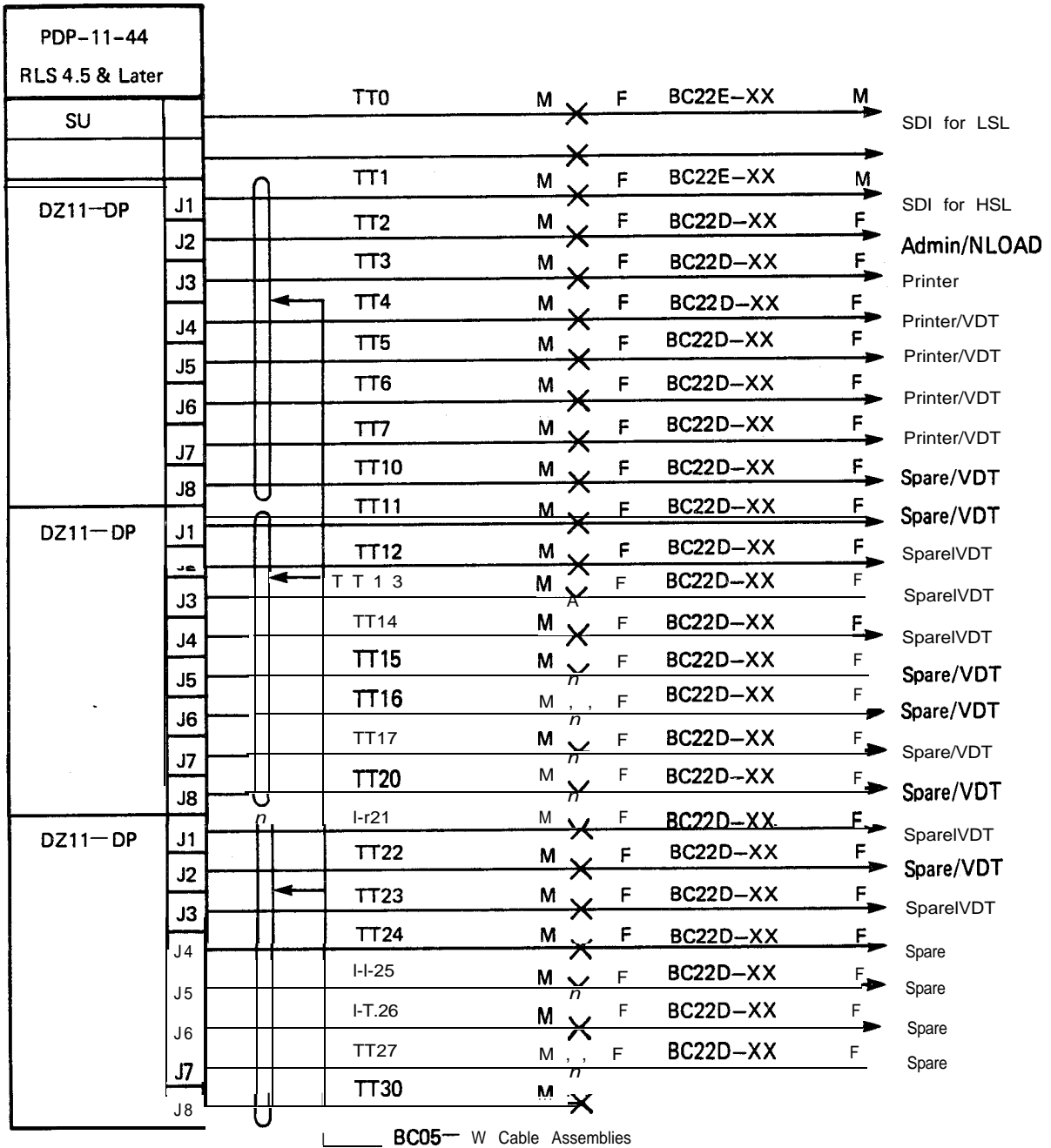
PDP-11/44 RLS 4.5 & Later				
MUX Card	Jack	Logical Unit Designation	Speed (Baud)	Function
SU	Note 1I	TT0	300	Low-Speed Link (LSL)
DZ11-DP	J1	TT1	9600	High-Speed Link (HSL)
	J2	TT2	300	Admin Port/NLOAD
	J3	TT3	1200	Printer # 1
	J4	TT4	1 20014800	Printer # 2/VDT
	J5	TT5	1200/4800	Printer # 3/VDT
	J6	TT6	120014800	Printer # 4/VDT
	J7	TT7	120014800	Printer # 5/VDT
	J8	TT10	4800	Spare/VDT
DZ11-DP	J1	TT11	4800	Spare/VDT
	J2	TT12	4800	Spare/VDT
	J3	TT13	4800	Spare/VDT
	J4	TT14	4800	Spare/VDT
	J5	TT15	4800	Spare/VDT
	J6	TT16	4800	Spare/VDT
	J7	TT17	4800	Spare/VDT
	J8	TT20	4800	Spare/VDT
DZ11-DP	J1	TT21	4800	Spare/VDT
	J2	TT22	4800	Spare/VDT
	J3	TT23	4800	Spare/VDT
	J4	TT24	4800	Spare
	J5	TT25	4800	Spare
	J6	TT26	4800	Spare
	J7	TT27	4800	Spare
	J8	TT30		Spare

Notes:

1. This jack is located on the System Unit (**SU**).
2. Printers are assigned on a one per customer basis starting at TT3.
VDTs are assigned starting at the first free port after the last assigned printer.
3. With more than one printer defined, the VDTs will have to move down the appropriate amount.
e.g. With 5 printers, TT10 = VDT #1, TT27 = VDT #16.

(III. 06377)

Fig. 3-7
PDP-11/44 Logical Unit Configuration RLS 4.5 and later



- Notes: 1. Replace BC22D-X null-modem cables with BC22-XX cables when modems are required. BC22D-XX and BC22E-XX cables are ordered from DEC in lengths as required.
2. NE-A25MQ-type standard EIA-RS232C interface cables may be used in place of BC22E-XX cables. These cables may be ordered from Northern Telecom in lengths as required.
3. With more than one printer defined, the VDTs will have to move down the appropriate amount. e.g. With 5 printers, TT10 = VDT #1, TT27 = VDT #16.
4. M = male connector; F = female connector.
- (Ill. 6378)

Fig. 3-8
PDP-11/44 Cabling Configuration RLS 4.5 and later

Table 3-C
PDP-11 SUPPORT DOCUMENTATION SUMMARY

TITLE	ORDERING NUMBER
PDP-11/23+ System Manual	Note 1
PDP-11/44 System Manual	Note 1
LA-210 User Guide	Note 1
LA-120 User Guide	Note 1
LA-100 User Guide	Note 1
VT-100 User Guide	Note 1
VT-220 User Guide	Note 1

Note 1: This documentation is supplied with the associated equipment.

PERIPHERAL
EQUIPMENT

3.04 The following peripheral equipment devices can be ordered for use with the ACD-ADS feature:

- | ACD supervisor Video Display (VDT) terminals (connect to the PDP-11 processor)
- ACD Management Reports printers (connect to the PDP-11 processor)
- | ACD Load Management terminals (connect to the Meridian SL-1)
- | modems or other data equipment
- | connecting cables.

3.05 Supervisor VDT. ACD-ADS can support a maximum of:

- | 8 Video Display terminals.
- | 16 Video Display terminals (11/44 RLS 4.5 and later). The terminal recommended for this application is the VT-220 (see Part 2 Table 2-C); The VT100 (Table 3-D) can be used. The terminal must be configured to operate at 4800 baud in full-duplex mode.

Table 3-D
VT100 TYPE VIDEO DISPLAY TERMINALS

Characteristics:

- | | |
|--|------------------------------------|
| Baud rate: 50 to 19,200 b/s | Normal or reverse screen image |
| Format: 24 lines x 80 characters | Adjustable tabs |
| Characters: 7 x 9 dot matrix with descenders | Full-duplex operation |
| Character set: 94 displayable-character ASCII set and 32-character special line-drawing graphics set | Keyboard selectable features |
| ● Double width/double height characters | Non-volatile set-up memory |
| Standard numeric and function keypads | Cursor control keys |
| Bidirectional vertical scrolling | ANSI/VT52 command modes |
| ● Selectable smooth or jump scrolling | 20 character answerback message |
| ● Split-screen capability | Selectable XON/XOFF buffer control |
| ● Format: 14 lines x 132 characters, selectable | Self -test diagnostics |

Ordering Information

VT100-AA(AB)	Video display terminal and U.S. power cord and plug.	VT100-VC(WD)	VT100-W with French-Canadian keyboard.
VT100-WA(WB)	Video terminal with advanced video, word processing keyboard, and U.S. power cord and plug.	VT100-WE(WF)	VT100-V with French keyboard.

Note: Cables are not provided and must be ordered separately. The recommended cables include **BC22B-xx** for connection to a modem, and **BC22D-xx**, **BC22E-xx**, and **BC22F-xx** for local connection to a host.

3.06 Reports Printer. ACD-ADS supports a maximum of 1 hard-copy printer per ACD-ADS defined customer (see Table 3-E). These printers+ are not included in the system package and must be ordered separately.

Table 3-E
REPORTS PRINTERS AVAILABLE

MODEL	KEYBOARD & QUALITY	PRINT SPEED	PAPER FEED	SPECIAL FEATURES
LA100-	(CA)Keyboard send/ receive or (ZA)receive only	240 cps dot matrix	Friction, tractor sheet optional	Plug-in fonts PC-compatible
LA120-	(DA)Keyboard send/ receive or (RA)receive only	180 cps dot matrix	Tractor only	For high duty cycle environment

3.07 Load Management Terminal. Each ACD-ADS customer can be equipped with one Load Management terminal (connected to the Meridian SL-1). The LA210 described in Part 2, Table 2-C is preferred+ but, either the LA100 or LA120 can be used (see Table 3-E).←
Alternatively, any 20 mA, RS232-D and ASCII-compatible device with keyboard may be used.

3.08 The Load Management function can, if desired, be performed from a VDT that is equipped as a senior supervisor position. This requires that a suitable customer-provided switching device be equipped to enable the senior supervisor to select between the ACD status display function of the auxiliary processor and the Load Management function of the Meridian SL-1. (Alternatively, the Meridian SL-1 Lanstar Data ← Service feature can be used to provide this switching function.) A ← cabling arrangement to support this option is shown in Part 2, Fig. 2-3. ← Use of this option means that a hard-copy record of Load Management transactions is not maintained, and ACD status displays are not available while in the Load Management mode.

3.09 Similarly, the Management Reports printer can be used to perform a Load Management function as well as being used to print reports. This option also requires that a suitable customer-provided switching device be equipped to alternate the functions. Use of this option means that some management reports may be lost while the printer is switched to Load Management mode. The lost reports would have to be requested a second time when the printer is restored to the Management Reports mode.

3.10 If on-site cabling between VDTs and the PDP-11 processor exceeds 50 ft (15 m), the use of Limited Distant Data Sets (LDDS) is recommended. If the VDTs are located in separate buildings from the PDP-11, an appropriate asynchronous modem and telephone facilities must be used. Modems are also required if the auxiliary processor's maintenance and administration port is to be accessed from a remote location.

DEC ACCESS TO
PROCESSOR

3.11 Digital Equipment Corporation personnel periodically require TTY **access** to the auxiliary processor for preventive maintenance purposes. The Meridian SL-1 maintenance TTY may be used (if a customer-provided switching device) configured and cabled as shown in Part 2, Fig. 2-4, is installed. ←

CABLING
CONFIGURATION

3.12 Fig. 3-2 or 3-4 (**PDP-11/23+**), and Fig. 3-6 or 3-71 (**PDP-11/44**) ← show typical cabling configurations for the auxiliary processor and related peripheral equipment. Follow DEC recommendations for cable lengths, modem requirements, etc. Ensure that at the time of processor installation, DEC personnel clearly designate the processor logical unit number (**TT0-TT21**) on all cables. Ensure that DEC personnel clearly identify logical unit numbers on the distribution panel of the **DZ11-DP** cards for the **PDP-11/44**.

SL-1*
BUSINESS COMMUNICATIONS SYSTEM
AUTOMATIC CALL DISTRIBUTION
BASIC FEATURES
OPERATION AND TESTS

CONTENTS	PAGE	CONTENTS	PAGE
1. GENERAL.....	2	4. SUPERVISOR FEATURES.....	7
2. TESTING REQUIREMENTS	2	Charts	
3. AGENT FEATURES..	3	4-1 AGENT Lamp	7
Charts		4-2 DISPLAY AGENTS Key	8
3-1 NOT READY Key	3	4-3 DISPLAY QUEUE Key/Lamp	9
3-2 Digit Display Verification	4	5. SYSTEM FEATURES	10
3-3 MAKE BUSY Key.....	5	Charts	
3-4 Agent DN Key.....	6	5-1 Agent Queue Verification	10
		5-2 Call Queue Verification	11
		5-3 Recorded Announcement	
		Verification	13
		5-4 Priority Trunks Test	14

* SL-1 is a trademark of Northern Telecom Limited

1. GENERAL

1.01 This section describes the operation of the Automatic Call Distribution (ACD) feature as applied to **SL-1** system. These operations should be performed as tests after the implementation of the ACD feature. Satisfactory completion of the tests in this section confirms that the ACD feature is installed and operating properly.

1.02 This section describes tests only for the ACD 'Basic Features'. For tests of other features, consult the appropriate sections in the 553-2671-302 series (see Index, **553-2YY** 1-000).

1.03 The tests described in this section apply only to SL-1 type telephone sets. NE-500 or NE-2500 type stations may not be used for ACD agent or ACD supervisor sets. The conducting of tests in this section may require two people for efficient completion.

2. TESTING REQUIREMENTS

2.01 The following requirements must be met before attempting the test in this section:

- Agent terminal installation must be completed as described in station installation practices (553-2001-211).

- Data modifications and additions for the ACD feature must be inputted by using service change programs in 553-2001-220 and 553-2001-221.
- SL- 1 telephone set tests in 553-2001-305 should be completed before testing the ACD feature.
- If certain incoming call types (routes) are to receive priority treatment in the call queue, calls to the ACD-DN can be modified by the priority assignments. This should be taken into account during testing.
- The SL-1 must be set up in such a way that at least three or four simultaneous incoming calls to the ACD-DN can be readily made by the testing personnel. This may be accomplished by the temporary assignment of tie lines from the CO or another PBX, or by other methods deemed suitable for that particular installation.
- o Some tests require large numbers of calls to be placed in the ACD-DN queue unless the office data is modified. To make these tests more manageable, it is recommended that service changeable thresholds be set to very low values. The thresholds can be reset after testing. See 553-2001-220 and 553-2001-221 for service **change** information.

3. AGENT FEATURES,

CHART 3-1
NOT READY KEY

STEP	PROCEDURE	VERIFICATION
1	Press the NOT READY key on all agent terminals associated with ACD-DN.	The lamp associated with the NOT READY key is lit steadily at all agent terminals.
2	Originate a call to the ACD-DN.	Ring-back tone is heard by calling party. Call is not presented to any agent terminal.
3	Press the NOT READY key (i.e., deactivate the feature) at one agent terminal. Note: Tone ringing is replaced by a 3-second burst of buzz tone, if agent terminals are using headset or plug-in handset.	The lamp associated with the NOT READY key is extinguished. Tone ringing is heard at the agent terminal. The lamp associated with the IN-CALLS key flashes at 60 ipm.
4	Press the NOT READY key (deactivating the feature) at another ACD terminal.	The lamp associated with the NOT READY key is extinguished.
5	Operate NOT READY key released in Step 3.	Tone ringing ceases. The lamp associated with the IN-CALLS key goes out. The lamp associated with the NOT READY key is lit steadily. Tone ringing is heard at agent idled in Step 4. The lamp associated with the IN-CALLS key flashes at agent terminal (Step 4).
6	Answer the call which is presented at the ACD terminal.	Ringling ceases. The lamp associated with the IN-CALLS key is lit steadily. Voice connection between caller and ACD position.
7	Disconnect the call by pressing the NOT READY key at ACD terminal.	Voice connection is broken. The lamp associated with the IN-CALLS key is extinguished. NOT READY key is lit steadily.
8	Release NOT READY key at all terminals and idle all ACD terminals.	The lamp associated with all NOT READY keys are extinguished. All agent terminal lamps go out. All agent terminal keys deactivated.

CHART 3-2
DIGIT DISPLAY VERIFICATION

STEP	PROCEDURE	VERIFICATION
1	<p>Originate a call to the ACD-DN.</p> <p><i>Note:</i> If agent terminals are equipped for headset or plug-in handset operation, tone ringing is replaced by a 3-second burst of buzz tone.</p>	<p>The lamp associated with the IN-CALLS key flashes. Tone ringing is heard at one agent terminal. Digit display indicates:</p> <p>(1) Calls within PBX-ACD; DN of the calling party.</p> <p>(2) Incoming calls to PBX-ACD;</p> <ul style="list-style-type: none"> ● trunk access code number (trunk route member number of the incoming call allowing the agent to give proper answer treatment) ● trunk member number identifying a specific trunk within the group of trunks (may be used for identifying faulty trunks).
2	Agent answers the call.	The lamp associated with the IN-CALLS is lit steadily . Ringing tone is removed. 2-way voice connection. Display remains lit, indicating trunk access code number and trunk member number or PBX-DN of the calling party.
3	Both parties disconnect.	The lamp associated with the IN-CALLS key goes out. Display is removed.
4	Repeat Steps 1 through 3 until a call has been presented to each ACD terminal.	See Steps 1 through 3.

CHART 3-3
MAKE BUSY KEY

STEP	PROCEDURE	VERIFICATION
1	Operate MAKE BUSY key at all agent terminals (except one).	MAKE BUSY lamp is lit steadily at agent terminals with MAKE BUSY key activated.
2	Originate a call to the ACD-DN.	The lamp associated with the IN-CALLS key flashes. Tone ringing is heard at the agent terminal without MAKE BUSY lamp lit. If agent terminals are equipped with headsets or plug-in handsets tone ringing is replaced by a 3-second burst of buzz tone. <i>Note:</i> Call is not presented to any agent terminal with MAKE BUSY key activated.
3	Operate MAKE BUSY key at the agent terminal presented with the call from Step 2.	The lamp associated with the MAKE BUSY key is lit steadily. Tone ringing ceases. IN-CALLS lamp goes out. Call will be directed to night service. (See Step 6).
4	Abandon call originated in Step 2. (Calling party hangs up). Night Service on ACD-DN.	
5	Ensure that all terminals assigned to the ACD-DN have a MAKE BUSY key assigned and that it is activated.	The lamp associated with each MAKE BUSY key is lit steadily.
6	Originate a call to the ACD-DN. Either one or both of the following can occur, depending on equipped features:	<ul style="list-style-type: none"> ● If night recorded announcement RAN equipped, the calling party hears recorded announcement. If call forward for Night Service equipped, the call is forwarded to a night number assigned for ACD.
7	Abandon the call.	
8	Deactivate all MAKE BUSY keys.	The lamp associated with the MAKE BUSY key at each terminal goes out. <i>Note:</i> Night Service treatment may be optionally specified, see 553-2001-221, Overlay Program 23.

CHART 3-4
AGENT DN KEY

Ensure that at least three agent terminals for the ACD-DN concerned are 'available' (i.e., MAKE BUSY and NOT READY lamps dark with no other features activated).

STEP	PROCEDURE	VERIFICATION
1	<p>Originate a call to the ACD-DN. (Do not answer call at the agent terminal.)</p> <p>Press the agent DN key at the first agent terminal.</p> <p>At the first agent terminal, dial the agent DN associated with the second agent terminal.</p> <p>At the second agent terminal depress the individual agent DN key next to flashing DN lamp.</p>	<p>Call is presented to the first agent where the lamp associated with the IN-CALLS key flashes. Ringing or buzz tone is heard at the first agent terminal.</p> <p>At the first agent terminal:</p> <ul style="list-style-type: none"> ● IN-CALLS lamp is extinguished. DN lamp is steadily lit. Dial tone is heard. The call originated in Step 1 is presented to the second agent terminal at which: <ul style="list-style-type: none"> (1) IN-CALLS lamp flashes, (2) Ringing or buzz tone is heard from speaker. <p>Lamp associated with DN of the second agent flashes. Ringing or buzz tone is heard at the second agent terminal.</p> <p>At the second agent terminal:</p> <ul style="list-style-type: none"> IN-CALLS lamp extinguished. Ringing tone ceases ACD call of Step 1 now presented to a third agent terminal. <p>At third agent terminal:</p> <ul style="list-style-type: none"> IN-CALLS lamp flashes. ringing or buzz tone heard.
5	<p>Disconnect all calls in progress. (Release DN key at the first and second agent terminals and abandon call originated in Step 1).</p>	<p>All agent lamps go out. All agent terminals idle.</p>

4. SUPERVISOR FEATURES

CHART 4-1
AGENT LAMP

Operation of the AGENT key has no effect on its associated lamp. All AGENT lamps are active simultaneously and are updated whenever an agent's status changes.

STEP	PROCEDURE	VERIFICATION
1	Make a call to the ACD-DN for the agent set concerned.	Ringing is heard at agent terminal.
2	At ringing agent set, either go off-hook or depress IN-CALLS key.	Ringing is silenced. At the ACD supervisor set under test, the AGENT lamp associated with the answering agent is steadily lit (indicating "busy on ACD call").
3	At agent set, depress NOT READY key.	At agent set, IN-CALLS lamp goes dark, NOT READY key is lit. At supervisor set, AGENT lamp is still steadily lit (indicating "busy on ACD call").
4	At agent set, depress NOT READY or IN-CALLS key. Do not make any calls to the ACD-DN.	At agent set, NOT READY lamp is extinguished. At supervisor set, AGENT lamp flashes at 60 ipm (indicating "waiting for ACD call").
5	At agent set, depress agent DN key.	At ACD set, dial tone heard. At supervisor set, AGENT lamp winks (indicating "busy on non-ACD call").
6	Put agent position in unmaned state by going on-hook and depressing MAKE BUSY key.	At agent set, MAKE BUSY lamp is lit. At supervisor set, AGENT lamp is dark (indicating "agent set not manned").

CHART 4-2
 DISPLAY AGENTS KEY

This key does not require an associated lamp.

STEP	PROCEDURE	VERIFICATION
1	Make various sets in the ACD queue busy on ACD calls, waiting for ACD calls, busy non -ACD calls, and unstaffed (see Chart 4-1).	
2	Depress DISPLAY AGENTS key on supervisor set.	<p>Digit display shows count of the number of agents reporting to this supervisor in each of 4 reported states at the moment of operation of the DISPLAY AGENTS key. The display is of the form:</p> <p>AA-BB-CC-DD</p> <p>Where:</p> <p>AA = number of agents busy on ACD calls (including post-call work time and including supervisor).</p> <p>BB = number of agents waiting for ACD calls (i.e., off-hook or IN-CALLS key depressed).</p> <p>CC = number of agents busy on non-ACD calls.</p> <p>DD = number of unstaffed agent terminals.</p> <p>The sum of these numbers is equal to the number of AGENT keys assigned to this supervisor set, as well as the supervisor set itself.</p>

CHART 4-3
DISPLAY QUEUE KEY/LAMP

Operating the DISPLAY Qx key activates only the digit display. It has no effect on the associated lamp, which is constantly updated by the system.

STEP	PROCEDURE	VERIFICATION
1	Add calls to the queue, one at a time, while periodically observing the Qx lamp and depressing the DISPLAY Qx key and observing the digit display.	
2	Observe the DISPLAY Qx lamp associated with the queue under test.	<p>The lamp can assume any one of the following states:</p> <p>unlit – no calls in the queue are waiting for agents.</p> <p>lit – one or more calls are waiting for agents.</p>
3	Depress DISPLAY Qx key associated with queue under test.	<p>Digit display (QSU3 or QSU7 sets only) shows the status of the queue. The display takes the form:</p> <p>AAA-BBB-CCC</p> <p><i>Where:</i></p> <p>AAA is the number of calls waiting for a free agent terminal.</p> <p>BBB is the number of agents assigned to this queue and manned (regardless of which supervisor they are assigned to).</p> <p>c c c is the length of time (in seconds) the oldest call in the queue has been waiting for service.</p> <p>The digit display is updated each time the DISPLAY Qx key is depressed.</p>

5. SYSTEM FEATURES

CHART 5-1
AGENT QUEUE VERIFICATION

STEP	PROCEDURE	VERIFICATION
1	Ensure that each agent terminal (ACD position) assigned to the ACD-DN is idle.	No keys or lamps on the agent terminal are activated.
2	<p>Originate a call via an incoming trunk to the ACD-DN.</p> <p><i>Note:</i> If agent terminals are equipped with headsets or plug-in handsets a buzz tone of 3 seconds replaces tone ringing at agent terminals.</p>	<p>Ringing or buzz tone is heard at one ACD terminal. IN-CALLS lamp flashes at 60 ipm, at that agent terminal.</p>
3	Answer the call at the called agent terminal by depressing the IN-CALLS key.	The lamp associated with the IN-CALLS key is lit steadily. Ringing tone is removed. Voice connection is set up between the calling party and the ACD agent.
4	Calling party and agent disconnect.	The lamp associated with the IN-CALLS key goes out. Voice connection is broken.
5	Repeat Steps 2 through 4 until call has been presented to each agent terminal assigned to the ACD-DN.	See Steps 2 through 4.
6	Originate a call via an incoming trunk to the ACD-DN.	Call is presented to the first position call in Step 2. The lamp associated with the IN-CALLS key flashes. Tone ringing or buzz is heard at the agent terminal.
7	Repeat Steps 3 and 4.	

CHART 5-2
CALL QUEUE VERIFICATION

STEP	PROCEDURE	VERIFICATION
1	<p>Make all agent terminals assigned to the ACD-DN busy for ACD calls by:</p> <ul style="list-style-type: none"> ● operating NOT READY key at each terminal equipped, or ● completing calls to ACD-DN until each terminal is engaged, or ● engaging each agent terminal by completing calls to or originating calls from each agent DN. 	<p>The lamp associated with the NOT READY key is lit steadily.</p> <p>The lamp associated with IN-CALLS key is lit steadily. (See Chart 5-1 Steps 2 and 3).</p> <p>The lamp associated with DN key is lit steadily.</p>
2	Originate another call to the ACD-DN.	Ringback tone is heard by the calling party.
3	<p>Originate another call to the ACD-DN.</p> <p><i>Note:</i> If equipped with recorded announcement (RAN), calls originated in Step 3 may receive announcement.</p>	Ringback tone is heard by the calling party.
4	<p>Make one agent available to receive ACD calls (see Step 1).</p> <p><i>Note:</i> If agent terminals are equipped with headsets, tone ringing is replaced by a 3-second burst of buzz tone.</p>	<p>Lamp lit in Step 1 is extinguished.</p> <p>Tone ringing is heard at the agent terminal. The lamp associated with IN-CALLS key flashes (60 ipm).</p>
5	Answer the call ringing at the agent terminal by depressing the IN-CALLS key.	The lamp associated with IN-CALLS lamp is lit steadily. Tone ringing ceases. Voice connection between the caller (Step 2) and the agent.
6	Disconnect from the call answered in Step 5 by depressing the RLS key. Do not answer the next call at this time.	The lamp associated with the IN-CALLS key is extinguished then flashes again. Tone ringing is heard at the agent terminal.

Chart Continued

CHART 5-2 Continued
CALL QUEUE VERIFICATION

STEP	PROCEDURE	VERIFICATION
7	Originate a call to the ACD-DN.	Ringback tone is heard by the caller.
8	At the agent terminal with the flashing lamp associated with the IN-CALLS key operate: <ul style="list-style-type: none"> I NOT READY key, or I MAKE BUSY, or ● DN key 	<p>The lamp associated with the IN-CALLS key goes out, and in addition; the lamp associated with the NOT READY key is lit steadily.</p> <p>The lamp associated with the MAKE BUSY key is lit steadily.</p> <p>The lamp associated with the DN key is lit steadily.</p>
9	At another agent terminal allow an ACD call to be presented. (See Step 1).	Lamp lit in Step 1 is extinguished. The lamp associated with the IN-CALLS key flashes. Ringing or buzz tone is heard at the agent terminal.
10	Answer the new call at agent terminal.	The lamp associated with the IN-CALLS key is lit steadily. Ringing tone ceases. Voice connection between call originated in Step 3 and agent terminal.
11	Disconnect all calls and make all agents idle. (See Steps 1 and 3).	No keys or lamps are activated at any agent terminal.

CHART 5-3
RECORDED ANNOUNCEMENT VERIFICATION

Before starting this test, take note of the first and second RAN times in effect for the ACD-DN under test. See 553-2001-221, overlay 23. Put all the agent terminal on the ACD-DN into the 'not ready' state. Make sure that at least one agent terminal is not in the 'make busy' state.

STEP	PROCEDURE	VERIFICATION
1	Make a call to the ACD-DN.	<p>Caller hears ringback. After the first RAN time has elapsed, the caller hears the first RAN.</p> <p>After the second RAN time has elapsed, the caller hears the second RAN. The second RAN is repeated at each 'second RAN time' interval.</p>
2	Press the NOT READY key on one agent terminal.	The call is presented on the IN-CALLS key of the terminal. The indications of this (buzz tone, tone ringing, etc.) may vary, depending on whether 'call forcing ' is in effect or not.
3	Answer the call at the agent terminal (go off-hook and press the IN-CALLS key).	2-way conversation.
4	Without terminating the call in Step 3 make a second call to the ACD-DN.	Caller hears ringback, then first RAN, then second RAN as in Step 1.
5	Release both calls.	Both agent terminals are idle.

Note: RAN delays will be greater than the thresholds assigned in overlay 23 (FRT, SRT) if the 'delayed start' option is in effect (overlay 16).

CHART 5-4
PRIORITY TRUNKS TEST

STEP	PROCEDURE	VERIFICATION
1	<p>Make all agent terminals assigned to the ACD-DN busy to ACD calls by:</p> <ul style="list-style-type: none"> operating NOT READY key at each terminal. (Ensure that at least one agent terminal has the MAKE BUSY deactivated.) ● completing nonpriority calls to ACD-DN until each terminal is engaged or, engaging each agent by completing calls to each agent individual DN. 	<p>The lamp associated with NOT READY key is lit steadily.</p> <p>The lamp associated with IN-CALLS key is lit steadily. (See Chart 2-1 Steps 2 and 3).</p> <p>The lamp associated with DN key is lit steadily.</p>
2	Originate a nonpriority call to the ACD-DN.	Ringback tone is heard by the calling party.
3	<p>Originate a priority call to the ACD-DN.</p> <p><i>Note:</i> If equipped with record announcement (RAN), calls originated in Steps 2 and 3 may receive announcement.</p>	Ringback tone is heard by the calling party.
4	<p>Make one agent available to receive ACD calls (see Step 1).</p> <p><i>Note:</i> If agent terminals are equipped with headsets or plug-in handsets tone ringing is replaced by a 3-second burst of buzz tone.</p>	<p>Lamp lit in Step 1 is extinguished.</p> <p>Tone ringing is heard at the agent terminal. The lamp associated with IN-CALLS lamp flashes (60 ipm).</p>
5	Answer the call ringing at the agent terminal.	The lamp associated with IN-CALLS key is lit steadily. Tone ringing ceases. Voice connection between the priority caller (Step 3) and the agent.
6	Release the (priority) call.	The nonpriority call is now presented to the same agent terminal. Tone ringing is heard at the agent terminal, and the IN-CALLS lamp flashes.
7	Answer the call.	2-way conversation; and the IN-CALLS lamp is steadily lit.
8	Make all agent positions idle.	

BUSINESS COMMUNICATIONS **SYSTEM**

MERIDIAN SL-1*

AUTOMATIC CALL **DISTRIBUTION**
ADVANCED FEATURES:
OPERATION AND **TESTS**

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Reason for Reissue: Supplement dated 82 10 30 is included in this revision. This provides operation and test procedures for the following ACD feature enhancements introduced with Generic X11 Release 2: Extended Agent Observe, ACD Set **Login/Logout** and Walkaway/Return-ACD Set. Due to the extent of the miscellaneous changes throughout the practice, revision marks have not been included.

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1. GENERAL

1.01 This practice outlines procedures to test the operation of the Automatic Call Distribution (**ACD**) Advanced Features as they pertain to ACD agent and ACD supervisor positions. (Read 553-2671-101 for a detailed description of the ACD Advanced Features.)

Note: Information presented in this practice assumes that the operation of the ACD Basic Features (553-2671-300) has been successfully completed.

2. ACD AGENT FEATURES

2.01 The ACD Advanced Features package provides additional key/lamp functions to the SL-1 telephone set equipped at an ACD agent's position. The added key/lamp functions are:

- CALL SUPERVISOR key
- EMERGENCY key
- CALLS WAITING lamps.

CALL SUPERVISOR Key

2.02 The CALL SUPERVISOR key enables an ACD agent to call the supervisor (to which the agent is assigned) through a single key operation (i.e., without dialing). Chart 2-1 outlines the procedures to operate and test the functions of the CALL SUPERVISOR key.

EMERGENCY Key

2.03 The EMERGENCY key enables an ACD agent, upon being presented with an emergency call, to call the assigned supervisor through a single key operation (Chart 2-2). In addition, when the EMERGENCY key is operated by an ACD agent:

- a message is printed at the maintenance teletypewriter (TTY) which provides details of the emergency call connection
- an optional customer-provided recording device can be connected automatically to the emergency call.

CALLS WAITING Lamp

2.04 The CALLS WAITING lamp provides a visual indication of the number of ACD calls that are in the ACD queue and not yet answered. (The key associated with the CALLS WAITING lamp performs no function.) The CALLS WAITING lamp can assume any of the following states:

- Dark. There are no ACD calls awaiting answer.
- ▮ Steadily Lit. There are ACD calls awaiting answer. However, the queue can still accept calls overflowed from another ACD-Directory Number (ACD-DN).
- Slow Flash (**30 ipm**). There are ACD calls awaiting answer. The queue cannot receive calls overflowed from another ACD-DN.
- ▮ Fast Flash (**120 ipm**). There are ACD calls awaiting answer. Some incoming calls to this ACD-DN are being overflowed to another ACD-DN (where this option is equipped).

Chart 2-1
USING THE CALL SUPERVISOR KEY

STEP	ACTION	VERIFICATION
1a	If there is no call currently active on the IN-CALLS key, lift the handset and/or plug in the headset and press the CALL SUPERVISOR key.	The lamp associated with the CALL SUPERVISOR key becomes steadily lit. Ringback tone is heard if the supervisor is idle; busy tone if the supervisor is busy.
1b	If there is a call on the IN-CALLS key, press the CALL SUPERVISOR key.	The calling party is put on hold (i.e., IN-CALLS lamp flashes). The lamp associated with the CALL SUPERVISOR key becomes steadily lit. Ringback or busy tone is heard.
2	The supervisor answers the call (Chart 3-3).	Two-way conversation between the ACD agent and supervisor.
	<ul style="list-style-type: none"> When the conversation is complete, the agent can either go on-hook or press the CALL SUPERVISOR key again. 	<ul style="list-style-type: none"> The lamp associated with the CALL SUPERVISOR key goes dark.
	<ul style="list-style-type: none"> To add the calling party to the agent/supervisor conversation, the agent presses the CALL SUPERVISOR key again. 	<ul style="list-style-type: none"> The calling party is bridged into the conversation: the CALL SUPERVISOR lamp goes dark; the IN-CALLS lamp becomes steadily lit.
	<ul style="list-style-type: none"> ● To transfer the calling party to the supervisor, the agent presses the Release (RLS)key. 	<ul style="list-style-type: none"> The agent is removed from the conversation: the IN-CALLS lamp goes dark.
<p>Note: If the supervisor initiates a call to the agent (CALL AGENT key - Chart 3-2), the call is presented (flashing lamp) and answered via the CALL SUPERVISOR key.</p>		

Chart 2-2
USING THE EMERGENCY KEY

STEP	ACTION	VERIFICATION
1	A call established on the IN-CALLS key is deemed an emergency call.	The IN-CALLS lamp is steadily lit; two-way conversation with the calling party. If the set is equipped with digit display, the trunk route and member number of the incoming call are displayed.
2	Press the EMERGENCY key.	<p>Ringback tone is heard over the two-way conversation. The EMERGENCY lamp:</p> <ul style="list-style-type: none"> remains dark if neither the assigned supervisor nor recording device is available flashes (60 ipm) if either the assigned supervisor or recording device is available.
3	The assigned supervisor or recording device is bridged into the conversation.	The EMERGENCY lamp becomes steadily lit.

Note: Cancellation of the EMERGENCY feature can only be accomplished by releasing the ACD call.

3. ACD SUPERVISOR FEATURES

3.01 The ACD Advanced Features package provides additional key/lamp functions to the SL-1 telephone set equipped at an ACD supervisor's position. The added key/lamp functions are:

- OBSERVE AGENT key
- CALL AGENT key
 - | ANSWER AGENT key
 - | ANSWER EMERGENCY key
 - | DISPLAY QUEUE lamp
 - | INTERFLOW key
- . EXTENDED AGENT OBSERVE.

OBSERVE AGENT Key

3.02 This key, used in conjunction with a selected AGENT key, enables the supervisor to monitor the conversation between the selected agent and a called/calling party. Chart 3-1 outlines the procedures to use this key.

CALL AGENT Key

3.03 The CALL AGENT key, used in conjunction with a selected AGENT key, enables the supervisor to establish a two-way conversation with the selected agent (see Chart 3-2).

ANSWER AGENT Key

3.04 Calls from an ACD agent are presented on this key. Procedures for its use are detailed in Chart 3-3.

ANSWER EMERGENCY
Key

3.05 If an ACD agent presses the EMERGENCY key while established on a call, this key/lamp notifies the supervisor of the emergency call status and enables the supervisor to enter the call (see Chart 3-4).

DISPLAY QUEUE Lamp

3.06 The ACD Advanced Features package enhances the visual indications provided by the lamp associated with the DISPLAY QUEUE key. This lamp can assume one of the following states:

- Dark. There are no calls waiting for an agent to become available.
- Steadily Lit. There are calls waiting for agents; the ACD-DN can still receive calls overflowed from another ACD-DN queue.
- | Slow Flash (**30 ipm**). There are calls waiting for agents; the ACD-DN cannot accept calls overflowed from another ACD-DN.
- | Fast Flash (**120 ipm**). There are calls waiting for agents; some calls are being overflowed to another ACD-DN (where this option is equipped).

INTERFLOW Key

3.07 This key enables the supervisor, when an ACD-DN queue is overloaded, to route the excess ACD calls to another ACD-DN or Directory Number (see Chart 3-5).

Chart 3-1
USING THE OBSERVE AGENT KEY

STEP	ACTION	VERIFICATION
1	Press the NOT READY key.	The associated lamp lights steadily.
2	Press the OBSERVE AGENT key.	The associated lamp lights steadily.
3	Press an AGENT key for which the associated lamp is steadily lit (indicates the agent is involved with an ACD call). see Note.	The conversation between the selected ACD agent and calling party is heard. The digit display shows the POS-ID of the selected agent.
4a	To enter the conversation between the selected ACD agent and calling party, press the CALL AGENT key.	The CALL AGENT lamp lights steadily. Three-way conversation between supervisor, agent and calling party is established.
4b	To leave the observation mode, press the OBSERVE AGENT (or NOT READY) key.	Conversation between the agent and calling party can no longer be heard. The OBSERVE AGENT (or NOT READY) lamp goes dark.

Note: The supervisor can monitor selected agents one-at-a-time at this point by pressing different AGENT keys.

Chart 3-2
USING THE CALL AGENT KEY

STEP	ACTION	VERIFICATION
1	Press the NOT READY key.	The associated lamp lights steadily.
2	Press the CALL AGENT key.	The associated lamp lights steadily.
3	Press the AGENT key associated with the agent to be called. See Note.	The digit display shows the POS-ID of the selected agent; ringback tone is heard. When the called agent answers, two-way conversation is established.
4	To terminate the call, press the CALL AGENT (or NOT READY) key.	Conversation is terminated; the CALL AGENT (or NOT READY) lamp goes dark.
<p>Note: The supervisor can call selected agents, one-at-a-time, by pressing different AGENT keys.</p>		

Chart 3-3
USING THE ANSWER AGENT KEY

STEP	ACTION	VERIFICATION
1	An ACD agent has pressed the CALL SUPERVISOR key.	Tone ringing is heard at the associated supervisor set and the ANSWER AGENT lamp slow flashes (30 ipm).
2	Press the ANSWER AGENT key.	Tone ringing stops; the ANSWER AGENT lamp goes dark; two-way conversation between the calling agent and supervisor is established.
3	To terminate the call, press the ANSWER AGENT key.	The associated lamp goes dark.

Chart 3-4
 USING THE ANSWER EMERGENCY KEY

STEP	ACTION	VERIFICATION
1	An agent has received a call and pressed the EMERGENCY key.	Continuous tone ringing (or buzz if a headset is being used) is heard. The ANSWER EMERGENCY lamp slow flashes (30 ipm).
2	Press the ANSWER EMERGENCY key. (Press HOLD key first, if engaged with a call.)	Continuous tone ringing or buzz stops: ANSWER EMERGENCY lamp becomes steadily lit; three-way conversation between supervisor (see Note 3), agent and calling party is established. Digit display shows POS-ID of involved agent.
3	To terminate the call , press the RLS key.	Call terminated, ANSWER EMERGENCY lamp goes dark.

Note 1: Once released from an emergency call, the supervisor cannot return to it. However, the emergency call can be placed on hold by pressing the HOLD key.

Note 2: Once the supervisor has released from the emergency call, the agent cannot re-initiate emergency status for the same call.

Note 3: A 'listen-only' connection to the emergency call can be established by unplugging the headset/handset before pressing the ANSWER EMERGENCY key.

Chart 3-5
USING THE INTERFLOW KEY

STEP	ACTION	VERIFICATION
1	Call the interflow destination to ensure it is ready and able to accept additional calls.	
2	Press the appropriate INTERFLOW key.	The associated lamp slow flashes (30 ipm) .
3	To disable the interflow feature, press the INTERFLOW key again.	The associated lamp goes dark.

Note 1: Use the DISPLAY QUEUE key/lamp to determine when and for which ACD-DN interflow should be enabled.

Note 2: With interflow enabled, calls are only forwarded to the interflow destination when the number of calls waiting in the ACD-DN queue exceeds the overflow threshold and no other target queue is specified or available through automatic overflow.

Extended Agent
Observe

3.08 This feature, introduced in Generic X11 Release 2, allows an ACD supervisor to observe/call any ACD agent or observe another ACD supervisor by dialing the POS-ID assigned to the agent or supervisor. The feature enables the AGENT key/lamps to be eliminated from a supervisor's set. Chart 3-6 details the procedures required to observe an ACD agent; Chart 3-7 details procedures required to observe another supervisor; Chart 3-8 details new procedures for calling an ACD agent.

Chart 3-6
OBSERVING AN ACD AGENT

STEP	ACTION	VERIFICATION
1	Press the NOT READY key.	The NOT READY lamp lights steadily.
2	Press the OBSERVE AGENT key.	The OBSERVE AGENT lamp lights steadily.
3	Dial the Position-ID (POS-ID) of the agent to be observed.	<ul style="list-style-type: none"> ● The conversation between the selected agent and calling/called party(s) is heard. The digit display shows the POS-ID of the agent. ▫ If the agent is already being observed by another supervisor, busy tone is heard. ● If the dialed POS-ID is invalid or the set is not an ACD set, overflow tone is heard.
4	To enter the conversation between the agent and caller(s), press the CALL AGENT key.	The OBSERVE AGENT lamp slow flashes (30 ipm). A conference between supervisor, agent, and caller(s) is established.
5	To exit the observation mode, press the OBSERVE AGENT key.	The conference is ended. OBSERVE AGENT lamp goes dark.
6	To resume normal ACD operation, press the NOT READY key.	The NOT READY lamp goes dark. The set can now receive ACD calls.

Chart 3-7
OBSERVING AN ACD SUPERVISOR

STEP	ACTION	VERIFICATION
1	Press the NOT READY key.	The NOT READY lamp lights steadily.
2	Press the OBSERVE AGENT key.	The OBSERVE AGENT lamp lights steadily.
3	Dial the POS-ID of the supervisor to be observed.	<ul style="list-style-type: none"> The conversation between the selected supervisor and calling/called party(s) is heard. The digit display shows the POS-ID of the supervisor. If the selected supervisor is currently in observation mode, busy tone is heard. Overflow tone is heard if the POS-ID is invalid or if the set is not an ACD set. If the set attempting to observe a supervisor is not assigned an Allow Observation of Supervisor class of service, overflow tone is heard.
4	Press the CALL AGENT key to enter the conversation between the supervisor and connected party(s).	The OBSERVE AGENT lamp slow flashes (30 ipm). A conference between, observing supervisor, observed supervisor and connected party(s) is established.
5	Press the OBSERVE AGENT key.	The conference is ended. OBSERVE AGENT lamp goes dark.
6	To resume normal ACD operation, press the NOT READY key.	The NOT READY lamp goes dark. The set can now receive ACD calls.

Chart 3-6
CALLING AN ACD AGENT

STEP	ACTION	VERIFICATION
1	Press the NOT READY key.	The NOT READY lamp lights steadily.
2	Press the CALL AGENT key.	The associated lamp lights steadily.
3	Dial the POS-IQ of the agent to be called.	If agent is idle, the agent's set rings and the SUPERVISOR lamp flashes. If the agent is conversing with another supervisor via the SUPERVISOR key, busy tone is heard. If the agent is conversing on a key other than the IN-CALLS or SUPERVISOR key, the agent hears buzz.
4	Agent presses SUPERVISOR key to answer the call.	The SUPERVISOR lamp at the agent's set lights steadily. Conversation between agent and supervisor is established.
5	To terminate the call, press the CALL AGENT (or RLS) key.	Conversation is ended. CALL AGENT lamp goes dark.

4. ACD AGENT AND SUPERVISOR FEATURES

ACD Set **Login/Logout**

4.01 Customers equipped with Package C (Management Reports, Load Management) and using Generic X11 Release 2 or later must logon on an ACD set before access to ACD features is allowed. Optionally, the customer may choose to operate in a Position-ID mode or an Agent-ID mode. Charts 4-1 to 4-3 provide **login** and **logout** procedures with or without the Agent-ID option being defined.

Chart 4-1
LOGIN OF AN ACD SET (AGENT-ID Dption not defined)

STEP	PROCEDURE	RESULT
1	Occupy an unmanned ACD position. Ensure that the headset/handset is unplugged from the set, then press the IN-CALLS key.	Nothing happens.
2	Plug the headset/handset into the lower jacks on the left-hand side of the set (QSU6B or QSU7C), or into the jacks on the front of the set (QSU6C or QSU7D).	Access to all features except the receiving of calls on the IN-CALLS key is allowed from the set.
3	Press the IN-CALLS key of the set.	<ul style="list-style-type: none"> The IN-CALLS key Light-Emitting Diode (LED) remains unlit. ● The NOT BEADY key LED lights. ● The MAKE BUSY key LED goes out if it was previously lit. The ACD-DN and supervisor position to which the agent position is assigned are shown in the digit display (if equipped) of the set. The position is now logged in and access to all ACD features is allowed.
4	Perform the ACD set operations and tests described in 553-2671-300.	

Chart 4-2
LOGIN OF AN ACD SET (AGENT-ID Option is defined)

STEP	PROCEDURE	RESULT
1	Perform Steps 1 and 2 of Chart 4-1.	As outlined in Chart 4-1.
2	Press the IN-CALLS key of the set.	The LED associated with the IN-CALLS key lights and a special (interrupted) dial tone is heard.
3	Key in the assigned 4-digit (0001-9999) Agent-ID code on the dial pad of the set noting one of the following : <ul style="list-style-type: none"> 1 The Agent-ID code is within range and no one else is already logged in with that code. 1 The Agent-ID code is out of range or the normal digit timeout elapsed before all four digits have been keyed in. 1 Someone else is currently logged in with your Agent-ID code. 	<p>As each digit is keyed in it appears in the digit display (if equipped) of the set.</p> <p>The IN-CALLS (and MSB, if previously lit) key LED goes out, the NOT BEADY key LED lights, and the ACD-DN and supervisor position to which the set is assigned are shown in the digit display (if equipped) of the set. The set is now logged in and access to all ACD features is allowed.</p> <p>Overflow tone is heard and, after normal timeout of overflow tone, the IN-CALLS key LED goes out. The login procedure must be repeated.</p> <p>Busy tone is heard and, after normal timeout of busy tone, the IN-CALLS key LED goes out.</p>

Chart 4-3
LOGGING OUT FROM A LOGGED IN ACD SET

STEP	PROCEDURE	RESULT
1	Occupy an agent position which is currently logged in but <ul style="list-style-type: none"> has no call active on the set, or has no calls held on the set. 	<ul style="list-style-type: none"> all feature keys are unlit the IN-CALLS or DN key LED is not fast flashing at 120 ipm. (Hold is not initiated when the NOT READY key LED is lit.)
↗ 2a	No MAKE BUSY key. Unplug the headset/handset from the set or place the handset on-hook.	The set is now logged out. This assumes that agents on a queue basis are allowed to log out by unplugging the headset/handset, or by pressing the set MAKE BUSY key. Unplugging will also cause log out when there is no MAKE BUSY key for the agent(s) to use. To regain access to ACD features, the login procedures in Chart 4-1 or 4-2 must be performed.
4		
2b	MAKE BUSY key exists. Press the MAKE BUSY key.	MAKE BUSY key LED goes from unlit to lit and the result is the same as described in step 2a.

Walkaway/Return ACD Set

4.02 This feature is available in Generic X11 Release 2 and later. Once logged in on an ACD set an agent or supervisor can leave the ACD position (walkaway) for a time and return to the position without having to **login** the ACD set again. Refer to Chart 4-4 for the **Walkaway/Return** procedure.

Chart 4-4

WALKAWAY /RETURN ACD SET

STEP	PROCEDURE	RESULT
WALKING AWAY FROM THE SET		
1	Using the procedures in Chart 4-1 or 4-2 login to an unoccupied agent position.	The NOT READY key LED lights.
2	Press the NOT READY key to allow ACD calls to be presented to the position.	The NOT READY key LED goes out .
3	a Manual Answer. When an ACD call is presented to the position (i.e., the IN-CALLS key LED flashes at 60 ipm and tone ringing is heard), press the IN-CALLS key to answer the call.	Tone ringing stops, the IN-CALLS key LED lights steadily, and a voice path is established between you and the calling party.
3b	Call Forcing. When an ACD call is presented to the position you will hear a 500-ms burst of tone and then be connected to the calling party.	As in Step 3a .
4	Inform the calling party that you are about to leave the line momentarily, then press the HOLD key of the set.	The IN-CALLS key LED fast flashes at 120 ipm to indicate the calling party is on hold.
5	Unplug the headset/handset from the set.	The IN-CALLS key LED on the set continues to fast flash.
RETURNING TO THE SET		
6	Plug the headset/handset into the set.	The IN-CALLS key LED continues to fast flash.
7	Press the IN-CALLS key.	The IN-CALLS key LED lights steadily. Voice communication is established between you and the calling party once again.

Note 1: The procedures in this chart also apply to calls originated from or received on the DN key of the set. **Walkaway** and return are also possible when the set is in the NOT READY state: i.e., the HOLD key is operated when the NOT READY key LED is lit. In either case, the DN key LED or NOT READY key LED will change from steadily lit to fast flashing (**120 ipm**) when the HOLD key is pressed and the headset/handset is unplugged from the set.

Note 2: If the calling party disconnects while the agent is in walkaway, the IN-CALLS key LED goes dark and the NOT READY key LED flashes. When the agent returns from the **walkaway** and plugs the headset/handset into the set, the NOT READY key LED lights steadily. The agent must press the IN-CALLS or NOT READY key to reenter the agent queue.

PRACTICE 553-2671-301



DN Key Activation

4.03 This feature allows the agent to place or receive calls using the DN key without logging on. Chart 4-5 details this DN key feature.

Chart 4-5

DN KEY ACTIVATION FOR ACD SETS

STEP	PROCEDURE	RESULT
1	Occupy an agent position which is currently logged in.	All active features operate as usual.
2	Press a DN key (not a ACD DN key) to	
	a) initiate a call, or	The associated LED indicator is lit and dial tone received.
	b) answer a non-ACD call.	Ringling ceases and the LED goes from flashing to steadily lit.

Note: The same results will occur for Step 2 if the procedure for Step 1 is “Occupy an agent position which is not restricted from using DN keys when logged out (or logged in)“.



BUSINESS COMMUNICATIONS SYSTEM

SL-1*

AUTOMATIC CALL DISTRIBUTION -
AUXILIARY DATA SYSTEM:
TELEPHONE SET OPERATION AND TESTS

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* SL-1 is a trademark of Northern Telecom Limited

1. GENERAL

1.01 This practice details the procedures required to USC Automatic Call Distribution (ACD) telephone sets when the SL-1 system is equipped with the Automatic Call Distribution - Auxiliary Data System (ACD-ADS). These procedures enable telephone set access to the ACD feature and must be performed before the tests in 553-2671-301 and -302 can be performed.

1.02 The SL-1 system is supplemented with an auxiliary processor (i.e., minicomputer) when the ACD-ADS feature is equipped. The auxiliary processor is located externally to the SL-1 and is connected to it through a high-speed link and a low-speed link. The prime function of the auxiliary processor is to calculate and store information (sent to it from the SL-1) relevant to incoming ACD call volume (demand) and agent capacity to serve it adequately (supply). This information includes:

- the assignment of agent positions to a supervisor position
 - | the assignment of ACD trunks to ACD-Director! Numbers (ACD-DN)
 - | the status of each agent position (idle, busy, etc.)
- the number of agent positions (spare, occupied, etc.)

1.03 The status of calls being queued, answered, released, etc., is sent to the auxiliary processor from the SL-1 to update the stored information whenever a change occurs. A supervisor or senior supervisor position which is assigned a Video Display (VDT) terminal, can receive a display which shows the current status of the ACD operation. The VDT is connected to the auxiliary processor. A more detailed description of the SL-1 ACD-ADS feature is contained in 553-2671-104.

2. ACD SET OPERATION AND TESTS

AGENT-ID/ POSITION-ID

2.04 Customers with the ACD-ADS feature have the option of operating in one of two modes of operation: Agent-ID mode or Position-ID mode. In the Agent-ID mode, historical performance data for an individual agent (person) is accumulated in the auxiliary processor, independent of the telephone position that the agent uses. In the Position-ID mode, the information is collected for an ACD position (telephone). If the Agent-ID mode of operation is chosen, each ACD agent is assigned an individual 4-digit (0001-9999) Agent-ID code by the senior supervisor (553-2671-105). This code is used as part of the login procedure.

LOGIN AND LDGOUT

2.02 Regardless of which mode of operation is selected by the customer, all ACD sets must be enabled individually for ACD operation by an ACD login procedure (Charts 2-1 or 2-2). Until an ACD set is logged in to a specific agent position, no ACD calls are presented to the set on the IN-CALLS key. However, access to all other features is allowed. Similarly, an ACD set which is already logged in, can be denied access to incoming ACD calls through an ACD set logout procedure (Chart 2-3).

WALKAWAY AND RETURN

2.03 Once an agent is logged in to a particular agent position, it is possible that, while handling a call or during Post Call Processing (PCP), the agent is required to leave the position and unplug the headset/handset from the set. To avoid an unnecessary logout and subsequent login upon return to the position, a walkaway and return procedure (Chart 2-4) is provided.

2.04 An agent can, by pressing the HOLD key and unplugging the headset/handset, initiate a walkaway under the following conditions:

- (a) Direct call-processing (IN-CALLS key LED lit)
- (b) Post call-processing (NOT READY key LED lit)
- (c) Non-ACD call handling (DN key LED lit).

2.05 Removal of the headset/handset under any conditions other than those listed will cause a logout of the position.

VDT DISPLAY OF POSITION STATUS

2.06 A supervisor position that is equipped with a VDT can retrieve from the auxiliary processor a display of the current status of the ACD operation. This display is updated whenever an agent position changes state: e.g., logout, walkaway, etc. Although Charts 2-1 through 2-4 deal primarily with ACD set operation, the effect of these operations on the VDT display is also noted, and should be verified during the ACD set tests to ensure communication between the SL-1 and the auxiliary processor. Table 2-A lists the VDT display characters which reflect the status of an ACD position.

PRACTICE 553 -2671-304

Table 2-A
MEANING OF VDT DISPLAY CHARACTERS

DISPLAY CHARACTER	POSITION STATUS
L	Agent is logged in but status is unknown (Note)
A	Agent is active on an ACD call
P	Agent is in post call-processing mode
D	Agent is active on a DN call
W	Agent is waiting for an ACD call
S	The agent position is spare
X	Appended to A, P, or D to indicate agent has walked away from the position. The X is removed when the agent returns to the position.

Note: This status appears against a position if the SL-1 initializes while an agent is logged in. When the SL-1 initializes, the position status (A, P, D, W) is lost to the auxiliary processor. The L status remains against the position until the set (or call) changes states (which sends a message to the auxiliary processor to update the status).

Chart 2-1
LOGIN OF AN ACD SET (AGENT-ID OPTION NOT ENABLED)

STEP	PROCEDURE	RESULT
1	Check the Agent Position Status Display on the supervisor's VDT to determine the current status of the agent position (telephone) to be occupied.	The VDT display should indicate a position status of Spare (S). Any other indication (i.e., P, D, A, etc.) means that the agent position is already logged in.
2	Ensure that the headset/handset is unplugged from the set, then press the IN-CALLS key.	Nothing happens.
3	Plug the headset/handset into the jacks on the front of the QSU6 or QSU7 set (Fig. 2-1).	Access to all features, except the receiving of calls on the IN-CALLS key is allowed from the set.
4	Press the IN-CALLS key of the set.	<ul style="list-style-type: none"> The IN-CALLS key light-emitting diode (LED) remains unlit. The NOT READY key LED lights. The MAKE BUSY key LED goes out, if it was previously lit. The ACD-DN and supervisor position to which the agent position is assigned, are shown in the digit display (if equipped) of the set. The status of the position, as indicated on the supervisor's VDT, changes from Spare (S) to Post Call-Processing (P). The position is now logged in, and access to all ACD features is allowed.
5	Perform the ACD set operations and tests described in 553-2671-300 and -301.	

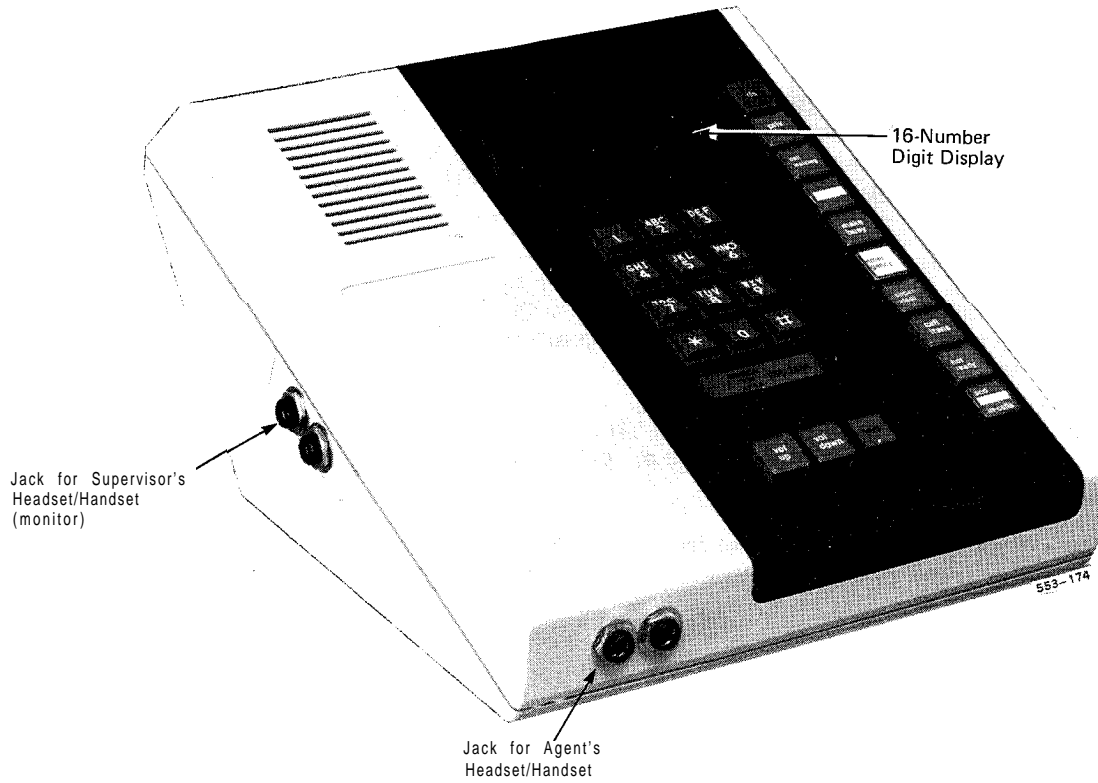


Fig. 2-1
Typical ACD Position **SL-1 Set (QSU7)**

Chart 2-2
LOGIN OF AN ACD SET (AGENT-ID OPTION IS ENABLED)

STEP	PROCEDURE	RESULT
1	Perform Steps 1, 2, and 3 of Chart 2-1.	As outlined in Chart 2-1.
2	Press the IN-CALLS key of the set.	The LED associated with the IN-CALLS key lights, and a special (interrupted) dial tone is heard.
3	Key in the assigned 4-digit (0001-9999) Agent-ID code on the dial pad of the set. (a) If the Agent-ID code is within range and if no one else is already logged in with that code: (b) If the Agent-ID code is out of range or if normal digit timeout elapsed before all four digits had been keyed in: (c) If someone else is logged in with your Agent-ID code: (d) If the code is within range, but not defined in the agent database (maintained by the senior supervisor):	As each digit is keyed in, it appears in the digit display (if equipped) of the set. The IN-CALLS (and MSB, if previously lit) key LED goes out, the NOT READY key LED lights, and the ACD-DN and supervisor position to which the set is assigned are shown in the digit display (if equipped) of the set. The set is now logged in, and access to all ACD features is allowed. The supervisor VDT status display is updated to show the agent name beside the position in the display and the status changes from (s) to (p). You hear overflow tone, and after normal timeout of overflow tone, the IN-CALLS key LED goes out. The login procedure must be repeated. You hear busy tone and, after normal timeout of busy tone, the IN-CALLS key LED goes out. Login occurs as in Step 3(a). Instead of the agent name appearing in the VDT display for the position, *****XXXX is shown, where XXXX is the code inputted by the agent.

Chart 2-3
 LOGGING OUT FROM A LOGGED IN ACD SET

STEP	PROCEDURE	RESULT
1	Occupy an agent position which is currently logged in but has no calls active on the set. or has no calls held on the set	All feature keys are unlit. The IN-CALLS or DN key LED is not flashing at 120 ipm. Hold is not initiated when the NOT READY key LED is lit.
2	(a) No MAKE BUSY key. Unplug the headset/ handset from the set or or place handset on-hook.	The position status on the supervisor's VDT display is changed to Spare (S). The set is now logged out. This assumes that agents on a queue basis are allowed to logout by unplugging the handset/ headset or by pressing the MAKE BUSY KEY. Unplugging will also cause logout when there is no MAKE BUSY key for the agent(s) to use. To regain access to ACD features, the login procedures in Charts 2-1 or 2-2 must be performed.
	(b) MAKE BUSY key. Press the MAKE BUSY key.	MAKE BUSY key LED lights and the result is the same as for (a).

Chart 2-4
WALKAWAY FROM AND RETURN TO AN ACD SET

STEP	PROCEDURE	RESULT
WALKING AWAY FROM THE SET		
1	Using the procedures in Charts 2-1 or 2-2, log in to an unoccupied agent position.	The NOT READY key LED lights, and the position status on the supervisor's VDT display changes from Spare (S) to Post call-processing (P) or Login (L).
2	Press the NOT READY key to allow ACD calls to be presented to the position	The NOT READY key LED goes out, and the supervisor's VDT display changes to Waiting for an ACD call (W).
3	(a) Manual Answer. When an ACD call is presented to the position (i.e., the IN-CALLS key LED flashes at 60 ipm and tone ringing is heard), press the IN-CALLS key to answer the call.	Tone ringing stops, the IN-CALLS key LED lights steadily, a voice path is established between you and the calling party, and the position status on the supervisor's VDT display changes from W, to A (Active) on an ACD call.
	(b) Call Forcing. When an ACD call is presented to the position you will hear a 500 ms burst of tone and then be connected to the calling party.	As in Step 3(a).
4	Inform then press the HOLD key of the set.	The IN-CALLS key LED fast flashes at 120 ipm to indicate the calling party is on hold. There is no position status change indicated on the supervisor's VDT display.
5	Unplug the headset/handset from the set.	The IN-CALLS key LED on the set continues to fast flash, and the position status on the supervisor's VDT display changes to AX, where X indicates walkaway status.
RETURNING TO THE SET		
6	Plug the headset/handset into the set.	The IN-CALLS key LED continues to fast flash, and the X is removed from the position status on the supervisor's VDT display to indicate that you have returned to the position.
7	Press the IN-CALLS key.	The IN-CALLS key LED lights steadily, and voice communication is established between you and the calling party once again.

Chart 2-4 Continued
WALKAWAY FROM AND RETURN TO AN ACD SET

STEP	PROCEDURE	RESULT
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Note 1: The procedures in this chart also apply to calls originated from or received on the DN key of the set. Walkaway and return are also possible when the set is in the NOT READY state: i.e., the HOLD key is operated and the NOT READY key LED is lit. In either case, the DN key LED or NOT READY key LED will change from steadily lit to fast flashing when the HOLD key is pressed and the headset/handset is unplugged from the set.

Note 2: If the calling party disconnects while the agent is in walkaway, the IN-CALLS key LED goes dark and the NOT READY key LED flashes. When the agent returns from the walkaway and plugs the headset/handset into the set, the NOT READY key LED lights steadily. The agent must press the IN-CALLS or NOT READY key to reenter the agent queue.

INTEGRATED SERVICES NETWORK

MERIDIAN SL-1*

AUTOMATIC CALL DISTRIBUTION —

AUXILIARY DATA SYSTEM:

ADMINISTRATION OF ACD-ADS CUSTOMERS

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Reason for Reissue: To add release 5.0 and 5.5 information.
Release 5.5 introduces the **MicroPDP-11/73** processor. Changes are
indicated by arrows in the margin.

* Meridian and SL-I are trademarks of Northern
Telecom Limited

1. GENERAL

1.01 This practice describes how the administration manager can access the Automatic Call Distribution - Auxiliary Data System (ACD-ADS) programs, and lists the input/output commands that can be used with the programs.

1.02 The Meridian SL-1 is supplemented with an auxiliary processor (i.e., minicomputer) when the ACD-ADS feature is equipped. The auxiliary processor is located externally to the Meridian SL-1 and connected to it through a high-speed and low-speed link. The high-speed link is used for transmission of high-volume, ACD-related messages between the Meridian SL-1 and auxiliary processor. The auxiliary processor uses the information from the ACD-related messages to calculate ACD call durations, percentages, etc., which are reflected in the management reports and status displays (553-2671-105). The function of the low-speed link is to transport low-volume maintenance messages between the Meridian SL-1 maintenance TTY and the auxiliary processor. Each link consists of a single Serial Data Interface (SDI) port in the Meridian SL-1, EIA RS232-C interface cable, and an interface port in the auxiliary processor. A more detailed description of the ACD-ADS feature can be read in 553-2671-104.

ADMINISTRATIVE PROGRAMS

1.03 Resident on the auxiliary processor are administrative programs which enable the telephone company to:

- | define to the auxiliary processor a new customer to be equipped with the ACD-ADS feature
- | update the system time used by the auxiliary processor
- | remove from the auxiliary processor the definition of an existing customer equipped with the ACD-ADS feature
- | change, add or delete the various options available to each customer equipped with the ACD-ADS feature.

Caution: All field updates must be completed at least 45 minutes before the midnight cross-over time, or the daily update will not be performed.

2. USING THE ADMINISTRATIVE PROGRAMS

ACCESSING THE AUXILIARY PROCESSOR

2.01 Access to the programs on the auxiliary processor can be gained in a number of ways:

- | from an on-site Meridian SL-1 maintenance Teletypewriter (TTY) i.e., with USER defined as 'mtc' in the configuration record (see Note)
- | from a remote TTY accessing the Meridian SL-1 through a dial-up port and modems. This port must have USER defined as 'mtc' in the configuration record (see Note).
- | from a TTY connected directly to a port on the auxiliary processor
- from a remote TTY accessing the auxiliary processor through a dial-up port and modems (e.g., the Meridian SL-1 Data Feature).

Note: The configuration record is an Meridian SL-1 overlay program (overlay 17) which is described in Practice 553-2001-221 for the Meridian SL-1 LE, VLE and XL or Practice 553-2YY1-310 for the Meridian SL-1 MS, N and XN.

2.02 Chart 2-1 outlines the operations required to establish communications between the Meridian SL-1 maintenance TTY and the auxiliary processor.

PROGRAMS

2.03 Once accessed to the auxiliary processor, the user inputs the command @MAINT. A menu of software programs/functions is presented.

2.04 A brief description follows of each program/function that is applicable to administration of ACD-ADS customers. A description of the other programs/functions shown in the menu can be obtained by inputting the question mark (?) command.

- (a) SVCCHG (S). The S command invokes the service change (SVCCHG) program of the auxiliary processor. This program enables the attributes of each customer equipped with the ACD-ADS feature to be defined/changed/deleted. Figure 2-1 illustrates the data items relevant to the SVCCHG program, and shows what is printed at the TTY for a simple administrative task. See Fig. 2-2 for RLS 4.5 and later. ←
- (b) RTECHG (R). The R command invokes the route change (RTECHG) program of the auxiliary processor. This program enables an 8-character alphanumeric route code to be assigned to each ACD trunk route defined for a customer. The route code is printed on route reports (553-2671-105) instead of the route number. When a change is made to a route code, the change is reflected in reports the same day the change is made. Fig. 2-3 shows what is printed at the TTY for a RTECHG task. See Fig. 2-4 for RLS 5.0 and later ←
- (c) HELP (?). The ? command presents a brief description of each of the programs that are listed in the @MAINT menu

4-J

(d) EXIT (*). This command exits the @MAINT mode. (Typically, this command is used if an @MAINT menu item is selected in error.) To reaccess the @MAINT menu, the command @MAINT must be input again. Note: In RLS 5.5 and later, when adding or deleting customers in SVVCHG there is no need to duplicate the action in RTECHG as it is done automatically.



LIMITATIONS

2.05 Ensure that all necessary SVCCHG functions have been completed before the auxiliary processor begins its End-of-Day (EOD) routines (00:00 to 00:05). The auxiliary processor EOD routines will abort the SVCCHG programs, if running, at the start of the routines. If RTECHG is running, the auxiliary processor waits until changes are completed before starting EOD routines. (See 553-2671-104 for a description of EOD routines.)

Agent -ID Option

2.06 If the Agent-ID mode is enabled on both the Meridian SL-1 and auxiliary processor, then an Agent-ID code must be inputted by an ACD agent when logging in to an ACD telephone. Statistics are then accumulated by the auxiliary processor, on the basis of the Agent-ID code, and reported in the following reports:

- | Agent First Login/Last Logout
- Agent All Login/Logout
- | Agent Detail Log.

2.07 If the Agent-ID mode is not enabled, then no entry of an Agent-ID is required and the above reports cannot be selected on the auxiliary processor.

2.08 If the customer is operating in the Agent-ID mode and decides to return to the Position-ID mode, the following steps must be performed.

- (1) Delete all report definitions that reference the agent login/logout reports.
- (2) Delete all Agent-ID codes from the Agent Data base.
- (3) Disable the Agent-ID option in the auxiliary processor via the SVCCHG program.

2.09 These changes will not take effect until the auxiliary processor has completed its regular End-of Day (EOD) routines (00:00 to 00:05). Once the EOD routines are completed, the Agent-ID option must be disabled in the Meridian SL-1 (ADS data block, overlay 23).

Time Changes

2.10 The Meridian SL-1 provides two real-time clocks, Meridian SL-1 time and auxiliary processor (AUX) time. On system load of the Meridian SL-1, the Meridian SL-1 will zero the Meridian SL-1 time but will not zero the AUX time if the auxiliary processor is linked to the Meridian SL-1 during SYSLOAD. During SYSLOAD, the AUX time stored in the Meridian SL-1 is being incremented. Hence, when the AUX is rebooted, the Meridian SL-1 will send two separate times to the AUX: a time of zero for the Meridian SL-1 time and a nonzero time for the AUX. The software in the AUX checks both times: if the Meridian SL-1 time is zero and the AUX time is not zero, the AUX will display and set its system time according to the AUX time in the Meridian SL-1 message.

2.11 If the Meridian SL-1 undergoes SYSLOAD without the AUX linked to it, the two times stored in the Meridian SL-1 will be zeroed. Then, when the AUX is rebooted, the time message sent to AUX will have both times equal to zero. The AUX will then halt and display the message informing the operator to set the Meridian **SL-1** time.

2.12 When the Meridian SL-1 time/date is changed, the new time/date is not sent automatically to the auxiliary processor. If the Meridian SL-1 time/date is changed, then access the auxiliary processor (Chart 2-1, Step 1) and input the following commands. (These commands will cause the auxiliary processor to reboot with the new time/date from the Meridian SL-1.1

@ACDABORT

@[1,2] STARTUP

Caution 1: When the Meridian SL-1 time is updated, care should be taken that a day boundary is not crossed. This could corrupt the previous day's stored data used in the historical reports.

Caution 2: If the Meridian SL-1 time is advanced by one hour, there will be a one-hour period in the reports which show no data, as that day has only 23 hours. If the Meridian SL-1 time is set back one hour, the data already collected for that hour will be lost as the auxiliary processor does not support a **25-hour** day.

2.13 Whenever possible, Meridian SL-1 time/date changes should be implemented when ACD traffic is very low to minimize the loss of stored historical data.

Chart 2-1
ACCESSING THE AUXILIARY PROCESSOR

STEP	PROCEDURE	RESULT
1	At the Meridian SL-1 maintenance TTY (or from a TTY accessing the Meridian SL-1 through a dial-up port), establish a link to the auxiliary processor with the LOGI, system password, and AX commands as shown in Fig. 2-1 or 2-2 (Fig 2-3 is applicable to SVCCHG RLS 4.5 and later).	AUX > is printed to indicate a link is established between the TTY and the auxiliary processor.
2	Input the command @MAINT, followed by a carriage return.	A menu of programs/functions is printed at the TTY.
3	Select the SVCCHG (S) or RTECHG (R) item from the menu.	
4	Using the appropriate commands listed in Table 2-A. (Table 2-B is used for SVCCHG RLS 4.5 and later, Table 2-C is used for RTECHG RLS 5.0 and later) perform the administrative changes for each customer equipped with the ACD-ADS feature. Refer to Table 2-D (Table 2-E is used for RLS 4.5 and later) for a list of valid responses/ranges.	
5	When all SVCCHG or RTECHG tasks are completed, input the command QUIT to exit the program. (To reenter the same program or invoke any other program, the @MAINT command must be input again.)	Refer to Table 2-F (use Table 2-G for SVCCHG RLS 4.5 and later) for interpretation of any error messages which are printed.
	<p>Note 1: If a new customer with the ACD-ADS feature is being added to the auxiliary processor, then the Meridian SL-1 Administration information must be used to define the new ACD-ADS customer to the Meridian SL-1. (See 553-2001-220 and -221 or 553-2YY1-310 and -311, depending on Meridian SL-1 machine type.)</p> <p>Note 2: When a single customer system is being converted to a multicustomer system, follow the steps shown in Chart 2-2.</p>	

Chart 2-2
 CONVERTING FROM A SINGLE TO A MULTICUSTOMER SYSTEM

The following assumptions are made:

- 1) The ACD system is running in the single customer mode.
 - 2) The hardware for the new customer is all setup.
 - 3) The following procedure is completed within a calendar day.
-

STEP	PROCEDURE
1	Use SVCCHG and RTECHG to define the new customer. For RLS 4.5 and later, the initial ADD command sets up default values. These must be modified to reflect the new customer's actual configuration before bringing the system up.
2	Input the command @MAINT , followed by a carriage return, to take the system down (menu is printed at TTY).
3	"RUN CLENUP " to enable the new customer.
4	Reboot the ACD system again.
5	After the system is up, the new customer shall have the supervisor display on the screens.
6	Log in the senior supervisor and define the following: <ol style="list-style-type: none"> a) interval and shift definitions. b) month and week definitions. c) answering delay spectrums. d) abandoning delay spectrums. e) threshold definitions. f) reporting control. Input the command @MAINT , followed by a carriage return, to take the system down (menu is printed at TTY). "RUN CLENUP " to enable the new customer's definitions. Use the following command to delete the current day files which do not have the above definitions in step 6. PIP C*.nnn;*/de where nnn=customer number (000 - 031)
10	Reboot the ACD system again.

Chart 2-2 Continued
CONVERTING FROM A SINGLE TO A MULTICUSTOMER SYSTEM

STEP	PROCEDURE
11	After the system is up, the following can be defined at any time. a) report definitions b) report schedule definitions c) agent definitions

```

OVL111 00 IDLE LOGI
PASS? System Password (Not Echoed)
> AX
AUX>
@ MAINT
> PLEASE ENTER A LETTER: A-ABORT, B-BACKUP, F-FLDUPD, H-HOTEST, R-RTECHG, S-SVCCHG,*- EXIT, ?- HELP
> * [S] : S
← AUX SVCCHG
LIS6

1 CUSTOMER ID = 0
2 DISPLAY OPTION = Y
3 AGENT IO OPTION = Y
4 DELAY BEFORE ANSWERING REPORT = Y
5 DELAY BEFORE ABANDONING REPORT = Y
6 AGENT PERFORMANCE REPORT = Y
7 ACO-ON PERFORMANCE (BY AGENT IO) REPORT = Y
8 SUMMARIZED ACO-ON REPORT = Y
9 ROUTE PERFORMANCE (BY ACO-DN) REPORT = Y
10 ACO-ON PERFORMANCE (BY ROUTE) REPORT = Y
11 AGENT FIRST LOGIN LAST LOGOUT REPORT = Y
12 AGENT ALL LOGIN AND LOGOUT REPORT = Y
13 AGENT DETAIL LOG REPORT = Y
14 SUPERVISOR CONTROL-POINT 1 = 3700
15 SUPERVISOR CONTROL-POINT 2 = 0
16 SUPERVISOR CONTROL-POINT 3 = 0
17 SUPERVISOR CONTROL-POINT 4 = 0
18 SUPERVISOR CONTROL-POINT 5 = 0
19 SUPERVISOR CONTROL-POINT 6 = 0
20 SUPERVISOR CONTROL-POINT 7 = 0
21 SUPERVISOR CONTROL-POINT 8 = 0
22 SUPERVISOR CRT 1 DEVICE NUMBER = 1
23 SUPERVISOR CRT 2 DEVICE NUMBER = 2
24 SUPERVISOR CRT 3 DEVICE NUMBER = 3
25 SUPERVISOR CRT 4 DEVICE NUMBER = 4
26 SUPERVISOR CRT 5 DEVICE NUMBER = 0
27 SUPERVISOR CRT 6 DEVICE NUMBER = 0
28 SUPERVISOR CRT 7 DEVICE NUMBER = 0
29 SUPERVISOR CRT 8 DEVICE NUMBER = 0
30 CUSTOMER PRINTER DEVICE NUMBER = 1
31 MAX ACO-ON/ROUTE COMBINATIONS = 75
32 MAX AC&ON/AGENT IO COMBINATIONS = 300
33 MAX CUST/ACD COMBINATIONS = 15
34 MAX CUST/ROUTE COMBINATIONS = 20
35 MAX CUST/AGENT IO COMBINATIONS = 300
36 LANGUAGE (0 OR 1) = 0
37 PASSWORD (6 CHARACTERS) = SRSPVR
38 CUSTOMER NAME (16 CHARACTERS) = ABCD LTO

AUX SVCCHG
CHG F15=3800
15 SUPERVISOR CONTROL-POINT 2 = 3800
4AUX SVCCHG
QUIT
AUX SVCCHG TERMINATED
- - % (Not Echoed)
SL1>
> LOGO
OVL111 00 IDLE

```

Note: Commands inputted by the user are underlined.

(III. 2112)

Fig. 2-1
 Example of a SVCCHG Function RLS 4.0 and earlier



```

OVL111 00 IDLE LOGI
PASS? <----System Password (Not Echoed)
> AX
AUX>
@MAINT
>;
>; ACD-D AUX Maintenance menu
>;
>; PLEASE ENTER A LETTER: A-ABORT, B-BACKUP, F-FLDUPD, I-IOTEST,
>; LF-L.FILE,R-RTECHG, S-SVCCHG, SP-SPACE,
>; *-EXIT, ?-HELP
>* {S}: S
<- AUX SVCCHG
LIS 0
GENERAL INFORMATION <GEN c#>
1 CUSTOMER NUMBER = 0
2 CUSTOMER NAME = NORTHERN TELECOM
3 SENIOR SUPERVISOR PASSWORD = NTC
4 LANGUAGE PREFERENCE = 0
5 STATUS UPDATE INTERVALS = 30
6 NO. OF DISPLAY SCREEN COLUMNS = 4
7 NO. OF SUPERVISOR CONTROL POINTS = 8
8 NO. OF VDT DEVICES ALLOCATED = 6
9 MAX. NUMBER OF VDTS IN MODE 1 = 3
10 MAX. NUMBER OF VDTS IN MODE 2 = 3
11 MAX. NUMBER OF VDTS IN MODE 3 = 3
<- AUX SVCCHG
CHG F5 10
5 STATUS UPDATE INTERVALS = 10
<- AUX SVCCHG
<CR> (NOT ECHOED)
OPTIONS <OPT c#>
1 DISPLAY ID = Y
2 AGENT ID = Y
<- AUX SVCCHG
<CR> (NOT ECHOED)
COMBINATIONS <COM c#>
1 ACD-DN / ROUTE = 150
2 ACD-DN / AGENT = 400
3 CUSTOMER / ACD-DN = 26
4 CUSTOMER / ROUTE = 100
5 CUSTOMER / AGENT = 500
<- AUX SVCCHG
<CR> (NOT ECHOED)
REPORTS <RPT c#>
1 DELAY BEFORE ANSWERING = Y
2 DELAY BEFORE ABANDONING = Y
3 AGENT PERFORMANCE = Y
4 ACD-DN (AGENT) PERFORMANCE = Y
5 SUMMARIZED PERFORMANCE = Y
6 ROUTE (ACD-DN) PERFORMANCE = Y
7 ACD-DN (ROUTE) PERFORMANCE = Y
8 AGENT FIRST LOGIN / LAST LOGOUT = Y
9 AGENT ALL LOGIN / LOGOUT = Y
10 AGENT DETAIL LOG = Y
11 AGENT DATABASE = Y
12 SMA HISTORICAL = N
13 SMA CALCULATOR = N
<- AUX SVCCHG
(III.06387)

```

→ Fig. 2-2A
 → Example of a SVCCHG Function RLS 4.5 and later

CHG F12 Y

ACCESS DENIED
 <- AUX SVCCHG <CR> (NOT ECHOED)

PRINTERS <PRT c#> CUSTOMER 0
 1 CUSTOMER PRINTER NUMBER = 1
 <- AUX SVCCHG

<CR> (NOT ECHOED)

SUPERVISOR CONTROL POINTS <SPV c#> CUSTOMER 0
 1 CONTROL POINT NUMBER = 1000
 2 CONTROL POINT NUMBER = 2000
 3 CONTROL POINT NUMBER = 3000
 4 CONTROL POINT NUMBER = 4000
 5 CONTROL POINT NUMBER = 5000
 6 CONTROL POINT NUMBER = 6000
 7 CONTROL POINT NUMBER = 7000
 8 CONTROL POINT NUMBER = 8000
 <- AUX SVCCHG

<CR> (NOT ECHOED)

VDT DEVICE NUMBERS <VDT c#> CUSTOMER 0
 1 VDT DEVICE NUMBER = 1
 2 VDT DEVICE NUMBER = 2
 3 VDT DEVICE NUMBER = 3
 4 VDT DEVICE NUMBER = 4
 5 VDT DEVICE NUMBER = 5
 6 VDT DEVICE NUMBER = 6
 <- AUX SVCCHG

SYS

SYSTEM INFORMATION <SYS>

		MAXIMUM
RELEASE NUMBER	= 4.5	
CUSTOMERS DEFINED	= 1	3
VDT DEVICES ALLOCATED	= 6	12
HIGHEST VDT DEVICE NUMBER USED	= 6	
HIGHEST PRINTER DEVICE NUMBER USED	= 1	
SUPERVISOR CONTROL POINTS ALLOCATED	= 8	32
ACD-DN/ROUTE COMBINATIONS ALLOCATED	= 150	200
ACD-DN/AGENT COMBINATIONS ALLOCATED	= 400	850
ACD-DNS ALLOCATED	= 26	30
ROUTES ALLOCATED	= 100	200
AGENTS ALLOCATED	= 500	1000

CUSTOMER ID NUMBERS IN USE : 0

ACCESS TO SYSTEM INFORMATION DENIED
 <- AUX SVCCHG

QUIT

AUX SVCCHG TERMINATED

> % (NOT ECHOED)
 SL1> LOGO
 OVL111 00 IDLE

NOTE: Commands input by the user are UNDERLINED.

(III. 06388)

Fig. 2-2B
 Example of a SVCCHG Function RLS 4.5 and later continued



PRACTICE 553 -2671-310

OVL11100 IDLE LOG I

PASS? _____ ← System Password (Not Echoed)

>AX
AUX>

@MAINT

>, PLEASE ENTER A LETTER: A-ABORT. B-BACKUP, F-FLDUPD, **I-IOTEST**, R-RTECHG. S-SVCCHG,*-EXIT, ?-HELP

>* [S] : R

← AUX RTECHG

LIS0

CUSTOMER ID = 0

<u>0</u>	ROUTE 0
<u>1</u>	ROUTE 1
<u>2</u>	ROUTE 2
<u>3</u>	ROUTE 3
<u>4</u>	ROUTE 4
<u>5</u>	ROUTE 5
<u>6</u>	ROUTE 6
<u>7</u>	ROUTE 7
<u>8</u>	ROUTE 8
<u>9</u>	ROUTE 9
<u>10</u>	ROUTE 10
<u>11</u>	ROUTE 11
<u>12</u>	ROUTE 12
<u>13</u>	ROUTE 13
<u>14</u>	ROUTE 14
<u>15</u>	ROUTE 15
<u>16</u>	ROUTE 16
<u>17</u>	ROUTE 17
<u>18</u>	ROUTE 18
<u>19</u>	ROUTE 19
<u>20</u>	ROUTE 20
<u>21</u>	ROUTE 21
<u>22</u>	ROUTE 22
<u>23</u>	ROUTE 23
<u>24</u>	ROUTE 24
<u>25</u>	ROUTE 25
<u>26</u>	ROUTE 26
<u>27</u>	ROUTE 27
<u>28</u>	ROUTE 28
<u>29</u>	ROUTE 29
<u>30</u>	ROUTE 30
	ROUTE 31
<u>3:</u>	ROUTE 32

4AUXRTECHG

CHG F30= INWATS 1

LINE 30 CHANGED TO INWATS 1

| ** CHANGE COMPLETED | *-

← AUX RTECHG

QUIT

AUX RTECHG TERMINATED

_____ ← % (Not Echoed)

SL1>

>LOGO

OVL111 00 IDLE

Note: Commands inputted by the user are underlined

(III. 2113)

Fig. 2-3
Example of a RTECHG Function (RLS 4.5 and earlier)

- AUX RTECHG
LIST 0

CUSTOMER ID = 0

RT#	RTE NAME	RT#	RTE NAME	RT#	RTE NAME	RT#	RTE NAME
0:	ROUTE 0	32:	ROUTE 32	64:	ROUTE 64	96:	ROUTE 96
1:	ROUTE 1	33:	ROUTE 33	65:	ROUTE 65	97:	ROUTE 97
2:	ROUTE 2	34:	ROUTE 34	66:	ROUTE 66	98:	ROUTE 98
3:	ROUTE 3	35:	ROUTE 35	67:	ROUTE 67	99:	ROUTE 99
4:	ROUTE 4	36:	ROUTE 36	68:	ROUTE 68	100:	ROUTE100
5:	ROUTE 5	37:	ROUTE 37	69:	ROUTE 69	101:	ROUTE101
6:	ROUTE 6	38:	ROUTE 38	70:	ROUTE 70	102:	ROUTE102
7:	ROUTE 7	39:	ROUTE 39	71:	ROUTE 71	103:	ROUTE103
8:	ROUTE 8	40:	ROUTE 40	72:	ROUTE 72	104:	ROUTE104
9:	ROUTE 9	41:	ROUTE 41	73:	ROUTE 73	105:	ROUTE105
10:	ROUTE1 0	42:	ROUTE 42	74:	ROUTE 74	106:	ROUTE106
11:	ROUTE1 1	43:	ROUTE 43	75:	ROUTE 75	107:	ROUTE107
12:	ROUTE 12	44:	ROUTE 44	76:	ROUTE 76	108:	ROUTE108
13:	ROUTE 13	45:	ROUTE 45	77:	ROUTE 77	109:	ROUTE109
14:	ROUTE 14	46:	ROUTE 46	78:	ROUTE 78	110:	ROUTE110
15:	ROUTE 15	47:	ROUTE 47	79:	ROUTE 79	111:	ROUTE 111
15:	5930196	48:	ROUTE 48	80:	ROUTE 80	111:	ROUTE112
17:	ROUTE1 7	49:	ROUTE 49	81:	ROUTE 81	113:	ROUTE113
18:	ROUTE 18	50:	ROUTE 50	82:	ROUTE 82	114:	ROUTE114
19:	ROUTE 19	51:	ROUTE 51	83:	ROUTE 83	115:	ROUTE115
20:	ROUTE 20	52:	ROUTE 52	84:	ROUTE a4	116:	ROUTE116
21:	ROUTE 21	53:	ROUTE 53	85:	ROUTE 85	117:	ROUTE117
22:	ROUTE 22	54:	ROUTE 54	94:	ROUTE 95	119:	ROUTE119
23:	ROUTE7 3	55:	ROUTE 55	87:	ROUTE a7	119:	ROUTE119
25:	ROUTE 24	56:	ROUTE 56	da:	ROUTE 88	120:	ROUTE120
	TOURS 78	57:	ROUTE 57	99:	ROUTE 99	121:	ROUTE121
26:	ROUTE 26	58:	ROUTE 58	90:	ROUTE 90	123:	ROUTE122
27:	ROUTE 27	59:	ROUTE 59	91:	ROUTE 91	123:	ROUTE123
28:	ROUTE 28	60:	ROUTE 60	92:	ROUTE 92	124:	ROUTE124
29:	ROUTE 29	61:	ROUTE 61	93:	ROUTE 93	125:	ROUTE125
30:	ROUTE3 0	62:	ROUTE 62	94:	ROUTE 94	126:	ROUTE126
31:	ROUTE 31	63:	ROUTE 63	95:	ROUTE 95	127:	ROUTE127
99:	INTERNAL						

- AUX RTECHG
CHG F16 5980196

ITEM 16 CHANGED TO 5980196

*** CHANGE COMPLETED ***

- AUX RTECHG
CHG F46 5950888

ITEM 46 CHANGED TO 5950888

*** CHANGE COMPLETED ***

- AUX RTECHG

INVALID COMMAND
- AUX RTECHG

(111. 6441)

Fig. 2-4
Example of a RTECHG Function (RLS 5.0 and later)

Table 2-A
 INPUT COMMANDS FOR SVCCHG (RLS 4.0 AND EARLIER)
 AND RTECHG

COMMAND	MEANING
S	Invokes the service change (SVCCHG) program.
R	Invokes the route change (RTECHG) program.
LIS n	Lists all the service change or route change items (Fig. 2-1, 2-2, 2-3 or Fig. 2-4) for a customer whose ID is n (O-31). If n is not specified, items for all defined ACD-ADS customers (maximum 5) are printed. If there are no ACD-ADS customers defined, then default values for the items of a customer whose ID is -1 are printed.
DEL n	Deletes the service change or route change items for a customer whose ID is n (O-31); i.e., customer n no longer has access to the ACD-ADS feature. Can be used only to completely delete a customer from the SVCCHG or RTECHG program. To delete a particular item of these programs, the CHG (change) command is used to input the value 0 (e.g., SUPV. ID, VDT. DEVICE #1).
ADD n	Adds to the auxiliary processor a customer whose ID is n (O-31). A printout of the service change or route change items (showing default values) is generated. This command must be followed by the LIS n command before changes to the service change or route change items for the customer can be made.
CHG Fxx=yy	Changes the values of the service change or route change items listed for a customer. xx denotes the service change or route change item (field) to be changed, and yy represents the new value to be assigned in that particular field. Use this command only after the service change or route change items for a customer have been printed through command LIS n. The changed field is printed with the new value shown after the command is inputted. See Table 2-D (Table 2-E for SVCCHG RLS 4.5 and later) for the permissible value/range for each field.
QUIT	Updates the changes or additions that have been made in the service change or route change program, and terminates access to the program. To reaccess either program, the @MAINT command must be inputted.

Table 2-B
 INPUT COMMANDS FOR SVCCHG (RLS 4.5 AND LATER)

COMMAND	MEANING
S	Invokes the service change (SVCCHG) program.
LIS <C#>	<p>List customer information. When the LIS command and customer number <C#> is entered, the first section of information (GEN) for the customer will be displayed. After the information has been listed, any of the displayed lines may be modified using the change command (CHG) (see below). A carriage return <CR> will display the next section of information for the customer option (OPT), this section can then be changed if desired. By entering carriage returns all seven sections can be displayed and/or modified.</p> <p>The following commands allow specific sections of customer data to be listed directly without the need to step through all previous sections with the LIS command. If carriage return is entered in response to one of the following commands the next section in the sequence will be displayed. For example, if <CR> is entered after the PRT command the Supervisor Control Point (SPV) section will be displayed.</p>
<p>Note: The section to be modified must be listed using LIS or one of the section headings before any values in that section can be changed.</p>	
GEN <C#>	List and open for modification General Information for the customer specified.
OPT <C#>	List and open for modification the Option Information for the indicated customer.
COM <C#>	List and open for modification the Combination Maximums of the indicated customer.
RPT <C#>	List and open for modification the Report option list of the indicated customer.
PRT <C#>	List and open for modification the Printer list of the indicated customer.
SPV <C#>	List and open for modification the Supervisor list of the indicated customer.
VDT <C#>	List and open for modification the VDT list of the indicated customer.
SYS	List the system information section. This section outlines the usage and maximum allowed values for all customers defined. Users are not allowed modification access to this section.
ADD <C#>	This command adds a new customer to the Service Change file. The customer number must be supplied.
DEL <C#>	This command deletes a customer from the Service Change file. The customer number must be supplied.
CHG F<#> <VALUE>	This command changes the value of the specified line number to the value supplied in the command. The only values that may be modified are those to which the user has access privileges.

Table 2-D Continued
 PARAMETERS FOR SVCCHG FIELDS (RLS 4.0 AND EARLIER)
 AND RTECHG FIELDS RLS 5.0 AND EARLIER

FIELD	VALID RESPONSE OR RANGE
30	Specify the device number for this customer's reports printer. Range is 0-5, where 0 means the customer is not equipped with a reports printer. A printer cannot be assigned to more than one customer.
31	Range is 50 to 200 Do not exceed maximums. Default is 5 when a new customer is being added (Note 4). Calculation Method: (Maximum number of routes for this customer) x (number of ACD-DNs). Example: Two ACD-DNs have five routes defined; therefore, $2 \times 5 = 10$.
32	Range is 10 to 100 (PDP-11/23) Range is 10 to 250 (PDP-11/23+, PDP11/24) Range is 10 to 500 (PDP-11/44) Do not exceed maximums. Default is 10 when a new customer is being added (Note 4). Calculation Method: (Maximum number of agent-IDs in the system) x (number of ACD-DNs). Example: Two ACD-DNs, 40 agent-IDs assigned: therefore, maximum combinations are $40 \times 2 = 80$. Note: Position-IDs can be substituted for agent-IDs.
33	Range is 2 to 30. Do not exceed maximums. Default is 2 when a new customer is being added (Note 4). Calculation Method: Add the total number of ACD-DNs for each customer equipped with ACD-ADS. Example: One ACD-ADS customer, three ACD-DNs assigned: therefore, maximum combinations = 3.
34	Range is 2 to 31. Do not exceed maximum. Default is 2 when a new customer is being added (Note 4). Calculation Method: Add the total number of routes for each customer equipped with ACD-ADS. Example: Two ACD-ADS customers, five routes assigned to customer No. 1, 10 routes assigned to customer No. 2; therefore, maximum combinations = $5 + 10 = 15$.
35	Range is 10 to 150 (PDP11/23) Range is 10 to 400 (PDP-11/23+, -11/24) Range is 10 to 850 (PDP-11/44) Do not exceed maximum. Default is 10 when a new customer is being added (Note 4). Calculation Method: Add the total number of agent-IDs assigned for each customer equipped with ACD-ADS. Example: Two ACD-ADS customers, 90 agent-IDs for customer No. 1, 45 agent-IDs for customer No. 2; therefore maximum combinations = $90 + 45 = 135$. Note: Number of agent-IDs is taken from actual number of codes assigned, not from number range assigned. Position-ID may be substituted for agent-ID.

Table 2-D Continued
 PARAMETERS FOR SVCCHG FIELDS (RLS 4.0 AND EARLIER)
 AND RTECHG FIELDS RLS 5.0 AND EARLIER

FIELD	VALID RESPONSE OR RANGE
36	Specify 0 for English, 1 for French. The language specified is the default language for the Configuration report generated when the EOD routines of the auxiliary processor are completed. (The VDT menu language is specified as part of the supervisor/senior supervisor login procedure.)
37	Specify up to six alphanumeric characters for the senior supervisor password.
38	Specify the customer name as it is to appear on all management reports (16 characters maximum).
RTECHG	
1 through 32	Specify a trunk descriptor with up to 8 alphanumeric characters. Default trunk descriptors are shown in Fig. 2-3 or Fig 2-4. See Note 3.
<p>Note 1: If the customer is to operate in the Agent-ID mode, the Agent-ID option must be enabled in the Meridian SL-1 ADS data block (overlay 23).</p>	
<p>Note 2: Ensure that the Position-ID specified corresponds to a supervisor Position-ID as defined in the Meridian SL-1 telephone data block (overlay 11).</p>	
<p>Note 3: Trunk descriptors can only be assigned to ACD trunk routes that are defined in the Meridian SL-1 route data block (overlay 16).</p>	
<p>Note 4: If sufficient resources are not available to provide a new customer with the default value, an error message (Table 2-F or Table 2-G SVCCHG RLS 4.5 and later) is printed and the customer is not added to the system.</p>	
<p>Note 5: In Meridian SL-1 Generic X11, trunk routes 0 to 127 can be assigned as ACD trunks. However, reporting on the Auxilliary Processor side will group trunks 31-127 together for data representation as trunk route 31. Route 99 is recognised by the Auxilliary Processor as the internal route, even if it is defined as a valid Meridian SL-1 route (in this case Route 99 should not be defined for ACD use). Any of trunks 32 to 127 can be used as recorded announcement trunks.</p>	
<p>Note 6: If all 8 supervisor positions are defined and you wish to change one of the positions, change the position to zero first, then change the position to the desired value.</p>	

Table 2-E
PARAMETERS FOR SVCCHG FIELDS (RLS 4.5 AND LATER)

FIELD	VALID RESPONSE OR RANGE
System Information	<p>Note 1: There is one system section for the entire ACD-D system.</p> <p>Note 2: The system section can not be modified by the customer.</p>
1	<p>Maximum Allowed VDT Devices</p> <p>This field defines the maximum number of terminals that all customers can have on the system. These terminals must be shared by all defined customers.</p> <p>The maximum number of VDT devices will vary with the PDP-system (11/23+, 11/44 or 11/73) and with the characteristics of the installation such as call traffic and system use. Refer to the Engineering Guideline for their exact values.</p>
2	<p>Maximum Allowed Customers</p> <p>This field contains the number of customers that can be defined on the PDP-system. The maximum number of customers supported is 5 on the 11/44 or 11/73 system and 1 on the 11/23+ system.</p>
3	<p>System ID</p> <p>This is the Identification Code of the ACD-D system.</p>
4	<p>Internal Route Number</p> <p>This field defines the route that the Meridian SL-1 software has currently designated as the internal route number. Two valid internal designators are 99 for Generic X11 RLS 5.21 and previous and 255 for later releases, check your current Meridian SL-1 release for the correct designator.</p>
General Information	<p>Note: This section defines general customer information that is frequently used by ACD-D software. There is a general section for each customer in the system.</p>
1	<p>Customer Number</p> <p>Specify the Meridian SL-1 customer number (O-31) for which ACD-ADS data is to be added, changed or removed.</p>
2	<p>Customer Name</p> <p>Specify a 16-character name representing the customer whose data is defined in this section. This is the name which will appear on all management reports.</p> <p>Senior Supervisor Password</p> <p>Specify the 1 to 6-character password that must be entered at login time to gain Senior Supervisor status.</p>

Table 2-E Continued
PARAMETERS FOR SVCCHG FIELDS (RLS 4.5 AND LATER)

FIELD	VALID RESPONSE OR RANGE
4	<p>Language Preference</p> <p>Specify the language (English = 0, or French = 1) that this customer's Load Configuration Report will be printed in at system startup.</p>
5	<p>Status Update Intervals</p> <p>This value defines that time in seconds between updates of the statistics portion of the Current Status Display. Specify 10, 20 or 30.</p>
6	<p>Number of Display Screen Columns</p> <p>This field gives the number of columns of agents to be displayed on the Current Status Display and Global Agent Display. Specify the number of columns (3) or (4) for a total of 39 or 52 agents on one Current Status Display screen respectively.</p> <p>Number of Supervisor Control Points</p> <p>Specify the number of supervisors that this customer can have defined (Maximum 16 PDP-11/23+, 32 PDP-11/44 or MicroPDP-11/73). ←</p> <p>Number of VDT Devices Allocated</p> <p>Specify the number of terminals allocated to this customer. The total terminals allocated to all customers cannot exceed the Maximum Allowed VDT's (Maximum 12 PDP-11/23-l-, 16 PDP-11/44, or MicroPDP-11/73) field in the System Section. ←</p> <p>Maximum Number of Devices in Mode 1</p> <p>Specify the number of terminals (1-8 PDP-11/23+ , 1-16 PDP-11/44 or MicroPDP-11/73) that can use the Current Status Display concurrently. Refer to Appendix 2 to 553-2671-151 for capacity information and associated engineering requirements. ←</p>
10	<p>Maximum Number of Devices in Mode 2</p> <p>Specify the number of terminals that can use the Report Scrolling Mode concurrently. Refer to Appendix 2 to 553-2671-151 for capacity information and associated engineering requirements.</p>
11	<p>Maximum Number of Devices in Mode 3</p> <p>Specify the number of terminals that can use the Report Definition Mode concurrently. Refer to Appendix 2 to 553-2671-151 for capacity information and associated engineering requirements.</p>
Options	Note: There is an options section for each customer in the system.

Table 2-E Continued
 PARAMETERS FOR SVCCHG FIELDS (RLS 4.5 AND LATER)

FIELD	VALID RESPONSE OR RANGE
-------	-------------------------

1

Display ID

Specify Y (yes) if a customer has access to the Current Status Display. When disabled N (no), the Display mode is disabled for this customer.

Agent ID

Specify Y (yes) if the customer's agents are identified by Agent ID or Position ID. When disabled N (no), the customer's agents are not identified by agent ID and the Agent Database will be disabled.

Note: If the customer is to operate in the AGENT-ID mode, the AGENT-ID option must be enabled in the Meridian SL-1 ADS data block (overlay 23).

Combinations

Note: There is a combinations section for each customer in the system.

1

ACD-DN/Route

This field defines this customer's portion of the system maximum number of ACD-DN/Route combinations. The maximum ACD-DN/Route combinations per system (all customers) is:



RLS 4.5 and earlier		
for	PDP-11/44	- 200
	PDP-11/23+	- 200
RLS 5.0 and later		
for	MicroPDP-11/73	- 1000
	PDP-11/44	- 1000
	PDP-11/23+	- 500



These figures are system maximums, but performance is affected by many factors and a heavily loaded system may not be able to support the full complement of **ACD-Route** combinations. See Appendix 2 to 553-2671-151 for capacity information.

Default is 0 when a new customer is being added.

Note: Calculation Method: (Maximum number of routes in an ACD-DN) X (number of ACD-DN).

Example: Two ACD-DN's have five routes each; therefore 2 x 5 = 10.

Table 2-E Continued
 PARAMETERS FOR SVCCHG FIELDS (RLS 4.5 AND LATER)

FIELD	VALID RESPONSE OR RANGE									
2	<p>ACD-DN / Agent</p> <p>This value gives this customer's portion of the number of ACD-DN/Agent combinations allowed for all customers on the system. The maximum ACD-DN/Agent combinations per system (all customers) is:</p> <table style="margin-left: 40px;"> <tr> <td style="text-align: right;">for</td> <td style="text-align: left;">MicroPDP-11/73</td> <td style="text-align: right;">= 850</td> </tr> <tr> <td></td> <td style="text-align: left;">PDP-11/44</td> <td style="text-align: right;">= 850</td> </tr> <tr> <td></td> <td style="text-align: left;">PDP-11/23+</td> <td style="text-align: right;">= 400</td> </tr> </table> <p>Default is 0 when a customer is being added.</p>	for	MicroPDP-11/73	= 850		PDP-11/44	= 850		PDP-11/23+	= 400
for	MicroPDP-11/73	= 850								
	PDP-11/44	= 850								
	PDP-11/23+	= 400								
3	<p>Customer /ACD-DN</p> <p>Specify the maximum number (0-30) of ACD-DN's allowed for this customer. There is a maximum of 30 ACD-DN's per system (all customers). Default is 0 when a customer is being added.</p>									
4	<p>Customer /Route</p> <p>Specify the maximum number of Routes allowed for this customer. For RLS 4.5 and earlier, there is a maximum of 32 Routes, including the internal route, per ACD-D system. For RLS 5.0 and later, there is a maximum of 129 routes, including the internal route, per ACD-D system. For PDP-11/23+ systems, it is recommended that the number of routes allocated for the entire system not exceed 64. See Appendix 2 to 553-2671-151 for complete capacity and performance information.</p>									
5	<p>Customer/Agent</p> <p>The value is the maximum number of Agent ID's allowed for this customer. There is a maximum of 1000 agent ID's per system (all customers).</p>									
Report	<p>Note: This section contains a set of fields (one for each report) indicating whether or not this report is available for this customer.</p> <p>There is a report section for each customer in the system. Specify Y (yes) the report is available or N (no) for each individual report (1-131).</p>									
1	Delay Before Answering									
2	Delay Before Abandoning									
3	Agent Performance									
4	ACD-DN (Agent) Performance									
5	Summarized Performance									
6	Route (ACD-DN) Performance									

Table 2-E Continued
 PARAMETERS FOR SVCCHG FIELDS (RLS 4.5 AND LATER)

FIELD	VALID RESPONSE OR RANGE
7	ACD-DN (Route) Performance
8	Agent First Login/Last Logout
9	Agent All Login/All Logout
10	Agent Detail Log
11	Agent Database
12	SMA Historical
13	SMA Calculator
Printer	
"n"	Specify the device number for this customer's reports printer. The "n" range is 0-5, where 0 means the customer is not equipped with a reports printer. A printer cannot be assigned to more than one customer.
Supervisor Control Point	There is one control point number for each supervisor defined in the general information section. There is a supervisor control point section for each customer in the system.
"n"	Control Point Number The identification code corresponding to supervisor "n". Note: If possible the identification code should correspond to a supervisor Position-ID (4 digits) as defined in the Meridian SL-1 Set data block (overlay 11). If this is not possible because the number of supervisor Position-ID's defined on the Meridian SL-1 exceeds the maximum allowed to this customer on the auxiliary processor, a dummy supervisor control point of 9999 must be created on the auxiliary processor. Any supervisor Position ID's defined to the Meridian SL-1 and not explicitly defined to the auxiliary processor will then be mapped into the 9999 supervisor control point. Failure to define the dummy supervisor control point will result in an error message from the statistics manager, and can cause corruption of current statistics data after load management commands regarding these undefined supervisors have been used. See Table 2-E SVCCHG - Fields 14 through 21, for Control Point specification examples.

Table 2-E Continued
 PARAMETERS FOR SVCCHG FIELDS (**RLS** 4.5 AND LATER)

FIELD	VALID RESPONSE OR RANGE
VDT Device Numbers	<p data-bbox="180 506 1446 590">"n" Specify one device number (1-16 for PDP-11/44 and MicroPDP-11/73) or ← (1-8 for PDP-11/23+) for each VDT allocated in the general information section. This is the logical number of terminal "n".</p> <p data-bbox="505 621 1370 646">There is a VDT device numbers section for each customer in the <i>system</i>.</p> <p data-bbox="505 678 1333 730">See Table C SVCCHG - Fields 22 through 29 for VDT specification examples.</p>

Table 2-F
 SVCCHG (RLS 4.0 AND EARLIER)
 AND RTECHG ERROR MESSAGES

MESSAGE	MEANING
ALL 8 SUPERVISOR CONTROL POINTS ALREADY IN USE	A maximum of 8 supervisor control points can be assigned.
AMOUNT AVAILABLE IS ONLY n; m IN USE BY OTHER CUSTOMERS. p RESOURCES MUST BE REALLOCATED AMONG USERS	A minimum SVCCHG value cannot be met (n < minimum). m of them are being used by other customers. p must be removed from other customers so they can be assigned to the new customer being created.
CANNOT FIND CUSTOMER	A service change or route change record for a specified customer does not exist in the service change or route change file.
CUST ID ALREADY ASSIGNED	An attempt was made to assign a customer ID which already exists in the service change file.
CUSTOMER # NOT ADDED	The customer record was not added because of problems.
DUPLICATE CUST ID	An attempt was made to create a service change or route change record for a customer already defined in the service change or route change file.
ERROR-DEFINE CUSTOMER BEFORE ATTEMPTING TO MAKE A CHANGE	An attempt was made to use the CHG command for a customer that is not defined to the auxiliary processor. (Make sure you have typed L n first, where n is the customer number.)
ID ALREADY EXISTS	An attempt was made to add a supervisor control point ID identical to one which already exists for the same customer.
INVALID BOOLEAN	The answer was not boolean (yes/no).
INVALID COMMAND	An invalid command string has been inputted.
INVALID CUSTID	Customer id either exceed the range limit or was non-numeric.
INVALID ID INPUT	The customer ID input was an invalid customer number.
INVALID LINE NUMBER	The line number (field) specified in the CHG command is not in valid range.
LUN ALREADY ASSIGNED	Another customer is already using this VDT number.
OUT-OF-RANGE	The customer ID input was not between 0 and 4, the valid range.

Table 2-F Continued
SVCCHG (RLS 4.0 AND EARLIER)
AND RTECHG ERROR MESSAGES

MESSAGE	MEANING
PRINTER DEVICE n ALREADY IN USE	This printer device (n) is already assigned to another customer.
TOO MANY CUSTOMERS	Maximum number of customers exceeded.
TOTAL FOR ALL CUSTOMERS EXCEEDED	The total of items (fields) 31 to 35 in the service change files for all defined customers exceeds the permissible maximum.

Table 2-G
SVCCHG ERROR MESSAGES (RLS 4.5 AND LATER)

MESSAGE	MEANING
DUPLICATE CUST ID	Customer ID to be added already exists.
TOO MANY CUSTOMERS	No room left in the file for new customers.
AMOUNT AVAILABLE IS:nnn AMOUNT USED BY OTHER CUSTOMERS:nnn RESOURCES MUST BE REALLOCATED AMONG CUSTOMERS TO CONTINUE	The limits of all resources is reached - to fix reduce the limits set for other customers.
CUSTOMER # NOT ADDED	The customer record was not added because of problems.
INVALID CUSTID	Customer ID either exceeded the range or was non-numeric.
DUPLICATE CUSTID	Attempt to change to an existing CUSTID denied.
SUPERVISOR CONTROL POINT ALREADY IN USE	Control point input is already assigned.
VDT DEVICE ALREADY ASSIGNED	LUN assigned for this customer has already been assigned.
PRINTER DEVICE NUMBER # ALREADY IN USE	The device number chosen is already in use by another customer.
ENTRY OUTSIDE LIMITS	The value input was not within the assigned boundaries.
NOT ACCEPTED.. PLEASE INPUT Y OR N	The answer was not boolean (yes/no).
INVALID LINE NUMBER	The line number input is not valid.
CUSTOMER NOT DEFINED	The Customer ID input was not defined in the SVCDAT.DAT file.
CANNOT FIND CUSTOMER	The customer ID not previously defined.
INVALID ID INPUT	The customer ID input was an invalid customer number.
INVALID PASSWORD . . . ACCESS DENIED	The access password input was not correct.
ACCESS DENIED	An attempt to gain access was denied as the user does not have access permission.

INTEGRATED SERVICES NETWORK

MERIDIAN SL-1 *

AUTOMATIC CALL DISTRIBUTION WITH AN
AUXILIARY DATA SYSTEM:
MAINTENANCE GUIDELINES

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Reason for Reissue: To add release 5.0 and later information.
Changes are indicated by arrows in the margin.

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1. GENERAL

1.01 The Meridian SL-1 is supplemented with a Digital Equipment Corporation (DEC) PDP-11 auxiliary processor (AUX) when the Automatic Call Distribution-Auxiliary Data System (ACD-ADS) feature is equipped. (Read 553-2671-104 for a description of the ACD-ADS feature.)

1.02 This practice provides information and procedures that may be required to isolate or repair faults that can occur on the auxiliary processor or associated peripheral equipment. Where required, reference is provided as to who to contact for technical assistance (i.e., Northern Telecom or Digital Equipment Corporation).

1.03 Information presented in this practice assumes the reader is familiar with the Meridian SE-1 ACD-ADS configuration and interface requirements as described in:

- 553-2671-151 Appendix 1 - ACD-ADS Engineering, Ordering and Configuration ↩
- 553-2671-200 - ACD-ADS Interface and Software Installation for MicroPDP-11/73
- 553-2671-200 Appendix 1 - ACD-ADS Software Updates
- 553-2671-200 Appendix 2 - ACD-ADS Interface and Software Installation for PDP-11/23, 11/23+, 11/24, and 11/44. ↩

1.04 The reader should **also** be familiar with PDP-11 disk handling and system booting procedures as described in:

- RL01/RL02 Disk Subsystem User's Guide - EK-RL012-UG-004 (ordered from DEC)
- The appropriate system manual (supplied) for the PDP-11 processor in use.

2. MAINTENANCE GUIDELINES

HIGH- AND
LOW-SPEED LINKS

2.01 The Meridian SL-1 and the PDP-11 processor are connected via a low-speed link (LSL) and high-speed link (HSL). Procedures to install and test these links are detailed in 553-2671-200. Should a problem occur on either link, use the link diagnostic overlay program (overlay 48) or the input/output diagnostic (overlay 37) to test the link (see diagnostic input/output reference manual) or the appropriate fault-clearing practice to clear faults on the associated Serial Data Interface (SDI) pack.

AUXILIARY
PROCESSOR WILL NOT
REBOOT

2.02 If, during the course of normal operation, the auxiliary processor (AUX) halts (indicated by the symbol @ printed at the Meridian SL-1 maintenance TTY) and fails to reboot, a fault has probably occurred in the operating disk (tape), disk (tape) drive(s), or Central Processing Unit+ (CPU) of the auxiliary processor.

2.03 Follow the procedures in Chart 2-1 to clear the fault or isolate the fault to a particular item. Ensure that all TTY output that is generated by the procedures is retained for use by Northern Telecom or DEC, should their assistance be required.

VDT TERMINALS DO
NOT RESPOND

2.04 Single VDT Affected. If only one VDT is not responding, substitute a known good VDT with the suspect VDT. If the known good VDT does not operate, then suspect a fault in the cables, cable connections, or modems (if any) between the VDT and the auxiliary processor. To localize the fault, access the @MAINT menu from the Meridian SL-1 maintenance TTY (Chart 2-3) and select the IOTEST function (Table 2-A). This program sends a test message over a selected TTY port at a specified baud rate. The VDT (or similar portable device) can be connected at the various cross-connect points in the cable run to the affected VDT to localize the problem to one specific area

2.05 Multiple VDTs Affected. If more than one VDT is not responding, a probable fault exists in the auxiliary processor software. Chart 2-2 gives procedures which will restore normal operation to all VDT's, and will also provide Northern Telecom with the necessary information to identify the cause of the problem. Northern Telecom, knowing the cause of the problem, can affect the necessary repairs to prevent its recurrence.

WARNING: If any RSX11M+ utility has been invoked so that it retains control (PIP>, BRO>, AT.>, etc.) it must be terminated before logging off from the processor (%). Failure to do so may result in a system failure. To terminate from any utility simultaneously press the CRTL ← and Z keys. ←

Chart 2-1
 AUXILIARY PROCESSOR WILL NOT REBOOT

Assumptions:

1. The HSL and LSL are operational.
 2. The auxiliary processor (AUX) is powered-up.
 3. The READY lamp on DLO: and DL1: are both steadily lit.
 4. The disk pack in DLO: is called the X-pack (top disk); the disk pack in DL1: is called the Y-pack (bottom disk).
-

STEP	ACTION	VERIFICATION
1	Attempt to reboot the system from the X-pack in DLO.	<p>If the AUX does not reboot, proceed to Step 2.</p> <p>If the AUX reboots successfully, there may have been an intermittent fault. Note all error messages, and resume normal operation.</p>
2	Remove and interchange the lamp covers designated OREADY and 1READY.	DLO: is now the bottom disk and DL1: is the top disk.
3	Attempt to reboot the system from the Y-pack (now in DL0:).	<p>If the AUX does not reboot, contact DEC and inform them that the AUX cannot be booted from either drive.</p> <p>If the AUX reboots successfully, abort the system and proceed to Step 4.</p>
4	Remove the Y-pack from the lower drive; set aside. Remove the X-pack from the upper drive and install it in the lower drive.	

Chart 2-1 Continued
 AUXILIARY PROCESSOR WILL NOT REBOOT

STEP	ACTION	VERIFICATION
5	Attempt to reboot the system from the X-pack in DLO: (lower drive).	<p>If the AUX reboots successfully, continue to operate in this mode. Inform DEC of a faulty upper disk drive (DLO:).</p> <p>If the AUX fails to reboot, reinstall the Y-pack in DLO: and reboot the system. Continue to operate in this mode and notify Northern Telecom of a faulty disk pack (X-pack). All data collected from midnight to the time of failure will be lost, including data for:</p> <ul style="list-style-type: none"> Agent login/logout • Agent/ ACD-DN performance Summarized ACD-DN performance Route/ACD-DN performance. <p>An attempt may be made to recover this data from the faulty DLO: using the PIP command. Even though rebooting the disk may not be possible, the data files may be copied. If this is the case, data will be lost only during the time the system is down.</p>

Chart 2-2
ALL VDT TERMINALS DO NOT RESPOND

STEP	ACTION	VERIFICATION
1	At the Meridian SL-1 maintenance TTY, input the command LOGI .	The prompt PASS? is printed.
2	Enter the system password.	The prompt > is printed.
3	Enter the command AX .	The prompt AUX > is printed.
4	Press the carriage return key.	If the prompt > appears, the operating system software is active; proceed to Step 5. If •U is the response, the system has crashed. Follow the crash dump sequence in the appropriate chart following. If the prompt > does not appear, save all TTY output and contact Northern Telecom. It may be necessary to obtain a crash dump before rebooting the system (see Charts 2-4 to 2-6). See Chart 2-1 if the AUX fails to reboot.
5	Input the command ACT .	A list of active tasks is printed at the TTY (see Fig. 2-1); proceed to Step 6. If a list of active tasks is not printed, reboot the AUX, save all TTY output and contact Northern Telecom. See Chart 2-1 if the AUX fails to boot.
6	Input the command ATL .	A list of all active ACD programs is printed (see Fig 2-2).
7	Input the command PAR .	A table of the processor's memory content is printed. (This is useful in diagnosing a 'dead' system as it shows what was in memory and presumably running before the system failed.)
8	Reboot the AUX, save all TTY output and contact Northern Telecom.	Upon completion of the reboot, all VDT terminals become operational. See Chart 2-1 if the AUX fails to reboot.

```

>ACT
-----
MCR... (TTO:)
ACTTO (TTO:)
LNKMGR (TTO:)
PFRCHK (TTO:)
CONTRL (TTO:)
AGDEF (TTO:)
-----
ASL (TTO:)
CMDPRC (TTO:)
CIDA (TTO:)
DSPLAY (TTO:)
PARMAD (TTO:)
RPTDEF (TTO:)
-----
STSMGR (TTO:)
SCROLL (TTO:)
DDFP (TTO:)
LDCNFG (TTO:)
SPOOL (TTO:)
RPTGEN (TTO:)
-----
TRKMGR (TTO:)
ACDSTS (TTO:)
LMLIB (TTO:)
>
>
>
-----
>
>
>

```

Fig. 2-1
Sample List of Active Tasks **(ACT)**

```

>ATL
...LDR 117434 IDRPAR 117734 00120000-00122400 PRI - 248. DPRI - 248.
STATUS: -CHK FXD STP -PMD PRV NSD
TI - C00: IOC - 0. EFLG - 000001 000000 PS - 170000 PC - 120354
REGS 0-6 000166 007733 177777 113476 045770 113444 120166
MCR... 114244 SYSPAR 116730 00164400-00174400 PRI - 160. DPRI - 160.
STATUS: STP -PMD PRV CLI NSD CAL
TI - TT0: IOC - 0. EFLG - 000001 040000 PS - 170000 PC - 122414
REGS 0-6 000000 120476 122016 120432 122420 000000 120366
...MCR 113444 GEN 045770 00574300-00614300 PRI - 160. DPRI - 160.
STATUS: -CHK CKD -PMD PRV MCR NSD
TI - TT0: IOC - 0. EFLG - 000001 040000 PS - 170017 PC - 120640
REGS 0-6 000000 131574 050712 000000 000000 000000 000622
LNKMGR 051624 GEN 045670 00454500-00520400 PRI - 150. DPRI - 150.
STATUS: CKD WFR -PMD
TI - TT0: IOC - 1. EFLG - 000000 140000 PS - 170000 PC - 155564
REGS 0-6 000000 000376 000002 034214 000000 034304 034176
F11ACP 116344 FCPPAR 047170 00174400-00220500 PRI - 149. DPRI - 149.
STATUS: STP ACP -PMD PRV NSD CAL
TI - C00: IOC - 0. EFLG - 000002 040001 PS - 170000 PC - 120644
REGS 0-6 116356 000000 000032 071554 045544 043336 120252
CIDA 050024 GEN 054554 00624500-00746400 PRI - 90. DPRI - 90.
STATUS: OUT CKP SEF STP WFR -PMD
TI - TT0: IOC - 0. EFLG - 000000 140000

```

Fig. 2-2
Sample List of Active ACD Programs (ATL)

Chart 2-3
ACCESSING THE @MAINT MENU

STEP	ACTION	VERIFICATION
1	At the Meridian SL-1 maintenance TTY, input the command LOGI .	The response is PASS?
2	Input the system password.	The response is >. The maintenance TTY is logged into the Meridian SL-1
3	Input the command AX to access the PDP-11 processor through the low-speed link.	The response is AUX > . A communications link is established between the TTY and processor.
4	Input the command @MAINT .	A menu of the @MAINT programs is printed at the TTY. See Table 2-A for a description of menu items.
5	Enter the desired item mnemonic from the @MAINT menu (see Note). ←	
6	When the function associated with the selected item is complete, input the command % to log the TTY out of the processor. (If access to the same or another item in the @MAINT menu is desired, the @MAINT command must be input again.)	The % command is not echoed on the TTY.
7	Input the command LOGO .	The TTY is logged out of the Meridian SL-1. The ACD Data Dump (ADD) program is loaded automatically when the LOGO command is input.

Note: If no option is selected or the **HELP (?)** option is selected, the **@MAINT** program (option *****) must be terminated before logging out of the Meridian SL-1. If **@MAINT** is not terminated, the auxiliary processor software suspends console activity until the console is released by **@MAINT**, potentially causing a failure of the ACD-ADS application software (Generic 9000). ←

Table 2-A
@MAINT MENU ITEMS

ITEM	DESCRIPTION/PURPOSE
→ ABORT (A)	This program shuts down the ACD-ADS (Generic 9000) software in an orderly sequence. This program should be used whenever it is necessary to halt the auxiliary processor, and is preferable to manually halting the processor.
→ BACKUP (B)	This program leads the user through the procedures required to create an off-line backup disk. (Part of the program includes shutting down the ACD-ADS (Generic 9000) software, if running.) An off-line backup disk should be created periodically to ensure that a customer's defined databases (report definitions, agent database, thresholds, etc.,) are not completely lost in the event that both disks (DL0: operating and DL1: on-line backup) become corrupted.
→ FLDUPD (F)	Field Update. This program shuts down the Generic 9000 software. This is the only time this program should be selected. When invoked, the program directs the user through the procedures required to replace the current Generic 9000 software release with a new release, without the loss of any existing customer statistical data.
IOTEST (I)	This program is a diagnostic tool that is useful during the physical installation and checkout of the ACID-ADS system. The program enables the user to select a TT port, and specify the speed (baud) at which the device connected to that port is to operate. The program then sends a test message to the device through the selected TT port. Caution: This program should not be used to test the HSL if it is connected to the Meridian SL-4. See Chart 2-1 in 553-2671-200 for procedures to check the HSL.
LFILE (LF)	List Files. This program prints a list of all ACD-ADS data files currently on disk. For each data file, the size (in blocks) and date of creation is printed. The data file names have the following conventions: XYyymmdd.nnn Where: X = C for current day D for daily W for weekly P for period Y = P for Agent/Position - ACD-DN R for Route/ ACD-DN S for Summarized ACD-DN yymmdd = file date nnn = Customer number (000 to 031) OTyymmdd.nnn Where: OT means that the midnight off -line tasks for customer nnn ran to completion for the date yyddmm. There are eight daily files, five weekly files and two period files per customer. Depending on the current date, it can be determined whether some data files are obsolete.
RTECHG (R)	This program invokes the auxiliary processor route change program. This administrative program is described in 553-2671-310.

Table 2-A Continued
 @MAINT MENU ITEMS

ITEM	DESCRIPTION/PURPOSE
SVCCHG (S)	This program invokes the auxiliary processor service change program. This administrative program is described in 553-2671-310.
SPACE (SP)	This program prints disk capacity information. For each disk (DL0: and DL1:), the number of available, used and free blocks is printed. There should be at least 1000 free blocks on the disk. When the disk is full, an error message will be printed by a task which requires disk space for creating files, pointing directly or indirectly to the disk space problem.
HELP (?)	This program prints a summary description of each of the @MAINT programs. Note: With the exception of the HELP (?) option, all selections automatically terminate the @MAINT menu program. If no options are selected, the @MAINT program must be terminated using the EXIT (*) option. If the program is not terminated, the TTY will remain tied to the program which will prevent error messages from the Auxiliary processor and may ultimately cause a system failure.
EXIT (*)	This command is used to exit the @MAINT mode. To reaccess any of the @MAINT programs, the command @MAINT must be input again.
CRASH DUMP SEQUENCE	<p>2.06 If the ACD-ADS (Generic 9000) software fails on the PDP-11← processor without any external indication of the cause of the failure, it may be necessary to obtain a crash dump to assist in isolating the cause. A crash dump is the contents of memory captured on the disk in drive DL1:. A special procedure (see Charts 2-4, 2-5 and 2-6 for the PDP-11/23, 11/23+, -11/24 and -11/44, respectively) is used to initiate a crash dump. The crash dump analysis sequence is provided in Chart 2-7.</p> <p>2.07 Once a crash dump has been taken, the crash dump disk should be marked with an identifying paper label affixed to the disk cover that indicates:</p> <ul style="list-style-type: none"> end user name ● date and time the crash dump was taken ● software generic and issue. <p>2.08 The crash dump disk and a copy of console output also taken at the time of the crash dump should be sent to Northern Telecom.</p>

Chart 2-4
 CRASH DUMP SEQUENCE FOR PDP-11123, -11/23+

STEP	ACTION	VERIFICATION
1	Push HALT switch (DEC's bezel) on processor down.	
2	Load a scratch disk on DL1:.	
3	Push HALT switch up. Note: DO NOT touch RESTART switch. If the RESTART switch is toggled, do not perform crash dump procedure.	@ prints out on the console.
4	Enter 40G <CR>.	System crash information prints out on the console. When the print is complete, @ prints out on the console.
5	Enter P <CR>.	The READY lights on the DL1: and DLO: blink. When the crash dump is finished, @ will be printed on the console.
6	Remove disk from DL1:, forward it to Northern Telecom.	
7	Place original DL1 disk back on the drive.	
8	Reboot the system.	

Chart 2-5
CRASH DUMP SEQUENCE FOR PDP-11124

STEP	ACTION	VERIFICATION
1	Put toggle switch to HALT position.	System responds with the following: nnnnn @ Where nnnnn is numeric output.
2	Load a scratch disk in DL:1.	Wait for the disk to become ready.
3	Put toggle switch to CONT position.	
4	Enter at console: 40G <CR>	System crash information prints out on the console. Ends with @.
5	Enter: P <CR>	Initiates crash dump activity; READY light on DL1: flashes, indicating that the contents of the system memory is being dumped to disk in DL1:. When finished, @ is printed.
6	Remove the disk from DL1: and send it to Northern Telecom.	
7	Place the original DL1 disk in the lower drive.	
8	Follow the restart procedure to reboot the system.	

Chart 2-6
CRASH DUMP SEQUENCE FOR **PDP-11/44**

STEP	ACTION	VERIFICATION
1	Put HALT switch to HALT position.	Console prints out the following CONSOLE nnnnnnn aaaaaa
2	Enter S <sp> 40 <cr>	
3	Put HALT switch back to CON position.	System crash information prints out on the console.
4	Mount a scratch disk on DL1: and press LOAD key.	READY light comes on.
5	Enter C <CR>	When the crash dump is finished, the console prints out the same message as in step 1.
6	Dismount the disk in DL1: and send it to Northern Telecom.	
7	Place the original DL1 disk in the lower drive.	
7	Follow the restart procedure to reboot the system.	

3. DIAGNOSTIC FEATURES

- Error Messages** 3.01 CONTRL is responsible for printing all task error messages on the Meridian SL-1 Maintenance TTY. The messages are either non-fatal or fatal. If a fatal error is received then the sending task will be aborted after the message has been printed. The task abort will cause the appropriate exit action to be taken by CONTRL. See Control Messages and ACD-ADS (Generic 9000) software Messages for a detailed explanation of error messages.
- Task Header Display** 3.02 Whenever CONTRL finds a task to be unresponsive to the periodic polling, CONTRL will spawn MCR with a command line of the form:
- `'RMD T,T=ttttt'`
- where the name of the unresponsive task (see Table 5-C for task names) will replace the 'ttttt' field. This causes the task's header to be displayed on the console, including all currently open files. See Fig 3-1 for a sample printout of RMD.
- Additional Diagnostic Hook** 3.03 In addition to the RMD command which will be run as described above, CONTRL will also run the following command at this time:
- `'xxx ttttt'`
- where the unresponsive task's name will replace the 'ttttt' field. This command provides the facility to run an additional diagnostic tool whose definition can be defined at a later date, and which may change with time.

➤ RMD T, T=LNMGR

RSX-11M-PLUS v2.1 BL15B 512K

11-DEC-8508:53:1 5

Task: LNMGR Partition: GEN status: WFR --FMDMCK
Own: TIO: I/O: 0. Dpri: 150. Pri: 150. Spri: 154. Len:00102200 -

R0 = 013572 R1 = 100074 R2 = 000000 R3 = 000030 R4 = 000000 R5 = 000012
PC = 002262 --FS = 170000 SF = 077766 \$DSW = 1. Eflg = 000000 100000

LUN File

LUN File

- 1. SY0:
- 2. SY0:
- 3. SY0:
- 4. SY0:
- 5. TIO:
- 5. TIO:
- 7. TT1:
- 8. TT1:
- 9. None
- 10. None

➤

1.

➤
➤

➤ (Ill. 6454)

Fig. 3-1
Sample Printout Of Diagnostic Tool RMD

4. CONTROL MESSAGES

Message Types	<p>4.01 The Control (CONTRL) task prints various messages on the Meridian SE-1 maintenance TTY. Messages are of the following types:</p> <ul style="list-style-type: none"> ● information only ● error conditions detected ● on behalf of other system tasks
Message Format	<p>4.02 Each message is preceded by a time stamp on the first line of the message. The time stamp prints the time of the message output in the form "hh:mm".</p> <p>4.03 "tttt" in the message field identifies the six character name. Table 5-C lists specific task names. Following are the CONTRL messages in upper case with the associated message explanation in lower case:</p> <p>Meridian SL-1 DATE MAY BE WRONG TYPE YES <CR> TO ACCEPT NO <CR> TO HALT:</p> <p>The date on the Meridian SL-1 is not what CONTRL is expecting. This message asks the operator to confirm that the Meridian SL-1 date is correct.</p> <p>Meridian SL-1 DATE INCORRECT: PLEASE SET Meridian SL-1 TIME THEN RESTART AUX</p> <p>The date on the Meridian SL-1 was not confirmed by the previous message. The system will shut down to allow the operator to set the Meridian SL-1 time.</p> <p>Meridian SL-1 TIME NOT SET: PLEASE SET Meridian SL-1 TIME THEN RESTART AUX</p> <p>The time on the Meridian SL-1 has not been set. This is vital to the ACD operation, so the system will be halted while the operator sets the time on the Meridian SL-1.</p> <p>AUX INITIALIZATION STARTED*</p> <p>This message signals the start of AUX initialization. All the online tasks will be started by the CONTRL task.</p> <p>AUX INITIALIZATION COMPLETED; DID YOU %LOGO?</p> <p>AUX initialization has been completed. This message also reminds the operator to log off of the Meridian SL-1 maintenance TTY since the Meridian SL-1 will be unable to supply the initialization data to the AUX otherwise.</p> <p>AUX INITIALIZATION FAILED: SYSTEM WILL SHUTDOWN</p>

During system initialization, CGNTRL encountered a problem that would not allow the proper operation of the system. Another message will have been printed which describes the exact problem.

TASK tttttt ACTIVATED

The task named in the message has been activated and CQNTRL is now waiting for the task to respond when it has completed its initialization. This message appears during the initialization of the AUX system.

TASK tttttt SPAWN FAILED, STATUS = nnn

CGNTRL attempted to spawn the named task but the spawn failed. nnn is the status returned from the spawn directive. The most likely status codes are:

- 1 = Insufficient dynamic memory.
- 2 = Task is not installed.

4.04 TASK tttttt INITIALIZED

This message is printed when the named task has successfully completed its initialization and has responded to the CQNTRL task.

TASK tttttt INITIALIZATION FAILED

The named task was successfully spawned but the task did not respond to CONTRL after it completed initialization. CONTRL will try 3 times to bring the task up. If it still cannot initialize the task then the system will be halted.

** AUX GOING OFF-LINE **

This message indicates that the AUX is shutting down to go offline. This message should only appear at midnight or when the AUX is started and the previous day's off-line tasks have not been run.

TASK tttttt DIED*

This message indicates that the named task has died successfully as part of the AUX shutdown procedure.

FATAL ERROR: CONTRL COULD NOT OPEN SVCDAT.DAT

During system startup, CQNTRL attempts to write the current date into the SVCDAT.DAT file. When it attempted to do this the operation failed. The system will be halted.

LOW POOL EVENT, tttttt WILL BE DISABLED

This message warns that available pool space is very low and that the named task is being disabled in an attempt to free up enough pool space so that AUX is able to continue running in a slightly degraded mode.

HIGH POOL EVENT, tttttt WILL BE RE-ENABLED

This message indicates that available pool space has recovered to the point that CONTRL has decided that the named task can be re-enabled.

OUT OF MEMORY... SYSTEM WILL BE RESTARTED

The system's available pool space is too low for AUX to continue running. If this occurs then there is most likely something wrong with the software, so CONTRL restarts the system in an attempt to correct the problem.

TASK ttttt DOES NOT RESPOND, TASK WILL BE ABORTED

The named task did not respond to the POLL message sent by CONTRL. CONTRL will run RMD to show that task's header and then abort the task. The task abort will be handled as an abnormal task exit, thus reviving the task or causing a system restart.

TASK ttttt REVIVED

The named task has been restarted by CONTRL. This may be due to any of the following reasons:

- | the task aborted due to a **runtime** error,
- | CONTRL aborted the task because it failed to respond to the poll,
- | or a high pool event has re-enabled the task.

TASK ttttt REVIVE FAILED, STATUS = nnn

CONTRL attempted to revive the named task but was unable to spawn the task. nnn is the status of the spawn directive and will most likely be one of the following 2 values:

−1 = Insufficient dynamic memory.

−2 = Task is not installed. ,lsk.OFFLINE TASK ttttt STARTED*

The named offline task has been started.

OFFLINE TASK ttttt SPAWN FAILED, STATUS = nnn

The named offline task could not be spawned by CONTRL.

OFFLINE TASK ttttt COMPLETED*

The named offline task has completed successfully.

TASK ttttt ABORTED

CONTRL has aborted the named task. It may have been aborted for any number of reasons including:

- The task did not respond to the polling
 - | The task has been aborted to free pool space.
- The task did not respond to **CONTRL's** request to die.
 - | The task was aborted to ensure that it really was inactive.

TASK ttttt ABORT FAILED, STATUS = nnn

When attempting the abort the named task, CONTRL encountered an error as given by the 'nnn' field in the message.

ttttt RECVD BAD MSG FROM ttttt, MESTYPE = nnn

<word3> <word4> <word11> <word12>

A task received a message from another task which could not be processed. The message has been discarded. Words 3 to 12 give the first 10 words of the message buffer in octal format.

AUX ERROR: <word1>, <word2> IN: ttttt

<word3> <word4> <word11> <word12>

A task encountered an I/O or task-specific error. See Tables 5-A to 5-T for details.

AUX WILL BE RESTARTED

CONTRL is exiting to perform a total system restart. Just before exiting, CONTRL restarts LNKMGR to bring the system back up automatically.

AUX IS HALTING . . .

CONTRL is exiting due to an error requiring operator intervention. A previous message should appear on the console to indicate what is wrong. Once the problem has been corrected, the system should be restarted manually.

5. ACD-ADS (GENERIC 9000) SOFTWARE ERROR MESSAGES

5.01 When an error condition is detected by any auxiliary processor task, the CONTRL task is informed by a message and prints the error information at the Meridian SL-1 maintenance TTY.

5.02 Auxiliary processor error messages that are printed at the TTY fall into two categories:

- (1) Messages that identify errors encountered by the DEC PDP-11 **RSX-11M/M+** operating system software.
- (2) Messages that identify errors encountered by the Northern Telecom ACD-ADS (Generic 90001 software). ←

5.03 This practice identifies and interprets error messages that can be printed by the ACD-ADS (Generic **9000**) software. Refer to DEC ← documentation (listed in 553-2671-151 Appendix 1) for identification and interpretation of **RSX-11M/M+** operating system error messages.

SEVERITY CODES

5.04 A severity code number (1–8) is associated with each error message identified in this practice. Refer to Table 5-A for the meaning/action of each severity code.

Table 5-A
ERROR MESSAGE SEVERITY CODES

SEVERITY CODE	MEANING/ACTION
1	Not a serious message unless it persists. If it does persist, contact Northern Telecom for assistance.
2	Not a serious message unless it persists. The message can relate to a hardware fault. Check peripheral equipment (VDT, printers) and associated TT port on the processor, then contact Northern Telecom for assistance.
3	This can be serious. If the message occurs together with disk hardware faults, call DEC to inspect the disk drives. Otherwise, contact Northern Telecom.
4	This can be a serious software error. Contact Northern Telecom for assistance.
5	The customer service change options could possibly be wrong or too small. Check the customer service change options (553-2671-310) and contact Northern Telecom if the problem is not discovered.
6	Reboot the processor. If the problem continues, contact Northern Telecom for assistance.
7	A basic system size/capacity restriction has been exceeded. Contact Northern Telecom for assistance.
8	A problem exists on the high-speed link. Check the HSL cable and use overlay 48 to determine the status of the link.

→MESSAGE OUTPUT
FORMAT

5.05 ACD-ADS (Generic 9000) software error messages are output at the Meridian SL-1 maintenance TTY in the following format:

```
hh:mm AUX-ERROR:<WORD 1>;<WORD 2>IN: <TASKNAME>
<WORD 3>, <WORD 4>, . . . <WORD 12>
```

5.06 The first line of output consists of the time (hh:mm) at which the error occurred and the name of the task which detected the error. The first and second line (see Note) of output contain diagnostic information which is presented in octal format. The information consists of 12 sets of octal numbers referred to as <WORD 1>, <WORD 2>, etc., up to <WORD 12>.

Note: The second line of the error message extends to a third line if an 80-column printer is used.

BAD MESSAGE
ERRORS

5.07 Bad message errors can occur in any task and are assigned a Severity Code of 1. Bad message errors are output in the following format.

```
hh:mm <TASKNAME> RCVD BAD MSG FROM <TASKNAME>
MESTYPE = nn
```

CLASSES OF ERROR
MESSAGES

5.08 There are two classes of error messages: common routine error messages and task-specific error messages. Task-specific error messages are identified by <WORD 1> being printed as -1. Common-routine error messages are identified by <WORD 1> being printed as other than -1. (Refer to Chart 5-1 for an example of conversion of an octal number to a decimal number.)

COMMON-ROUTINE
ERRORS

5.09 Common-routine errors can occur in any task and are identified by <WORD 1> having a value greater than -1. All words in a common-routine error message are printed in octal. <WORD 1> is actually a combination of an Error Type Code and a Common-Routine Identifier. The high byte of <WORD 1> contains the Error Type Code, the low byte contains the Common-Routine Identifier.

5.10 Possible Common-Routine Identifiers (in octal) are:

1	(N)BLDOPN	- Opens a file by file name
2	CLREFN	- Clears an event flag
3	CRMKTM	- Cancels a mark time request
4	DELCLS	- Deletes an unopened file
5	DELOPN	- Deletes a currently open file
6	DISC10	- Disk I/O operation
7	DEQMSG	- Dequeues next message
10	DEQFRM	- Dequeues next message from a specific task
11	EFWAIT	- Waits for logical OR of event flags
13	GETINI	- Gets the INICMD message from CONTRL
14	MRKTM	- Specifies a mark time
15	OPNFID	- Opens a file by file ID
20	RENAME	- Renames a file
22	SETEFN	- Sets an event flag
23	SNDMSG	- Sends a message
24	TERM10	- Terminal I/O operation.

5.11 Possible Error Type Codes (in octal) are:

0	Bad parameter passed
1	Terminal read failure
2	Terminal write failure
3	File open error
4	File read failure
5	File write failure
6	File delete failure
7	File rename failure
10	MCR directive error.

5.12 To separate the Error Type Code from the Common-Routine ID, perform the following steps:

- (1) Divide <WORD 1> by 400 (octal).
- (2) The result of the integer division is the Error Type Code.
- (3) Multiply the Error Type Code by 400 (octal) and subtract this number from <WORD 1>.
- (4) The result is the Common-Routine ID.

5.13 In common routine errors, <WORD 2> to <WORD 12> usually contain additional information relating to the error. This information differs from one Error Type to the next. The next sub-sections describe what information will be contained by the message.

BAD PARAMETER ERRORS

5.14 **Bad** parameter errors will be structured so that <WORD 2> ← specifies the parameter number, <WORD 3> will display the parameter value.

FILE RELATED ERRORS 5.15 File-related errors cause the RSX-11 I/O error code to be displayed in <WORD 2>, <WORD 3> to <WORD 10> contain the file descriptor. (Software maintenance personnel should refer to Appendix 1 of the IAS/RSX-11M/M-PLUS Executive Reference Manual.)

MCR DIRECTIVE ERRORS 5.16 <WORD 2> will contain the RSX-11 error code. (Software maintenance personnel should refer to Appendix B of the RSX-11M/M-PLUS Executive Reference Manual.) <WORD 3> will contain the directive address.

5.17 Table 5-C lists the TASKNAME and corresponding error code table for each type of task-specific error code. The task-specific error code that corresponds to <WORD 2> is listed in the referenced table.

Table 5-C
REFERENCE TABLE FOR **TASKNAME** ERROR MESSAGES

TASKNAME	ERROR MESSAGE TABLE
LNKMGR	Table 5-D
CIDA	Table 5-E
RPTDEF	Table 5-F
DDPP	Table 5-G
STSMGR	Table 5-H
DSPLAY	Table 5-I
SCROLL	Table 5-J
SPOOL	Table 5-K
PARMAD	Table 5-L
AGDEF	Table 5-M
ASL	Table 5-N
ACDSTS	Table 5-O
LDCNFG	Table 5-P
TRKMGR	Table 5-Q
CMDPRC	Table 5-R
RPTGEN	Table 5-S
OFFLINE	Table 5-T
RPTGNZ	Table 5-U

Table 5-D
LNKMGR ERROR MESSAGES

LNKMGR (link manager) handles messages that are transmitted over the link between the Meridian SL-1 and the PDP-11.

<WORD 2>	INTERPRETATION	SEVERITY CODE	
1	Input timeout, no message received from the Meridian SL-1.	8	
2	Output buffer overflow.	1	
3	Output timeout; no ACK/NAK received from the Meridian SL-1 in response to a message sent from the auxiliary processor.	8	
6	SOH not the first character received from the link.	8	
7	The count of NAK's from the Meridian SL-1 to an output message from the auxiliary processor exceeded threshold. Output message is thrown away.	8	
8	An error was encountered while reading the service change file.	1	←
9	Number of characters received from the Meridian SL-1 is greater than 128.	8	←
10	Invalid customer ID received from the Meridian SL-1. The invalid customer ID will appear in <WORD 3>	1	← ←
11	Error status received when a read I/O issued to the link.	8	
13	Checksum on an Meridian SL-1 message is not correct.	8	←
14	Timeout waiting for PFRCHK to respond (30 seconds).	8	
15	An invalid message type was received from the Meridian SL-1. WORD3 contains the bad message type, WORD4 is the message number within the packet received.	2	
16	Too few characters received from Meridian SL-1 for message type. The message type will appear in WORD3, the message number within the packet appears in WORD4.	2	
17	Free pool space has become critical. LNKMGR has increased its throttling level. This means that more messages will be discarded. WORD3 will contain the new throttling level. Throttling level 2 is a partial throttle in which call processing messages are no longer sent to the real-time statistics processor (STSMGR). Throttling level 3 is more severe; position state change messages are no longer sent to STSMGR. Throttling level 4 is full throttle: all messages are discarded by LNKMGR at this level.	7	←

Table 5-D Continued
LNKMGR ERROR MESSAGES

<WORD 2>	INTERPRETATION	SEVERITY CODE
→ 18	Free pool space has increased. LNKMGR has decreased its throttling level. This means that fewer messages will now be discarded by LNKMGR. WORD3 will contain the new throttling level. If the throttling level has returned to normal (i.e. WORD 3 is 1) then WORD4 to WORD8 will contain the number of calls the Meridian SL-1 accepted during throttling for customers 1 to 5.	*
20	LNKMGR has started the Meridian SL-1 link data tracing	*
21	LNKMGR has stopped the Meridian SL-1 link data tracing.	*
22	The operator has attempted to redefine the trace parameters while a trace was in progress.	*
23	Error writing to trace file SLTRACE.DAT on disk DL1.	3
	* information only	

Table 5-E
CIDA ERROR MESSAGES

CIDA is the Current Interval Data Administrator. It collects call information for reporting purposes

<WORD 2>	INTERPRETATION ■ RLS 4.5 and Earlier	SEVERITY CODE ←
1	GETTINI returned a false.	4
2	EOF detected on reading service change file.	3
3	EOF detected on reading report parameter file.	3
4	Incorrect flag number returned from EFWAIT.	4
10	Number of positions exceeded the maximum.	5
11	DMPINI message is invalid.	4
25	Threshold value for received ACD-DN not specified in report parameter file.	5
101	No more free data records.	5
102	Record count exceeded limit specified in service change file.	5
103	Error creating SMDMP file.	3
104	Error creating AGDMP or RTDMP file.	3
<WORD 2>	INTERPRETATION ■ RLS 5.0 and Later	SEVERITY CODE ↗
1	INICMD message was not received from CONTRL .	4
2	Unexpected EOF detected while reading SVCDATxx.DAT.	3
3	Unexpected EOF detected while reading RPTPRM.DAT.	3
4	Incorrect event flag number returned from EFWAIT.	4
5	Previous dump has not been processed by DDPP by the time the next dump is performed by CIDA.	4
10	Number of positions exceeded the maximum.	5
11	Number of ACD-DN's for a customer exceed the maximum defined for this customer in Service Change.	5
12	Number of Agents for a customer exceed the maximum defined for this customer in Service Change.	5 ↘

Table 5-E Continued
CIDA ERROR MESSAGES

↖ <WORD 2>	INTERPRETATION ▪ RLS 5.0 and Later	SEVERITY CODE
13	Number of Routes for a customer exceed the maximum defined for this customer in Service Change.	5
4	25 Threshold value for the destination ACD-DN is not defined in RPTPRM.DAT.	5

Table 5-F
RPTDEF ERROR MESSAGES

RPTDEF (report definition) handles report definition from supervisor VDT.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	Cannot open service change file.	3
2	Cannot read service change file.	3
3	Cannot open or read report parameter file.	3
4	Cannot find file containing report and schedule definitions. New (empty) file created.	1
5	Cannot read language file. (error 5 is not applicable to RLS 4.5 and later.)	3
7	I/O completion event flag set when VDT in IDLE state.	2
9	Cannot create file to contain report and schedule definitions.	3
10	Cannot create or access file to contain schedule of reports to print later in the day. Ignore this message unless it occurs twice.	3
11	Cannot open language file.	3
15	Cannot attach the terminal	2
17	Too many reports scheduled to print later in the day (maximum 600)	7

Table 5-G
DDPP ERROR MESSAGES

DDPP is the disk data preprocessor. It formats current interval reporting data into the Current Day File on disk.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	GETINI routine returned a false.	4
2	EOF detected while reading service change file.	3
3	EOF detected while reading report parameters file.	3
4	Cannot find matching ACD in agent/route records.	4
5	Cannot find matching ACD in summary records	4
101	EOF detected while reading header of current interval file.	3
102	EOF detected while reading customer index area of current interval file.	3
103	EOF detected while reading ACD index area.	3
104	EOF detected while reading daily file.	3
105	Unable to write header record of current day file.	3
106	Unable to write interval data of current day file.	3
107	Unable to write ACD index area of current day file.	3
108	Unable to write data area of current day file.	3
	The following messages apply only to RLS 5.0 and later	
↗	201 10 minute timeout waiting for RPTGEN to finish using current day file after adhoc dump request. DDPP will continue regular processing regardless.	1
4	202 DIE message receives while waiting for RPTGEN to finish using current day file after adhoc dump request. DDPP will continue with the shutdown sequence .	1

Table 5-H
STSMGR ERROR MESSAGES

STSMGR (statistics manager) processes call messages for later display on supervisor's statistics display.

<WORD 2>	INTERPRETATION	SEVERITY CODE	
0	ACD-DN link list overflow. Too many ACD-DN's have been defined in Meridian SL-1 - exceeding the limit imposed on the auxiliary processor.	5	
	Agent position link list overflow. Too many agent positions have been defined in Meridian SL-1 - exceeding the limit imposed on the auxiliary processor.	5	
10	Duplicate agent position defined. This message is always printed when one attempts to add control point supervisor ID that already exists.	1	
11	Attempt to delete a nonexistent ACD-DN after receiving a delete ACD-DN (DPDMSG) message from the Meridian SL-1.	1	
12	Attempt to delete a nonexistent agent position after receiving a delete agent position message from the Meridian SL-1.	1	
13	Attempt to delete an ACD-DN with at least one agent position assigned.		
14	Attempt to delete a control point supervisor ID. Action ignored. The deletion (or addition) of control point supervisor ID can be achieved only by an off -line auxiliary processor service change (553-2671-310) followed by a reinitialization of the auxiliary processor.	1	
20	Timestamp in call messages from Meridian SL-1 might have the incorrect time, possibly caused by lack of time synchronization between the auxiliary processor and the Meridian SL-1. (error 20 is not applicable to RLS 4.5 and later.)	1	a-
50	Invalid parameter in message received. Bad ACD-DN.	1	
51	ACD-DN does not exist.	1	
52	ACD-DN does not exist, cannot be deleted.	1	
53	Report parameter file corrupted.	3	
54	Attempt to reassign a position to a nondefined ACD-DN.	5	
55	Supervisor not defined in service change file. (applicable to RLS 4.5 and later).	5	

Table 5-1
 DSPLAY ERROR MESSAGES

DSPLAY (display) handles the supervisor display operation of the VDT.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	<p>Data access denied by STSMGR for display. This is caused by either of the following</p> <p>(a) Invalid control point supervisor ID.</p> <p>(b) STSMGR is currently busy. Try to request display again.</p> <p>I/O failure. This is usually a problem caused by terminal hardware problem during input or output.</p> <p>Attempt to delete an ACD-DN which does not exist.</p> <p style="margin-left: 40px;">Note: The preceeding DSPLAY error numbers apply to RLS 4.0 and earlier. The following DSPLAY error numbers aply to RLS 4.5 and later.</p>	1
1	Terminal I/O failed.	2
2	Terminal read error.	2
3	Terminal write error.	2
4	Terminal status error.	2
5	Queue overwritten error.	1
6	File open error.	3

Table 5-J
SCROLL ERROR MESSAGES

SCROLL handles the scrolling of management reports on the VDT.

<WORD 2>	INTERPRETATION	SEVERITY CODE	
1001	Invalid VDT number in SCROLL message.	1	↖
1002	Customer ID in message does not match VDT in message.	5	
1003	Could not attach to terminal.	2	
1005	Processing read when VDT I/O status is idle.	2	
1006	Could not open a data file during initialization.	3	
1007	EOF occurred when SVC DAT.DAT read attempted.	3	
1008	GETINI failed.	4	↙

Table 5-K
SPOOL ERROR MESSAGES

SPOOL prints reports at the reports printer.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1601	GETINI failed.	4
1602	EOF occurred when SVCDAT.DAT read attempted.	3
1603	Cannot open SVCDAT.DAT.	3
1604	Customer ID in spool message not found in records.	5
1605	Cannot attach to printer.	2
1606	An error was detected trying to write a file to the printer. The file probably had an invalid character which is causing the problem, and should be deleted.	2

Table 5-L
PARMAD ERROR MESSAGES

PARMAD is the Parameter Administration task. It allows the senior supervisor to change the period/week definitions, interval/shift definitions, answering/abandoning delay spectra, and to enable/disable items for reporting.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1401	PARMAD message contained an invalid terminal ID.	4
1402	Could not attach to terminal.	2
1403	Could not create a new RPTPRM file.	3
1404	Terminal I/O error.	2
1405	Could not open a LANGUAGE file.	3
1406	Terminal read error.	2
1407	Failed to open a data file.	3
1408	EOF reached on reading SVCDAT file.	3
1409	GETINI failed.	4

Table 5-M
AGDEF ERROR MESSAGES

AGDEF handles agent definition by the senior supervisor.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1301	EOF reached when reading SVCDAT file (SVCDAT.DAT is bad).	3
1302	GETINI failed.	4
1303	AGENTS file initialization failed.	3
1304	AGENTS file could not be opened.	3
1305	Bad index record in AGENTS file.	4
1306	Could not find LANGUAGE1 file (English).	3
1307	Could not find LANGUAGE2 file (French).	3
1308	Too many agents (in total, for all customers).	7
1309	Could not attach to terminal (VDT), after initial request of supervisor.	2
1310	Terminal I/O error.	2
1311	Terminal read failed.	2
1312	RPTGEN did not set global event flag that indicates AGENTS file is free within the expected time: either RPTGEN failed, or the system is very heavily loaded.	1
1313	Re-opening of AGENTS (after RPTGEN) file failed.	3
1314	Write I/O error when writing to AGENTS file	3

Table 5-N
ASL ERROR MESSAGES

ASL handles Agent Status Logging for use in the Agent Log reports.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	Initialization message not received from CONTRL.	4
2	End-of-file on service change file. (SVCDAT.DAT is bad).	3
3	Error when creating, opening, or when reading or writing header record of "manned" file. (MAN.DAT)	3
4	Error when creating, opening, or when reading or writing header record of work file. (T) WRK.DAT.	3
5	Error when creating, opening, or when reading or writing header record of "walkaway" file. (WLK.DAT)	3
7	Error creating WRK.DAT file for RPTGEN.	3
8	Error when writing a file using record access mode.	3
9	Error when extending a file using record access mode.	3
10	Number of agents exceeded maximum. (1000 total for all customers in one day).	7
11	Infinite loop in position state automaton (program bug; should never occur).	4
12	Error while copying one file to another (will be while copying TWRK.DAT to WRK.DAT for RPTGEN).	3

Table 5-O
ACDSTS ERROR MESSAGES

ACDSTS generates a snapshot report of the supervisor display.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1101	Attempt to open DAT file failed.	3
1102	Wait for STSMGR dump complete message timed out.	1
1103	Bad message received from STSMGR.	4
1104	STSMGR dump of ACDTAB failed.	4
1105	Attempt to open dump file failed.	3
1106	ACDSTS initialization failed.	4
1107	Unexpected dump complete message from STSMGR.	4

Table 5-P
LDCNFG ERROR MESSAGES

LDCNFG generates a Load Configuration report.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1201	An unexpected EOF occurred on LDCNFG.DAT or the LANGUAGE files. Delete all LDCNFG.DAT and retry.	1
1202	The common routine CLREFN failed. Retry.	4
1203	TRKTAB.DAT was found to be inconsistent. A level one entry without any level two records attached was found. Rebooting will restore the file.	6
1204	The file LDCNFG.DAT as copied by STSMGR is inconsistent.	1
1205	An unexpected EOF occurred on SPLDMD.	3
1206	An unexpected EOF occurred on TRKTAB.DAT. Deleting TRKTAB.DAT and then rebooting would restore the file.	6
1207	STSMGR's attempt to copy AACDTB to LDCNFG.DAT failed - probably due to an I/O error. Delete all LDCNFG.DAT and retry.	1

Table 5-Q
TRKMGR ERROR MESSAGES

TRKMGR controls trunk information for printing in the Load Configuration report.

<WORD 2>	INTERPRETATION	SEVERITY CODE
620	The incoming message from the Meridian SL-1, via LNKMGR, contained faulty data.	8
630	Trunk not found in TRKTAB.DAT.	4
640	TRKTAB.DAT is full. A trunk record is discarded. Rebooting will reinitialize TRKTAB.DAT and recover space from out-of-date records.	6
650	Attempt to add a trunk record which already exists.	1
670	A redundant ITDMSG is received from the Meridian SL-1.	1
680	The common routines SETEFN/CLREFN returned a failed code.	4

Table 5-R
CMDPRC ERROR MESSAGES

CMDPRC is the command processor. It handles supervisor **login** and mode selection at the VDT.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1501	Terminal not ready.	2
1502	Read error.	2
1503	Write error.	2
1504	Detach error.	2
1505	Attach error.	2
1506	Input on deactivated terminal.	2

Note: All error conditions, except bad message, result in a message with the following format: <WORD 2> is the error code, <WORD 3> is the terminal ID, and <WORD 4> is the RSX-11 error status. In the case of bad **CCMSG's**, <WORD 1> gives the terminal ID read from those messages: the remaining words are exactly as received by CMDPRC.

Table 5-S
RPTGEN ERROR MESSAGES

RPTGEN is the report generator task.

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	Initialization message not received.	4
2	Invalid output device specified in the message received from RPTDEF.	4
3	Invalid customer ID received.	4
4	Invalid report ID received.	4

Note: The **preceding** RPTGEN error numbers apply to RLS 4.0 and earlier.
The following RPTGEN error numbers apply to RLS 4.5 and later.

Table 5-S Continued
RPTGEN ERROR MESSAGES

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	No INICMD message received from CONTRL. Task initialization has failed.	4
2	One or both of the language files could not be opened. Task initialization has failed.	3
3	An invalid customer ID was found in a report specification <WORD 3> will contain the invalid customer ID.	4
4	Invalid report specification received.	4
5	An agent report was specified for a customer with the Agent ID option disabled.	5
6	Unable to read the RPTPRM.DAT file.	3
7	Timeout waiting for a dump from AGDEF, ASL or CIDA. <WORD 3> will contain the Report Type <WORD 4> will contain the Report Level <WORD 5> will contain the Report Time Frame. The report will not be generated.	1
8	RPTGN2 dies, or encountered an error while generating a report. <WORD 3> will contain the Report Type <WORD 4> will contain the Report Level <WORD 5> will contain the Report Time Frame.	4
	The following message applies only to RLS 5.0 and later.	
↗	9 An invalid shift specification was received i.e., shift 5 was specified when only 3 shifts are defined. <WORD 3> will contain the Customer ID <WORD 4> will contain the Report Type <WORD 5> will contain the invalid shift specification	
4	* Information only.	

Table 5-T
OFFLINE ERROR MESSAGES

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	Initialization message not received (DSGEN, DRGEN, DPGEN, WSGEN, WRGEN, WPGEN, PRGEN, PPGEN, PSGEN tasks).	1
4	Bad daily file (PRGEN, PSGEN, PPGEN tasks).	3
8	A file does not exist on DLO: (CLEANUP).	3

Table 5-U
RPTGN2 ERROR CODES FOR RLS 4.5 AND LATER

<WORD 2>	INTERPRETATION	SEVERITY CODE
1	Initialization message is not received.	4
2	Invalid output device specified in the message received from RPTDER.	4
3	Invalid customer ID received.	4
4	Invalid report ID received.	4
5	Invalid supervisor ID received will not be generated.	4
6	EOF encountered while reading the RPTPRM.DAT file.	3

SL-1”
BUSINESS COMMUNICATIONS SYSTEM
CENTRALIZED ATTENDANT SERVICE (CAS)
FEATURE DESCRIPTION AND ENGINEERING

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3. FEATURE DESCRIPTION	2	
A. Controlling CAS Service.....	2	
B. Listed Directory Number Calls	2	1.02 Testing and feature operation can be found in 553-2681-300.
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F. Dial “0” Calls	4	
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H. Calls to Tie Trunks	4	
I. Access to Special Services (WATS, FX, etc.)	5	2. GENERAL DESCRIPTION
J. Night Service	5	
K. Trunk Group Busy	5	
L. Flexible Attendant Directory Number	5	2.01 A typical CAS configuration consists of one or more remote locations, each served by its own private branch exchange (PBX) and attendant(s), and a single PBX location where the CAS attendant(s) are located. (See Fig. 2-I.) Each remote location has access to the CAS attendant through Release Link Trunks (RLT). In addition, the remote locations are interconnected by tie trunks”
4. EQUIPMENT	6	
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B. CAS Attendant Features not Supported for Remote Locations	7	2.02 When a call from a PBX in a remote location requires attendant assistance, an idle RLT at the remote PBX is seized and the call is presented to the CAS attendant. If an idle RLT is not available, the call is queued until an RLT becomes idle. The CAS attendant can then extend the call to a station at the remote location. The CAS attendant must also handle calls originating from the main PBX which require attendant assistance.
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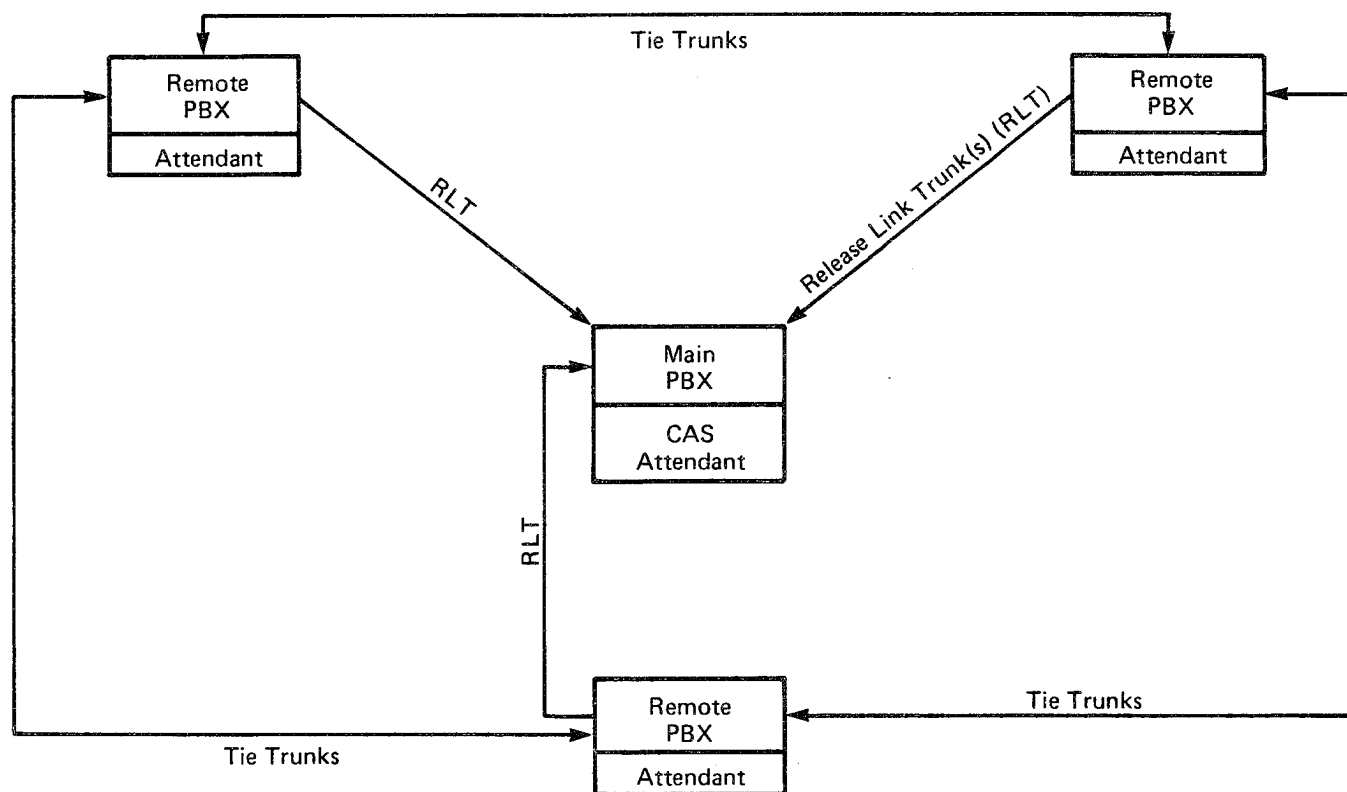


Fig. 2-1 — Typical CAS Configuration

2.03 The types of call which require attendant assistance and can be handled by a CAS attendant are :

- (a) Listed Directory Number (LDN) calls,
- (b) Dial "0" calls ("0" is optional if the flexible attendant DN is used),
- (c) Recalls, intercepts, or transfer to attendant,
- (d) Access to special services such as WATS or FX trunks.

2.04 Tones are sent from the remote PBX to the CAS attendant when the CAS attendant answers to identify the type of incoming call, and to provide confirmation of special features during dialing; i.e., camping-on a busy destination.

2.05 The SL-1 is compatible as a main PBX or a remote PBX at installations which consist of SL-1 and non-SL-1 PBX types. Non-SL-1 PBX-types must be arranged in accordance with AT&T technical advisory number 10 (TA-10) to accept the CAS option.

3. FEATURE DESCRIPTION

A. Controlling CAS Service

3.01 The attendant console at the remote PBX is equipped with a key to activate/deactivate CAS. When the CAS key is operated, its associated LED changes state. If the LED is lit, then the remote PBX is in CAS service and all calls requiring attendant service will be routed to the CAS attendant over RLT. If the LED is dark, the remote PBX is in local service and all calls are handled by the local attendant. The CAS key, assigned automatically, is located above the LPK 5 key on the console.

3.02 If an attendant console is not required at a remote location, up to ten SL-1 sets with GAS keys can be equipped.

B. Listed Directory Number Calls

3.03 Calls to the LDN of the main PBX are treated in the normal manner; i.e., an Incoming Call Indication (ICI) is presented to the attendant and normal call-handling procedures

are used. If the remote location is in the CAS-deactivated mode, then LDN calls to the remote location are also handled in the normal manner by the attendant at the remote location

3.04 When the CAS feature is activated at the remote location, then LDN calls to that location are presented to the CAS attendant with an RLT X ICI indication, where X represents the location of the remote PBX. If the CAS attendant console is equipped with numeric display, the RLT access code and route member number are displayed.

3.05 When the CAS attendant selects the incoming call by operating the loop key or ICI key, answer supervision is sent to the remote PBX over the RLT. Two 100 ms bursts of 440 Hz are given, if defined in customer overlay program 15. The RLT X ICI indication identifies the call and prompts the appropriate verbal response.

3.06 After obtaining the desired party's number, the CAS attendant presses the signal remote (SIG REM) key, which is assigned automatically, located above the LPK 5 key. The main SL-1 responds by sending a timed flash over the RLT to the remote PBX. This signal prompts the remote PBX to connect a register or DIGITONE* receiver to the RLT. When an idle register or Digitone receiver has been applied to the RLT, dial tone from a Tone and Digit Switch (TDS) at the remote location is sent to the CAS attendant. The CAS attendant then keys in the destination DN on the dial pad. On completion of outpulsing, the CAS attendant hears ringback or busy tone. At this time, the CAS attendant can disconnect from the RLT by operating the release (RLS) key and thus be available to serve other calls.

3.07 If all RLT from a remote location are in use, LDN calls are queued and routed to an RLT when one becomes idle.

C. Timed Reminder Recall

3.08 If the remote PBX is arranged to recall the attendant on "no answer" calls, then, after the prescribed timeout (and the station has not answered, or the CO trunk has not disconnected), an idle RLT is seized and the call is presented to the CAS attendant. This time a different attendant may answer the call; however, an RLT X ICI indication is still received.

3.09 When the CAS attendant answers the call, the remote PBX detects answer supervision over the RLT, sends ringback tone for 2 seconds

to the attendant, and then connects the call. The ICI and the ringback tone prompt the attendant to answer in the appropriate manner. During answer, the CAS attendant and calling party talk over the ringback of the called station. The attendant can press RLS DEST to stop ringback.

3.10 If the calling party wishes to continue waiting, the CAS attendant at the main location releases. Disconnect is detected at the remote PBX, and the RLT connection is removed. Further timed recalls cause the above process to be repeated until either the called station answers, or the calling party disconnects.

3.11 When the remote location is arranged for Direct Inward Dialing (DID), and the called station is arranged to forward calls to the attendant on no answer, the call is treated as in 3.08 and 3.09.

D. Camp-On

3.12 When the CAS attendant attempts to complete a call to a station which is busy, the call is automatically camped-on to the busy station, provided a call is not already camped-on to that station. The remote PBX transmits a camped-on conflation tone (100-ms burst of 440-Hz tone) to the CAS attendant over the RLT, and then connects to the calling party. If the calling party wishes to wait, the CAS attendant releases, and a disconnect signal is transmitted to the RLT, which releases. When the camped-on station goes on-hook, the camped on call is automatically presented.

3.13 If the called station does not answer within the specified timeout period (timed reminder feature), the remote PBX seizes an idle RLT and presents the calling party to the CAS attendant. When the CAS attendant answers, answer supervision is detected by the RLT and the remote PBX transmits a 100-ms burst of 440-Hz tone to the CAS attendant so that the attendant can answer appropriately. The CAS attendant can then release from the calling party by operating the release (RLS) key, or disconnect from the called station by operating the release destination (RLS DEST) key.

3.14 The call-forward-busy feature, where it exists, automatically causes Direct Inward Dial (DID) calls which encounter a busy station, to be routed to an RLT, and when the CAS attendant answers, also causes the same tone as for a dial 0 call to be sent. This occurs only for stations which have a call-forward-busy allowed (CF-BY) class of service, and no other forwarding or hunting arrangements to route the call to another station when busy.

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E. Silent Hold

3.15 The attendant may wish to hold incoming calls at the PBX before connecting them. To accomplish this, a silent hold feature is provided. When the attendant activates the silent hold feature, the CO trunk is placed in the hold state, and the RET is released. After the timed reminder has timed out, the RLT is resealed and the CO trunk is **reconnected** to the CAS attendant.

3.16 When an incoming call has been answered and the decision is made to place the call in the hold condition, the CAS attendant presses the signal remote (SIG REM) key to cause a **flash** to be sent to the **RLT**. The attendant receives dial tone from the remote PBX and dials the DN assigned for the silent hold code. The hold code is outputted to the remote PBX, translated, and the CO trunk is placed on hold. The remote PBX then sends a hold confirmation tone (4 through 6, **50-ms** bursts of 440-Hz tone) to the attendant who then releases by operating the position release (RLS) key. The remote PBX recognizes the disconnect signal and releases the RLT.

3.17 A recall occurs after the recall timer has elapsed. An idle RET is seized and the held call is presented to the GAS attendant. **When** the attendant answers, the remote PBX receives answer supervision, transmits an information tone (4 through 6, **50-ms** bursts of **440-Hz** tone) to the attendant and removes the hold condition from the CO trunk. The attendant can then complete the call.

3.48 The CAS attendant can also hold the RLT at the console by operating the HOLD key. This differs from silent hold in that the CO trunk is held at the main PBX (over the RET), rather than at the remote end.

F. Dial "0" Calls

3.19 When the remote PBX translates a "0" (zero) request, the calling station or trunk is connected to an idle RLT, and the call is presented to the CAS attendant. When the CAS attendant answers, the RLT detects answer supervision, transmits two **100-ms** bursts of 440-Hz tone to the attendant, and connects the calling station or trunk to the CAS attendant. The call is then handled as an LDN call. See 3.36 for flexible attendant DN information.

3.20 The CAS attendant is restricted in placing dial "0" calls on hold. If the hold code is dialed, overflow tone is sent to the CAS attendant.

G. Transfer to Attendant

3.21 The CAS attendant can be recalled by a station so that the attendant can transfer a CO trunk or other trunk to another station. The **call** is presented to the **CAS** attendant in the same manner as a dial "0" call. The CAS attendant can then transfer the **call** by operating the release destination (RLS DEST) key followed by the procedure for an LDN call. Operation of the RLS DEST key causes a timed flash to be sent over the RLT, which drops the initial station party connection.

3.22 The CAS attendant can be recalled as a result of an NE-500 or NE-2500 set user's flashing the switchhook and dialing "0" or an SL-1 set user's operating an attendant recall key.

3.23 The above procedure is also followed for **all** calls intercepted to the CAS attendant.

H. Calls to the Tie Trunks

3.24 The CAS attendant can **connect** an incoming CO call to an extension at another PBX, or transfer calls to the connecting PBX. The operation is the same as that for completing an LDN or transfer call at the main PBX, except that the attendant dials a trunk access code instead of a station Directory Number (DN). When second dial tone is sent from the remote **PBX**, the attendant can dial the station DN.

3.25 This type of call is limited in that:

- (a) The recall on no-answer feature is inactive.
- (b) The camp-on feature is inactive.
- (c) **The call** transfer feature operates in accordance with the capabilities of the system on which the **call** transfer is being attempted.

Note: Disconnect supervision must be provided either by the incoming CO trunk, or repeated by the tie trunk from the remote PBX. If not, SE-1 disallows these types of **connection**.

I. Access to Special Services (WATS, FX Trunks, etc.)

3.26 Special service circuits can either be located and controlled at one location, or distributed among various locations with each location controlling access locally.

3.27 When special service circuits are located at the main PBX the CAS attendants have trunk group busy indicators and control of trunk group access. In this case, all remote locations access the special service circuits via tie trunks. Tie Trunk Group Access Restrictions (TGAR) at the main PBX can be arranged to provide dial access to the special services, or automatically route incoming calls to the CAS attendants to control access manually.

3.28 If the special service circuits are situated at a remote location, they can be dial accessed and controlled through TGAR, or a unique access code can be assigned to access the local attendant and bypass selection of an RLT. The local attendant DN can also be used by station users desiring attendant controlled connections.

3.29 Attendant conference connections can only be set up by the attendant at each remote location. The CAS attendant **cannot** set up a conference call for a remote location.

J. Night Service

3.30 Night service at the CAS attendant location is applicable to incoming calls, dial "0" calls, etc., at the main location, and calls routed over the RLT from remote locations.

3.31 Calls from remote locations over an RLT can be transferred by the night answer station at the main PBX, back to the remote location over a tie trunk; however, the RLT is held for the duration of the call.

3.32 Night service is activated separately by each remote location at its local console (or SL-1 set). This is usually prearranged for a given time of day, or in response to a call from the chief CAS attendant, and is done prior to the CAS attendant activating night service. If the CAS attendant activates night service before the remote locations, calls routed to the RLT by the remote PBX will be directed to the night station at the main PBX.

3.33 The three types of night service offered on ~~SL-1~~; i.e., flexible, night answer station, or Trunk Answer From Any Station (TAFAS) can be assigned independent of any CAS arrangements. CAS service and night service are independent and when both features are activated, CAS takes precedence.

3.34 When night service is activated at a remote SL-1, any calls in the RLT queue are automatically rerouted to the assigned night answer station or TAFAS as applicable. Calls in progress on the RLT are completed by the CAS attendants.

K. Trunk Group Busy

3.35 Trunk group busy is modified, so that operation of the trunk group busy key on the CAS console associated with an incoming RLT trunk route is ignored.

L. Flexible Attendant Directory Number

3.36 Each remote location can be assigned two attendant directory numbers to provide access to the attendant(s). The first is called the attendant DN (ATTN DN). Calls to this DN are presented over an RLT to the CAS attendant if the remote location is in CAS service. If the remote location is not in CAS service, the call is presented to the local attendant, or the night number if the remote location is in night service. The second is called the local attendant DN (LOCAL ATTN DN). Calls to this DN are presented to the local attendant, or night number if the customer is in night service. A call **dialed to the LOCAL ATTN DN is never terminated on an RLT.**

3.37 The customer data block **service change** program (553-2001-221) **allows the** modification of these two DN. The DN assigned must be DN taken from the numbering plan, and must not conflict with any other DN. If it is desired that "0" be the access code for the attendants, then it must be specified as the ATTN DN.

3.38 These DN cannot be made Do-Not-Disturb (DND) busy, and if they appear on a lamp field array, the associated LED is always dark.

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4 . EQUIPMENT

A. Tone and Digit Switch (TDS)

4.01 A remote PBX which is arranged for the CAS feature requires the new **QPC251** TDS card. No mix or match with any other TDS card is allowed - all TDS cards at the remote PBX must be **QPC25 1**. There are no restrictions on the type of TDS card used at the main CAS location. Tables 4-A and 4-B list the tones used to accommodate CAS, including **interruption** rates and durations.

TABLE 4-A
SL-1 PRECISE TONES WITH CAS

STONE	FREQUENCY (Hz)	LEVEL (dB)
Dial Tone	350 i- 440	16*
Busy Tong, Overflow Tone	480 + 620	23*
Audible Ringback	440 + 480	18*
Miscellaneous Tone	440	16

* per tone, combined is 3 dB higher.

TABLE 4-B
SL-1 TONE INTERRUPTIONS WITH CAS

STONE	CADENCE
LDN	100 ms of 480 Hz; 100 ms of 440 Hz, 100 ms of 480 Hz
camp-on	100 ms on/off
Dial "0" Recall	100 ms on/100 ms off/ 100 ms on
No Answer Recall	300 ms on/off
Busy	0.5 s on/0.5 s off
Overflow	0.25 s on/0.25 s off
Hold Confirmation, Recall	0.05 s on/0.05 s off
Standard Ringing	2 s on/4 s off

4.02 The maximum delay on application of any of the CAS tones, subsequent to detecting answer supervision from the main CAS location, is 150 ms. A break in the middle of a tone cycle is inhibited so that a complete tone is always heard by the GAS attendant.

B. Release Link Trunk

4.03 Various trunk types can be used to interface a remote BBX to the main PBX when the CAS feature is required. The preferred configuration of an

WET circuit is to have either a 2-wire (QPC71) or 4-wire (QPC237) tie trunk at both the remote and main locations with a repeater facility between the two. Alternately, the remote location can be equipped with a **QPC250 RLT** pack (553-2681-180) to interface with a **QPC70 CO** trunk pack at the main location. This application is limited to a maximum loop resistance of 1408 ohms. Part 6 of this section describes how to calculate the quantity of RLT circuits required.

5 . APPLICATION CONSIDERATIONS

A. Interaction with Other Features

Call Detail Recording (CDR)

5.0 I When an SL-1 CAS network is equipped with CDR, the following points should be noted:

- (1) The CDR should be located at the main location. This also applies to any special service trunks.
- (2) Each remote Sk-1 should be connected to the main CDR through the CDR concentrator option
- (3) Downstream-processing of the CDR tapes must correlate tandem calls from a remote BBX through the main PBX via tie trunks to special services. A record showing the facilities used or stations involved in any tandem connection, is given for each location separately. The CDR downstream-processing must rebuild the tandem call based on the time correlation and trunk route correlation. Every effort should be made to ensure time synchronization among the main location and remote locations.
- (4) Incoming calls completed by a CAS attendant only show the record for the completed connection to the destination station; i.e., the same as if a local attendant had served the call.

Automatic Route Selection (ARS)

5.02 Automatic Route Selection should be provided at the main CAS location and accessed by the trunks from all remote locations where the least cost routing to all special services and toll routes is desired. All special services and outgoing toll recalls are placed through the main location, eliminating potential supervision problems with tandem tie trunk connections when ARS is provided separately at each remote location.

5.03 To ensure adequate station class-of-service restrictions for controlling access to special services and toll, the following actions should be taken

- (1) Two separate tie trunk routes from each remote location to the main location should be supplied. **One** route should be unrestricted or conditionally toll-denied, and the other route should be toll-restricted.
- (2) Assign TGAR codes to control access from stations at each remote location, to one or the other of these trunk routes.

Remote Peripheral Equipment (RPE)

5.04 For the purpose of CAS, each RPE location in a CAS network is considered to be a part of the SE-1 system to which it is connected; i.e., it shares the same numbering plan and is identified to the CAS attendant as part of whatever location with which it is sharing the common equipment and network.

Automatic Number Identification (ANI) and Automatic Identification of Outward Dialing (AIOD)

5.05 When **ANI** using direct toll **trunks** to the **CAMA** center, or **AIOD** using the data link method, are used in conjunction with **CAS**, each location must be individually connected to its respective **CO** or toll center if station **detail** billing for all locations is to be provided.

5.06 Where possible, the CDR option is recommended for passive cost **control** and cost allocation when arranged in a CAS network. This is only practical if the main and all remote locations are SL- 1. Another alternative is to provide toll-restriction at all remote locations and force all toll calls to be routed through the CAS attendants for manual record-keeping,

B. CAS Attendant Features not Supported for Remote Locations

Alarm Lamps

5.07 The CAS attendants do not receive major or minor alarm indications for remote locations. These alarms are provided at the local console at each remote location.

Barge-in/Busy Verify

5.08 The **CAS** attendants are able to barge-in on trunks located at the main location, and busy verify stations at the main location, but are not able to perform these functions for remote locations. The local console at each remote location can barge-in/busy verify its local trunks/stations. Neither the CAS attendant or local attendant can barge-in on an idle or busy RLT.

Lamp Field Array

5.09 The lamp field array associated with a CAS attendant console indicates the status of stations or trunks only for the main location.

Call-Waiting

5.10 Calls waiting for service from an RLT at each remote location are not included in the **call**-waiting indication given to the CAS attendants; calls, however, which have seized an RLT and are queued for service at the main location, are included in the call-waiting indication.

Calling Number Display

5.11 The station number, or access code and route member number of the originating station or trunk at the remote location is not displayed to the CAS attendant answering the call. Instead, the access code and member number of the RLT over which the call is presented, is indicated.

Trunk Group Busy/Control of Trunk Group Access

5.12 Release Link Trunks from remote locations are assigned to a Trunk Group Busy key at the CAS attendant console. However, the ability to busy out a group of RLT from the CAS console, is denied.

Conference

5.13 The CAS attendants cannot set up conference connections involving stations or trunks at the remote locations. All conference connections must be made by the local attendant at each remote location.

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6. ENGINEERING INFORMATION

A. Traffic Measurements

6.01 **Traffic** measurements are provided separately for each location. To coordinate the accumulation of **traffic** data on a network basis, the system clocks and **traffic** schedules for each location should be set and maintained in reasonable synchronization.

TRUNKS – TFC002

6.02 At each remote location, the Release Link Trunks (RLT) are considered an outgoing trunk route. Outgoing usage and peg count are accumulated for **traffic** schedule TFC002. All other fields in this **traffic** schedule are zeroed. Section 553-2001-450 details the traffic schedule outputs.

6.03 At the main location, the RET are considered an incoming trunk route for each remote location. Usage, peg count, failure to match, and all trunks busy measurements are recorded for each RLT group in traffic schedule TFC002.

QUEUE – TFC003

6.04 If the CAS feature is not active, queue measurements are recorded at the main and remote locations in the **normal manner**. When the CAS feature is active, calls that originate at the remote PBX and are handled over an RLT by the CAS attendant at the **main** location, are measured and recorded in TFC003 at the **remote PBX**. For these measurements, the remote PBX interprets the RLT as a **local** attendant. Calls dialed **specifically** to the **local** attendant at the remote PBX are also reflected in TFC003, when the CAS feature is active. TFC003 measurements are recorded in the normal manner at the main PBX.

6.05 TFC003 definitions should be interpreted as follows for measurements at the **remote PBX** when CAS is active (Fig. 6-1).

(1) *Average Attendant Response*. The average time between a call being presented to an attendant console; i.e., RLT, and the CAS attendant answering it.

(2) *Average Time in Queue*. The time that calls spend in the attendant queue; i.e., RLT queue, averaged over all calls that are placed into that queue.

(3) *Average Speed of Answer*. The time that a call waits for its request to terminate at an attendant; i.e., RLT, is recognized by the system and before it is answered.

(4) *Peg Count of Calls Delayed*. This count is incremented whenever a call is removed from the attendant; i.e., RLT, queue.

(5) *Peg Count of Abandoned Calls*. Incremented whenever a call abandons before being answered by the CAS attendant.

CONSOLE – TFC004

6.06 When the CAS feature is not active, the local attendant at the remote PBX handles all calls which require attendant services; i.e., LDN, recall, etc. These calls are measured and recorded in TFC004 at the remote PBX. When the CAS feature is active, TFC004 measurements will only reflect calls dialed specifically to the local attendant – all other calls requiring attendant assistance are directed to the CAS attendant at the main location.

6.07 At the main PBX, TFC004 measurements are recorded in the normal manner. When the CAS feature is active, calls originating from the remote PBX are treated as incoming trunk calls at the main PBX.

B. Service Change

6.08 The following data blocks are affected with the **implementation of CAS**:

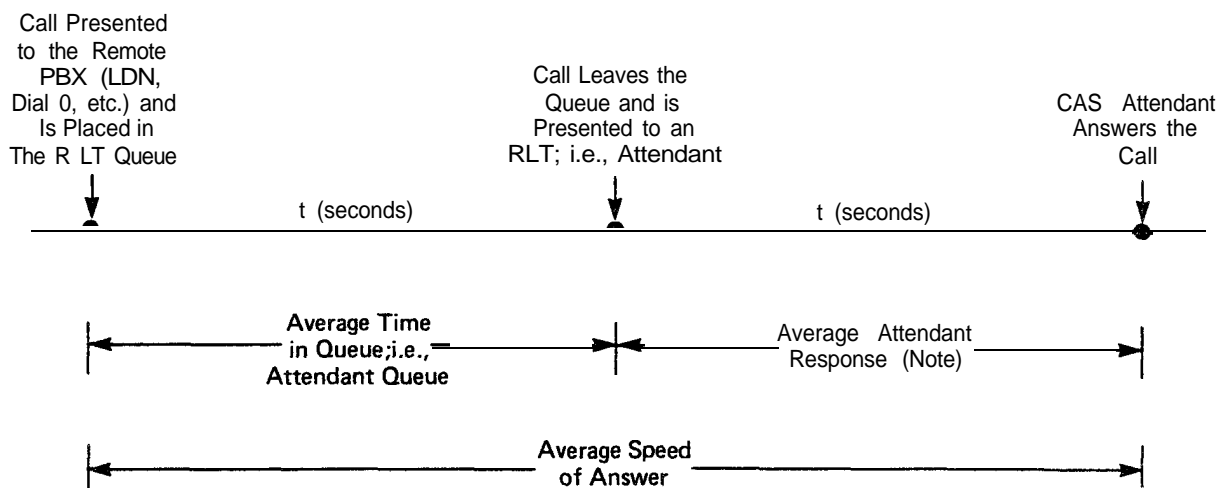
At remote locations:

- (1) customer data block,
- (2) SL-1 set data block (if a console is not equipped),
- (3) trunk data block,
- (4) route data block.

At the main location:

- (1) customer data block,
- (2) trunk data block,
- (3) route data block.

→ Refer to 553-2001-220 and -221 or 553-2yy1-310
→ and -311 for complete service change information.



Note: Where ground start RLT trunks are used, i.e., a QPC250 at the remote PBX and a QPC70 at the main PBX, the average attendant response time includes a minimum of 2.5 s delay from the time the RLT is seized at the remote PBX to the time that it is interpreted as an incoming call at the main PBX.

Fig. 6-1 – TFC003 Measurements at a Remote PBX when a Call Waits for a RLT to Become Idle

C. Memory and Real Time Requirements

6.09 The impact of the CAS feature on memory and real time requirements of a particular system can be calculated using 553-2001-1 5 1 or 553-220 1- 15 1 and their associated appendices. ←

D. Maintenance and Testing

6.10 An RLT at a remote location can be tested two ways. The first method is through a manually invoked overlay program which performs the normal trunk diagnostics (see 553-2001-505 or 553-2301-511). The second method allows access to a specific RLT by dialing the assigned directory number of that RLT. This method permits a tester to dial access a particular RLT, and when the CAS attendant answers, perform a transmission test on the RLT.

6.11 At the main CAS location, the trunk maintenance diagnostics of 553-2001-505 or 553-2301-511 are used to test a QPC70 trunk circuit pack associated with an RLT. Section 553-2681-300 details tests of CAS feature operation. ←

E. Provisioning

6.12 A remote PBX arranged for the CAS feature must be equipped with QPC25 I tone and digit switches. No other type of tone and digit switch is acceptable. The main PBX can use any type of TBS currently available.

6.13 Tie trunks required between the remote locations, and between the remote locations and the main location, depend on the location of special service circuits and the amount of inter-PBX traffic.

RLT Circuit Requirements

6.14 To calculate the number of RLT circuits required between a particular remote PBX and the main PBX, the following grades of service must be specified. ←

(a) **Average Speed of Answer.** The average time in seconds a call from the remote PBX, which requires an attendant, waits at the *main* PBX for an attendant to answer.

(b) **Average RLT Delay.** The average time in seconds a call from the remote PBX, which requires an attendant, waits at the *remote* PBX for an RLT to become idle under Average Busy-Season Busy-Hour (ABSBN) traffic.

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Note 1: Where ground start signaling is used for RLT circuits; i.e., a QPC250 RLT circuit pack at a remote PBX and a QPC70 CO trunk circuit pack at the main PBX, Add a minimum of 2.5 seconds to the average RLT delay. There is a minimum average delay of 2.5 seconds from the time the RLT is seized at the remote PBX to the time that it is interpreted as an incoming call at the main PBX.

Note 2: RET 20% Overload Delay — This delay is fixed at 10.0 seconds and is reflected in the values shown in Table 6-A through 6-R.

6.15 Furthermore, the calculation of the number of RLT circuits required between a remote PBX and the main PBX requires the following information.

- (a) The number of calls per hour in the remote PBX which require an attendant
- (b) The Attendant Work Time (AWT). This is the average time in seconds taken by an attendant at the main PBX to service a call.

6.16 The engineering procedure is:

- (1) Calculate the attendant traffic (T_a) generated at the remote PBX by using the formula:

$$T_a = \tau \times \frac{AWT}{100} \text{ CCS}$$

Where:

τ = the number of calls per hour in the remote PBX which require an attendant

AWT = attendant work time.

- (2) Select the appropriate table (Table 6-A through 6-R) according to the specified grade of service, i.e., average speed of answer and average RLT delay.

- (3) Select the appropriate column in the table according to the attendant work time.

- (4) Select a number (T) in the column chosen which is greater than or equal to the value T_a determined in 1); i.e., $T \text{ CCS} \geq T_a \text{ CCS}$. The number of RLT circuits required is indicated horizontally to the left of the value T .

Example:

Given:

Average speed of answer	= 2s grade of
Average RLT delay	= 2s service
Attendant work time (AWT)	= 20s
Attendant calls/hour in the remote PBX (7)	= 500

Then,

$$T_a = 500 \times \frac{20}{100} \text{ CCS}$$

$$T_a = 100 \text{ CCS}$$

Table 6-A is chosen because it meets the given grade of service values. From this table, in the AWT 16-20 column, the value T of 120 CCS is selected ($T \text{ CCS} \geq T_a \text{ CCS}$). For the value $T = 120 \text{ CCS}$, the table shows that six RLT circuits are required for this particular application.

CAS Attendant Requirements

6.17 To calculate the number of attendants required at the main PBX serving multiple remote PBX, the Average Speed of Answer grade of service must be specified. This delay constitutes the average time in seconds a call from either a remote PBX or the main PBX, which requires an attendant, waits at the main PBX for an attendant to answer.

6.18 Furthermore, the following information is required.

- (a) The attendant **traffic** (T_a) generated at the main PBX,
- (b) The attendant traffic (T_a) generated at each remote PBX.

Use the formula $T_a = \tau \times \frac{AWT}{100}$ CCS

Where:

τ = the number of **calls** per hour in a remote PBX or the main PBX which require an attendant.

AWT = the attendant work time. This is the average time in seconds taken by an attendant at the main PBX to service a call.

6.19 The engineering procedure is:

- (1) Calculate the total attendant **traffic** (T_t) by adding the attendant **traffic** (T_a) from each remote PBX to the attendant **traffic** (T_a) at the main PBX.
- (2) Select the appropriate table (Table 6-S through 6-X) according to the specified grade of service; i.e., average speed of answer.

(3) Select the appropriate column in the table according to the attendant work time (AWT).

(4) The number of attendants required corresponds to a value T CCS in the column chosen, where $T \text{ CCS} \geq T_t \text{ CCS}$.

Note: The values shown in Table 6-S through 6-X reflect the following factors:

- (a) A 5 percent attendant overload delay (10.0 second),
- (b) 92 percent attendant occupancy.

Example:

Given:

Average speed of answer	= 2s grade of service
Attendant work time (AWT)	= 20s
Attendant traffic (T_a) at main PBX	= 170 CCS
Attendant traffic (T_a) at remote PBX #1	= 100 ccs
Attendant traffic (T_a) at remote PBX #2	= 120 ccs

Then:

$$T_t = 170 + 100 + 120 \text{ CCS}$$

$$T_t = 390 \text{ CCS}$$

Table 6-S is chosen because it meets the given grade of service value. From the table, in the AWT 16-20 column, the value T of 395 CCS is selected ($T \text{ CCS} \geq T_t \text{ CCS}$). For the value T = 395 CCS the table shows that 14 attendants are required at the main PBX to handle this particular application.

TABLE 6-A
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 2.00 SECONDS AVERAGE RLT DELAY \leq 2.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	5	4	3	2	2	1	1	1	1	1	1	1
2	24	22	20	18	17	16	15	14	13	13	12	12
3	45	45	43	41	39	37	36	34	33	32	31	31
4	68	70	68	66	63	61	60	58	56	55	54	53
5	92	96	95	92	90	87	85	83	82	80	79	77
6	116	123	122	120	117	115	112	110	108	107	105	104
7	138	150	150	148	145	143	140	138	138	134	132	131
8	159	178	179	177	174	171	169	167	164	162	160	158

TABLE 6-B
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 2.00 SECONDS AVERAGE RLT DELAY \leq 4.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	9	1	6	5	4	3	3	2	2	2	2	2
2	31	29	27	25	23	22	21	20	19	18	17	17
3	53	55	53	51	49	41	45	43	42	41	40	39
4	74	81	81	78	76	14	72	70	68	61	65	64
5	95	105	106	105	104	102	100	98	96	94	92	91
6	117	130	132	131	130	128	126	125	123	121	120	118
7	138	154	158	158	156	155	153	151	149	148	146	145
8	159	179	183	184	183	182	180	178	176	174	173	171

TABLE 6-C
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 2.00 SECONDS AVERAGE RLT DELAY \leq 6.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	11	9	8	7	6	5	4	4	3	3	3	3
2	32	33	32	30	28	26	25	24	23	22	21	20
3	53	57	56	54	53	51	49	48	47	45	44	43
4	74	81	81	80	78	76	74	73	71	70	69	67
5	95	105	106	105	104	102	100	98	97	95	94	93
6	117	130	132	131	130	128	126	125	123	121	120	118
7	138	154	158	158	156	155	153	151	149	148	146	145
8	159	179	183	184	183	182	180	178	176	174	173	171

TABLE 6-D
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 4.00 SECONDS AVERAGE RLT DELAY \leq 2.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	3	3	2	2	2	1	1	1	1	1	1	1
2	17	18	17	16	15	14	14	13	13	12	12	11
3	33	37	37	36	35	34	33	32	31	31	30	29
4	50	58	59	59	58	57	56	55	54	53	52	51
5	68	80	83	83	82	81	80	79	78	77	75	74
6	86	102	107	108	108	107	105	104	103	102	101	100
7	105	125	132	134	134	133	132	131	131	129	127	126
8	122	149	157	160	160	160	159	158	156	155	154	153

TABLE 6-E
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 4.00 SECONDS AVERAGE RLT DELAY \leq 4.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	6	5	4	4	3	3	3	2	2	2	2	1
2	22	24	23	22	21	20	19	18	18	17	16	16
3	39	45	46	45	44	43	42	41	40	39	38	37
4	56	68	70	70	69	68	67	66	65	63	62	61
5	72	88	94	95	95	95	94	92	91	90	89	87
6	89	109	116	119	120	119	119	118	117	116	115	114
7	105	130	139	143	144	144	144	143	142	141	140	139
8	122	151	162	167	169	170	169	169	168	167	166	165

TABLE 6-F
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 4.00 SECONDS AVERAGE RLT DELAY ≤ 6.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	7	1	6	5	5	4	4	3	3	3	3	2
2	22	24	27	21	26	25	24	23	22	21	21	19
3	40	47	49	49	48	47	46	45	44	43	42	42
4	56	68	71	12	71	71	70	69	68	67	66	65
5	72	88	94	95	95	95	94	93	92	91	90	89
6	89	109	116	119	120	119	119	118	117	116	115	114
7	105	130	139	143	144	144	144	143	142	141	140	139
8	122	151	162	161	169	110	169	169	168	161	166	165

TABLE 6-G
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 6.00 SECONDS AVERAGE RLT DELAY ≤ 2.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	2	2	2	1	1	1	1	1	1	1	1	0
2	12	14	15	14	14	13	13	12	12	11	11	11
3	25	31	32	33	32	32	31	30	30	29	29	28
4	39	49	52	53	53	53	52	52	51	50	49	49
5	53	68	73	75	76	76	75	75	74	73	72	72
6	68	87	95	98	99	99	99	99	98	97	97	96
7	83	107	117	121	123	124	124	124	123	123	122	121
8	98	128	140	145	148	149	150	150	149	149	148	147

TABLE 6-H
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 6.00 SECONDS AVERAGE RLT DELAY \leq 4.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	4	4	4	3	3	2	2	2	2	2	1	1
2	16	20	20	20	19	18	18	17	11	16	16	15
3	31	38	40	41	40	40	39	38	38	37	36	36
4	44	58	62	64	64	63	63	62	61	61	60	59
5	58	76	83	87	88	88	88	87	87	86	85	84
6	71	94	104	108	111	112	112	112	112	111	111	110
7	85	112	124	130	134	135	136	136	136	136	135	134
8	98	131	145	153	157	159	160	160	160	160	160	159

TABLE 6-I
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 6.00 SECONDS AVERAGE RLT DELAY \leq 6.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	5	6	5	5	4	4	3	3	3	3	2	2
2	18	23	24	23	23	22	21	21	20	19	19	18
3	31	40	43	44	44	44	43	43	42	41	41	40
4	44	58	63	65	66	66	66	65	64	64	63	62
5	58	76	83	87	88	89	89	88	88	87	86	86
6	71	94	104	108	111	112	112	112	112	111	111	110
7	85	112	124	130	134	135	136	136	136	136	135	134
8	98	131	145	153	157	159	160	160	160	160	160	159

TABLE 6-J
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 5.00 SECONDS AVERAGE RLT DELAY \leq 2.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	2	2	2	2	1	1	1	1	1	1	1	0
2	14	16	16	15	14	14	13	13	12	12	11	11
3	28	34	35	34	34	33	32	31	31	30	29	29
4	44	53	55	56	55	55	54	53	52	51	51	50
5	60	73	78	79	79	78	77	77	76	75	74	73
6	76	94	101	103	103	103	102	101	101	100	99	98
7	93	116	124	127	128	128	128	127	126	125	124	123
8	109	137	148	152	154	154	154	154	153	152	151	150

TABLE 6-K
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 5.00 SECONDS AVERAGE RLT DELAY ≤ 4.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	5	5	4	3	3	3	2	2	2	2	2	1
2	19	22	22	21	20	19	18	18	17	17	16	16
3	35	41	43	43	42	41	40	40	39	38	37	36
4	50	62	66	67	66	66	65	64	63	62	61	60
5	64	82	88	91	92	91	91	90	89	88	87	86
6	79	101	110	113	115	116	115	115	114	114	113	112
7	94	121	131	136	139	140	140	140	139	138	138	137
8	109	140	153	160	163	164	165	165	164	164	163	162

TABLE 6-L
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 5.00 SECONDS AVERAGE RLT DELAY \leq 6.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	6	6	6	5	4	4	4	3	3	3	2	2
2	21	25	25	25	24	23	22	21	21	20	19	19
3	35	43	46	46	46	45	45	44	43	42	41	41
4	50	62	67	68	69	68	68	67	66	65	64	64
5	64	82	88	91	92	92	91	91	90	89	88	87
6	79	101	110	113	115	116	115	115	114	114	113	112
7	94	121	131	136	139	140	140	140	139	138	138	137
8	109	140	153	160	163	164	165	165	164	164	163	162

TABLE 6-M
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 8.00 SECONDS AVERAGE RLT DELAY \leq 2.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	1	1	1	1	1	1	1	1	1	1	0	0
2	10	12	13	13	13	12	12	11	11	11	11	10
3	20	26	29	29	30	29	29	29	28	28	27	27
4	32	42	46	48	49	49	49	49	48	48	47	47
5	43	59	65	69	70	71	71	71	70	70	70	69
6	56	76	85	90	92	93	94	94	94	93	93	93
7	68	93	105	111	115	116	117	118	118	118	117	117
8	81	111	126	133	138	140	142	142	143	143	143	142

TABLE 6-N
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 8.00 SECONDS AVERAGE RLT DELAY ≤ 4.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	3	3	3	3	2	2	2	2	2	2	1	1
2	13	17	18	18	17	17	17	16	16	15	15	14
3	25	33	36	37	37	37	37	36	36	35	35	34
4	37	50	55	58	59	59	59	59	58	58	57	57
5	48	66	75	79	82	83	83	83	83	82	82	81
6	59	82	94	100	103	105	106	107	107	107	106	106
7	70	98	112	120	124	127	129	130	130	130	130	130
8	82	115	131	140	146	149	152	153	154	154	154	154

TABLE 6-Q
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 8.00 SECONDS AVERAGE RLT DELAY \leq 6.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	4	4	4	4	4	3	3	3	3	2	2	2
2	15	19	21	21	21	20	20	19	19	19	18	18
3	26	35	38	40	41	41	41	40	40	39	39	38
4	37	50	56	59	61	62	62	62	61	61	61	60
5	48	66	75	79	82	83	84	84	84	84	83	83
6	59	82	94	100	103	105	106	107	107	107	106	106
7	70	98	112	120	124	127	129	130	130	130	130	130
8	82	115	131	140	146	149	152	153	154	154	154	154

TABLE 6-P
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 10.00 SECONDS AVERAGE RLT DELAY \leq 2.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	i-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	1	1	1	1	1	1	1	1	1	0	0	0
2	8	10	11	11	11	11	11	11	11	10	10	10
3	17	23	26	27	27	27	27	27	27	26	26	26
4	26	37	42	44	46	46	46	46	46	46	46	45
5	36	52	59	63	65	66	67	67	67	67	67	67
6	47	61	77	82	86	88	89	89	90	90	90	89
7	58	8	3	95	103	107	110	111	112	113	113	113
8	68	98	114	123	129	132	134	136	137	137	137	137

TABLE 6-Q
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 10.00 SECONDS AVERAGE RLT DELAY ≤ 4.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME IN (AWT) SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	2	2	2	2	2	2	2	2	1	1	1	1
2	11	14	16	16	16	16	15	15	15	14	14	14
3	20	28	32	34	34	35	35	34	34	34	33	33
4	31	44	50	53	55	55	56	56	56	55	55	55
5	41	59	68	73	76	78	78	79	79	79	79	78
6	50	73	85	92	96	99	101	102	102	102	102	102
7	60	87	102	111	116	120	122	124	125	125	125	125
8	70	102	120	130	137	141	144	146	147	148	149	149

TABLE 6-R
RELEASE LINK TRUNK CIRCUIT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 10.00 SECONDS AVERAGE RLT DELAY ≤ 6.00 SECONDS												
RLT CIRCUITS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	3	4	4	3	3	3	3	3	2	2	2	2
2	12	17	19	19	19	19	19	18	18	18	17	17
3	21	30	35	37	38	38	38	38	38	38	37	37
4	31	44	51	55	57	58	58	59	59	58	58	58
5	41	59	68	73	76	78	79	80	80	80	80	80
6	50	73	85	92	96	99	101	102	102	102	102	102
7	60	87	102	111	116	120	122	124	125	125	125	125
8	70	102	120	130	137	141	144	146	147	148	149	149

TABLE 6-S
CAS ATTENDANT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 2.00 SECONDS												
CAS AT-I-S REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	10	5	4	3	2	2	1	1	1	1	1	1
2	38	29	24	21	19	17	16	15	14	14	13	12
3	70	57	51	46	43	40	38	37	35	34	33	32
4	103	88	80	74	70	67	64	62	60	58	56	55
5	136	120	110	104	99	95	91	89	86	84	82	81
6	170	153	142	134	129	124	120	117	114	112	110	108
7	205	186	174	166	159	154	150	146	143	140	138	136
8	240	219	206	197	190	185	180	176	173	170	167	165
9	274	253	239	230	222	216	211	207	203	200	197	194
10	309	287	272	262	254	248	243	238	234	230	227	224
11	344	321	306	295	287	280	274	269	265	261	258	255
12	380	355	340	328	319	312	306	301	297	293	289	286
13	415	390	373	362	352	345	339	333	328	324	320	317
14	450	424	407	395	385	378	371	365	360	356	352	348
15	485	459	441	429	419	411	404	398	393	388	384	380

**TABLE 6-T
CAS ATTENDANT REQUIREMENTS**

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 4.00 SECONDS												
CAS ATTS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
ATTENDANT TRAFFIC (T) IN CCS												
1	15	10	7	5	4	4	3	3	2	2	2	2
2	47	38	33	29	26	24	23	21	20	19	18	17
3	81	70	62	51	54	51	48	46	45	43	42	40
4	116	103	94	88	84	80	77	74	72	70	68	67
5	151	136	127	120	115	110	107	104	101	99	97	95
6	186	170	160	153	141	142	138	134	131	129	126	124
7	221	205	194	186	179	174	169	166	162	159	156	154
8	257	240	228	219	212	206	202	197	194	190	188	185
9	292	214	262	253	245	239	234	230	226	222	219	216
10	328	309	296	281	279	212	261	262	258	254	251	248
11	362	344	331	321	313	306	300	295	291	281	283	280
12	396	380	366	355	347	340	334	328	324	319	316	312
13	430	415	401	390	381	373	361	362	357	352	348	345
14	463	450	435	424	415	407	401	395	390	385	381	378
15	496	485	410	459	449	441	435	429	423	419	415	411

TABLE 6-U
CAS ATTENDANT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 5.00 SECONDS												
CAS ATTS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	17	11	8	7	5	5	4	3	3	3	2	2
2	50	41	35	32	29	27	25	23	22	21	20	19
3	85	74	66	61	57	54	52	50	48	46	45	44
4	120	107	99	93	88	84	81	79	76	74	72	71
5	155	141	132	125	120	116	112	109	106	104	102	100
6	190	176	166	158	153	148	144	140	137	134	132	130
7	225	211	200	192	186	180	176	172	169	166	163	160
8	259	246	234	226	219	213	209	204	201	197	194	192
9	294	281	269	260	253	247	242	237	233	230	226	224
10	328	316	304	294	287	280	275	270	266	262	259	256
11	362	351	339	329	321	314	308	304	299	295	292	288
12	396	385	373	363	355	348	342	337	332	328	324	321
13	430	419	409	398	390	382	376	371	366	362	358	354
14	463	453	443	433	424	417	410	405	400	395	391	387
15	496	487	477	468	459	451	444	439	433	429	424	421

**TABLE 6-V
CAS ATTENDANT REQUIREMENTS**

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 6.00 SECONDS												
CAS ATTS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-35	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	19	13	10	8	6	5	5	4	4	3	3	3
2	53	44	38	34	31	29	27	26	24	23	22	21
3	87	77	70	65	61	57	55	53	51	49	48	46
4	123	111	103	97	92	88	85	82	80	78	76	74
5	157	145	136	130	124	120	116	113	110	108	106	104
6	191	180	170	163	157	153	148	145	142	139	136	134
7	225	215	205	197	191	186	181	177	174	171	168	166
8	259	249	240	231	225	219	214	210	206	203	200	197
9	294	283	274	266	259	253	248	243	239	236	233	230
10	328	317	308	300	293	287	281	277	272	269	265	262
11	362	351	342	334	327	321	315	310	306	302	298	295
12	396	385	376	368	361	355	349	344	340	335	332	328
13	430	419	410	402	395	389	384	378	373	369	365	362
14	463	453	443	435	429	423	417	412	407	403	399	395
15	496	487	477	469	462	456	451	446	441	437	433	429

TABLE 6-W
CAS ATTENDANT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER \leq 8.00 SECONDS												
CAS ATTS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	22	15	12	10	8	7	6	5	5	4	4	4
2	55	47	42	38	35	33	31	29	27	26	25	24
3	89	81	75	70	66	62	60	57	55	54	52	51
4	123	114	107	102	98	94	91	88	86	84	82	80
5	157	148	140	135	130	126	122	119	117	114	112	110
6	191	181	174	167	162	158	154	151	148	145	143	141
7	225	215	207	201	195	190	186	183	180	177	174	172
8	259	249	241	234	228	223	219	215	212	209	206	203
9	294	283	274	267	261	256	252	248	244	241	238	235
10	328	317	308	301	295	289	284	280	276	273	270	267
11	362	351	342	334	328	322	317	313	309	306	302	299
12	396	385	376	368	361	356	351	346	342	338	335	332
13	430	419	410	402	395	389	384	379	375	371	367	364
14	463	453	443	435	429	423	417	412	408	404	400	397
15	496	487	477	469	462	456	451	446	441	437	433	430

TABLE 6-X
CAS ATTENDANT REQUIREMENTS

GRADE OF SERVICE												
AVERAGE SPEED OF ANSWER ≤ 10.00 SECONDS												
CAS ATTS REQ'D	ATTENDANT WORK TIME (AWT) IN SECONDS											
	1-5	6-10	11-15	16-20	21-25	26-30	31-35	36-40	41-45	46-50	51-55	56-60
	ATTENDANT TRAFFIC (T) IN CCS											
1	22	17	13	11	9	8	7	6	6	5	5	4
2	55	48	43	39	36	34	32	30	29	27	26	25
3	89	81	75	70	66	63	61	58	56	55	53	52
4	123	114	107	102	98	94	91	88	86	84	82	80
5	157	148	140	135	130	126	122	119	117	114	112	110
6	191	181	174	167	162	158	154	151	148	145	143	141
7	225	215	207	201	195	190	186	183	180	177	174	172
8	259	249	241	234	228	223	219	215	212	209	206	203
9	294	283	274	267	261	256	252	248	244	241	238	235
10	328	317	308	301	295	289	284	280	276	273	270	267
11	362	351	342	334	328	322	317	313	309	306	302	299
12	396	385	376	368	361	356	351	346	342	338	335	332
13	430	419	410	402	395	389	384	379	375	371	367	364
14	463	453	443	435	429	423	417	412	408	404	400	397
15	496	487	477	469	462	456	451	446	441	437	433	430

SL-1*
BUSINESS COMMUNICATIONS SYSTEM
 CENTRALIZED ATTENDANT SERVICE (CAS)

RELEASE LINK TRUNK-DESCRIPTION, OPERATION, INSTALLATION

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553-2681-300 for operation of the feature. This section describes the installation and operation of **the QPC250/QPC70** RLT combination, as it applies to the CAS feature on SL-I. Figure I-1 illustrates a typical CAS configuration.

1.02 An RLT circuit consists of a trunk circuit at a remote private branch exchange (**PBX**) to be served by a CAS attendant, and an interfacing trunk circuit at the main PBX where the CAS attendant is located. The type of circuit pack that can be used at either the remote PBX or main PBX depends on the distance between the remote PBX and the main PBX.

1.03 The following trunk combinations may be used to establish an RLT circuit between a remote PBX and a main PBX.

- (a) A **QPC71 2-wire** E&M trunk at both main and remote location, with a repeater facility between the two.
- (b) A **QPC237 4-wire** E&M trunk at both the main and remote location, with a repeater facility between the two.
- (c) A **QPC250** RLT at the remote PBX and an interfacing **QPC70** Central Office (CO) trunk at the main PBX. This combination is limited to a maximum loop resistance of **1400** ohms.

1. GENERAL

1.01 The Centralized Attendant Service (CAS) feature allows customers with multiple locations to have attendant services provided from a single location. When the CAS feature is activated at a remote location, calls normally presented to the attendant at that location are presented to the CAS attendant over a Release Link Trunk (**RLT**). The CAS attendant then extends the call to the requested station at the remote location. See **553-2681-100** for a complete description of the CAS feature and

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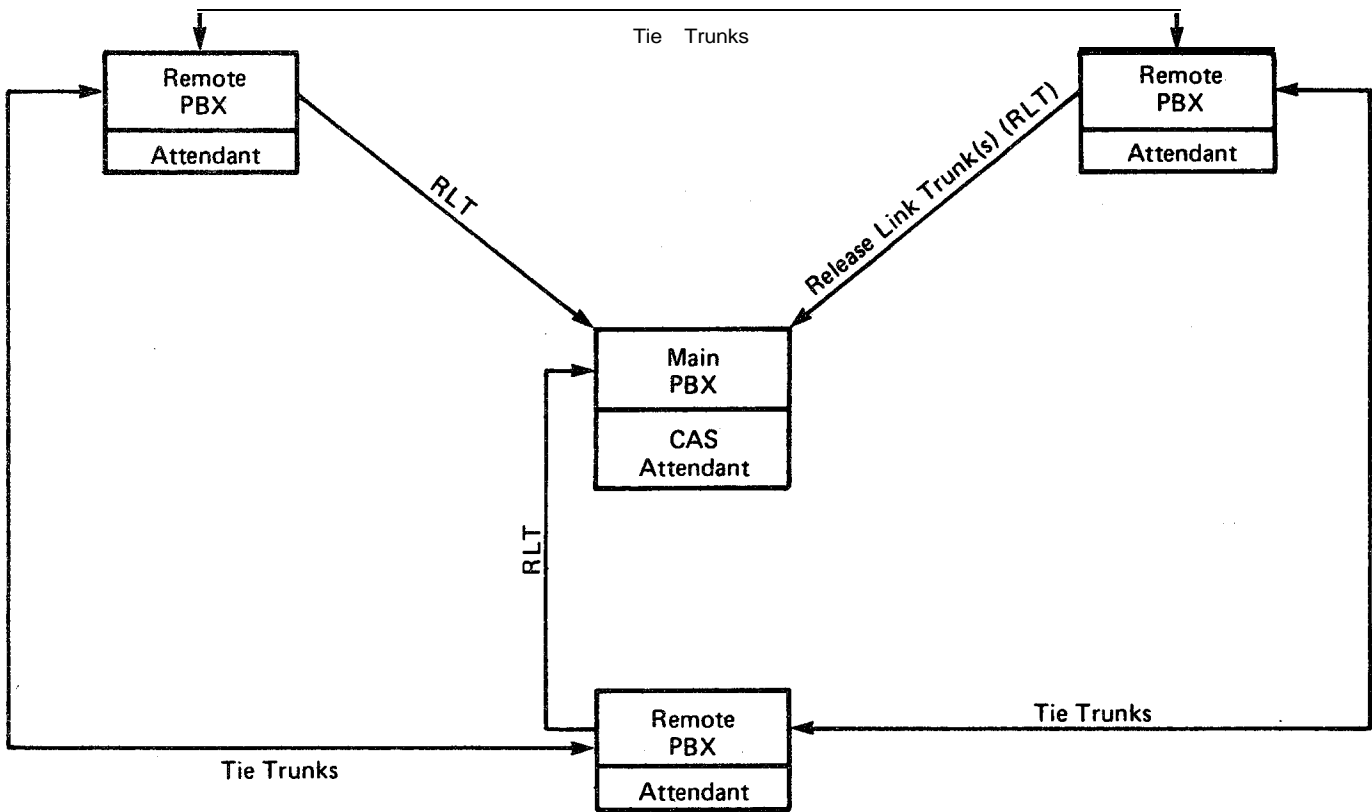


Fig. I-1 - Typical CAS Configuration

2. DESCRIPTION OF THE QPC250 RLT

2.01 A QPC250 RLT circuit pack contains two identical trunk circuits and common circuitry mounted on a printed circuit board. The pack can be inserted into any Peripheral Equipment (PE) shelf slot. Each circuit on the pack connects with the switching system and with the external apparatus through an 80-pin connector at the back of the pack and the cross-connect field. There are no option switches on the QPC250 RLT circuit pack.

FUNCTIONS

2.02 Each circuit on a QPC250 RLT circuit pack at a remote PBX interfaces with a circuit on a QPC70 CO trunk circuit pack at the main PBX to allow access to the CAS attendant. In addition, each circuit on the QPC250 RLT circuit pack performs the following functions:

- (a) terminates the loop tip and ring conductors with a balanced 900-ohm termination,
- (b) supervises the loop current to determine answer/disconnect status,
- (c) provides a dc battery supply from ground and -48 V through a 400-ohm battery feed resistance,
- (d) connects 20-Hz ringing current and other identifying tones to the loop,
- (e) isolates foreign potentials on the loop from portions of the transmission and signaling circuitry,
- (f) converts the 2-wire transmission path of the loop to a 4-wire transmission path,
- (g) provides analog-to-digital and digital-to-analog conversion of the transmission signals.

2.03 Table 2-A lists the characteristics of the circuits on a **QPC250** RLT pack.

TABLE 2-A
CHARACTERISTICS OF THE **QPC250** RLT CIRCUIT

CHARACTERISTICS	
Circuits per Pack	2.
Options	Nil.
Impedance	900 ohms.
Loop Limit	1400 ohms at nominal -48 V.
Leakage Resistance	30 000 ohms.
Ring Trip	During silent or ringing intervals.
Ringing Voltage	85-V rms , 20 Hz superimposed on -48 v dc.
Signaling	Ground start.
Supervision	Normal battery conditions are applied (-48 V to ring, ground to tip) when the interfacing CO trunk circuit is to be seized.
Power Input from Shelf	+10 V dc, ±6 V dc, -48 V dc, 85-V rms, 20 Hz, superimposed on -48 V dc.
Effective Loss	0.5 dB at 1020 Hz (pads in), -1.5 dB (pads out).
Insertion Loss	1 dB.
Answer Supervision	Does not reverse battery when the terminating end answers.
Disconnect Supervision	Does not momentarily open-circuit the tip and ring leads on disconnect.

3. OPERATION

A. Release Link Trunk Circuit at the Remote PBX

Idle State

3.01 The RLT circuit at the remote PBX is connected to a CO trunk circuit at the main PBX by tip and ring leads (Fig. 4-1). The RLT circuit provides open tip and ring leads during the idle state.

Outgoing Seizure

3.02 Upon seizure of the RLT by the remote PBX, the RLT provides ground on the tip lead, battery on the ring lead, and waits 256 ms.

(a) If the dc resistance across the tip and ring leads is greater than 20k ohms, the PBX provides steady ringing superimposed with -48 V battery through up to **220-ohms** of resistance on the ring lead.

(b) If the dc resistance across the tip and ring leads is less than 1300 ohms, then the CO trunk at the main PBX has not released from a previous call, and the RLT is locked out until the main PBX removes the loop. A trunk error message (**TRK 241**) is **printed** and another RLT circuit is used. See 553-2001-505 for an explanation of trunk error messages.

Note: The **ringing** supply at the remote PBX should generate ac ringing voltage between **75-** and **100-V rms** at a frequency of 20 Hz (**±3 Hz**).

Detecting Answer Supervision

3.03 The RLT detects a decrease in dc resistance across the tip and ring leads from at least 20k ohms to 1300 ohms or less, as an off-hook signal **from** the main PBX equipment. If the off-hook signal persists for at least 256 ms, the signal is an answer supervision. The RLT trips ringing within **100 ms** when it detects the answer signal. Ringing can be tripped **during** both the silent and ringing intervals.

Talking State

3.04 **After** answer, and until disconnect, the RLT provides ground through up to 425 ohms of resistance on the tip lead, and -48 V battery through up to 425 ohms of resistance on the ring lead.

Flash Signal

3.05 The CAS attendant at the main location controls the **RLT through** operation of the console signal remote (SPG REM) key, or release destination (RLS **DEST**) key. The operation of this key sends a timed flash signal to the RLT. There are only two RLT functions the CAS attendant is required to control by flashing:

- (1) request for a **DIGITONE*** receiver to complete the call for the calling party, and
- (2) cancellation of a prior attendant request (release destination).

3.06 The **RLT** recognizes an increase in the dc resistance across the tip and ring leads from 1300 ohms or less to 20k ohms or more as an **on-hook** signal. The on-hook should be timed for 200 ms to 1.0 second. If the conductor loop resistance changes back to low resistance (off-hook) in this interval, the RLT interprets that as a flash signal from the CAS attendant.

3.07 When the RLT has been seized for an incoming CO call, dial "Q", intercept call, or a hold recall, and after the CAS attendant has talked to the calling party, the detection of the **first** flash signal prompts the connection of a **Digitone** receiver to the **RLT**, and splits the calling party away from the RLT.

3.08 Dial tone is sent to the CAS attendant when the **Digitone** receiver is prepared to receive address signals. Dial tone is removed within 50 ms of the start of dialing the first address character. If the RLT detects a subsequent flash signal while connected to a receiver; i.e., attendant flash to clear the receiver, it disconnects the receiver and returns to its initial state; i.e., connected to the calling party.

3.09 When the receiver has translated the dialed address characters, **different confirmation** tones are sent to the CAS attendant, depending on the status of the called station.

Disconnect

3.10 The RLT recognizes and times an increase in the dc resistance across the tip and ring leads from **1300** ohms or less to 20k ohms or more as an, on-hook from the CAS attendant. If the on-hook lasts longer than 1 second (this time is programmable, see **553-2001-220**), the RLT interprets the change as a disconnect.

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3.1 I **After** interpreting disconnect from the attendant, the RLT opens the tip and ring leads to the interfacing CO trunk at the main PBX, i.e., sends a disconnect signal to the main PBX. To prevent the RLT from being seized from a new call before the CO trunk at the main PBX has released, the RLT provides a minimum of a 1-second busy guard (this time is programmable, see 553-2001-220).

B. CO Trunk Interface at the Main PBX

Idle State

3.12 There is a resistance greater than **20k** ohms across the tip and ring leads at the CO trunk interface during the idle state.

Incoming Seizure

3.13 The CO trunk at the main PBX detects seizure by an RLT at a remote PBX when the RLT provides simultaneously:

- (a) a ground through **1.2k** ohms of resistance or less on the tip lead, and
- (b) ringing with superimposed -48 V battery on the ring lead through a resistance of **1.2k** ohms, or less.

The maximum delay of ringing must be less than **5** seconds after ground and battery detection, and the ringing must last more than 640 ms. If any one of the above conditions is not satisfied, the call is not presented to the CAS attendant.

Disconnect Before the Attendant Answers

3.14 Before the attendant answers the call, tone ringing should be 2 seconds on and 4 seconds off. If there is no ringing for longer than a programmable timer, the call is withdrawn from presentation to the attendant

CAS Attendant Answers

3.15 When the CAS attendant answers, a resistance of 800 ohms or less is applied across the tip and ring conductors at the CO trunk. A 2-way transmission path between the RLT and the CAS attendant is established within 160 ms to avoid clipping initial speech. The circuit maintains the low resistance across the tip and ring leads until the RLT is flashed or disconnected by the attendant.

Flash Signal

3.16 When the CAS attendant presses the signal remote (SIG REM) key on the console, a timed flash (on-hook) signal is sent to the remote

PBX. The flash signal changes the resistance across the tip and ring leads from low (off-hook) to **high (on-hook)** resistance for a **timed interval**. The flash signal should be **600 ms (± 200 ms)** with the preferred signal being 512 ms.

3.17 After the CAS attendant has pressed the SIG REM key, a **Digitone receiver** at the remote end is connected to the RLT and dial tone is sent to the attendant as **confirmation** that an idle receiver has been connected. Digits dialed **after** receiving dial tone are outpulsed as soon as an-outpulsor is available.

3.18 When the CAS attendant wishes to release from a prior request; i.e., a call extended to a busy station, the release destination (**RLS DEST**) key on the console is pressed. Operation of this key sends a timed flash to the remote PBX, and connects the attendant to the original calling party.

Disconnect

3.19 The CO trunk at the main PBX interprets removal of ground from the tip lead and **-48 V** battery from the ring lead that persists for more than 1 second, as a disconnect signal from the RLT. **After interpreting** disconnect from the remote PBX, the CO trunk disconnects from the CAS attendant, and removes the loop across the tip and ring leads which acts as a disconnect supervision.

3.20 When the CAS attendant disconnects, the resistance of **800** ohms or less applied to the tip and ring leads at the CO trunk changes to **20k** ohms or more which is an on-hook signal to the RLT at the remote PBX.

4. INSTALLATION AND CONNECTIONS

4.01 The QPC250 RLT circuit pack can be inserted into any trunk pack position on a PE shelf. The circuit pack handling precautions of 553-2001-205 must be observed when installing trunk packs in BE shelves. Installation procedures for trunk circuit packs are provided in 553-2yy1-210.

4.02 **When** telephone lines connected to the trunk circuit are exposed to foreign voltages by direct contact or induction; i.e., power line crosses or lightning, protection devices must be installed at the customer's premises. These devices must be capable of providing a path to ground from the tip and ring leads for foreign voltages that exceed a **600 V** peak.

4.03 Typical cross-connections between a QPC250 RLT trunk at a remote PBX and a QPC70 CO trunk at the main PBX are shown in Fig. 4- 1.

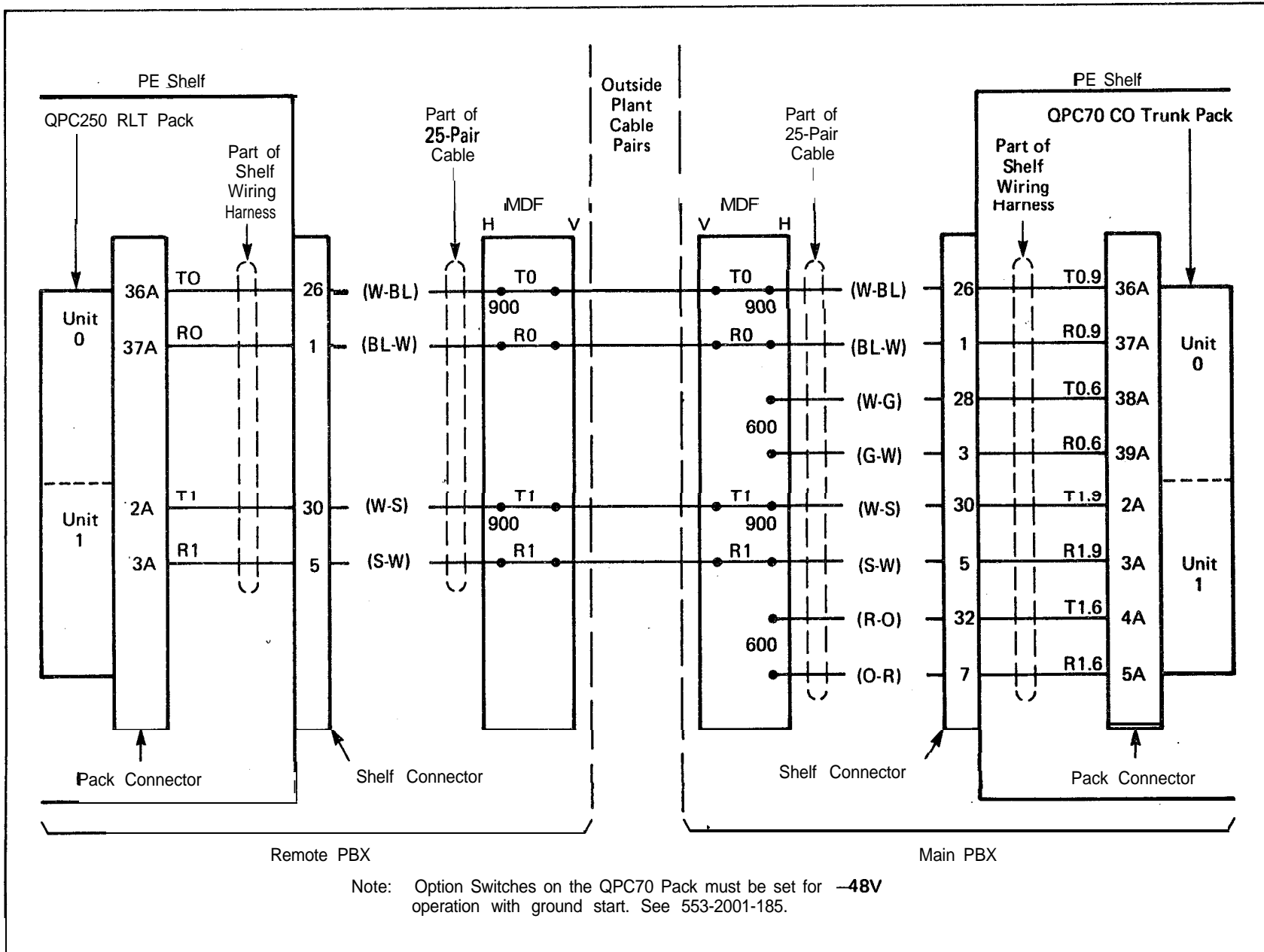


Fig. 4-1 - QPC250 RLT Cross-Connections to QPC70 CO Trunk

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BUSINESS COMMUNICATIONS SYSTEM
 CENTRALIZED ATTENDANT SERVICE
 FEATURE OPERATION **AND** TESTS

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calls normally handled by an attendant at a remote location, are presented to a CAS attendant at the main location. The CAS attendant then extends the call to **the** desired station at the remote location. Refer to 553-268 1- 100 for a complete description of features.

1.02 This section pertains only to calls directly related to the CAS feature. If the CAS feature is not equipped, or not activated, calls to the CAS attendant or to an attendant at any of the remote locations are handled in the normal manner (**553-2001-300**).

1.03 This section describes tests to verify the proper operation of the CAS feature at the main private branch exchange (PBX), and at any remote PBX arranged to be served by a CAS attendant (**Fig. 1-1**). The operation of the CAS feature on the GAS attendant console is also detailed. All tests outlined in this section are based on the assumption that all locations (main and remote) can perform normal call-processing, and are operating correctly.

1. GENERAL
 1 .01 The Centralized Attendant Service (CAS) feature allows customers with multiple locations to have attendant services provided from a single location. When the CAS feature is active, any

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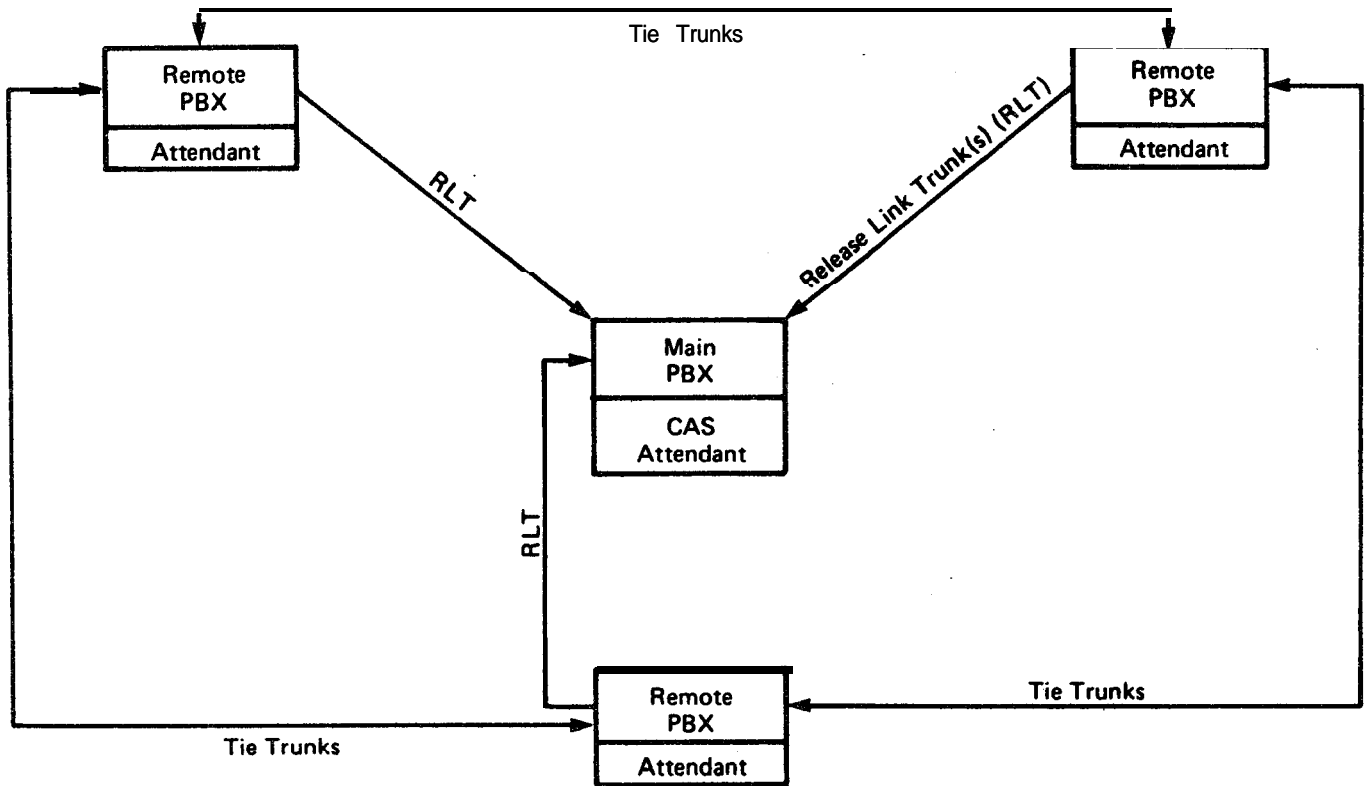


Fig. 1-1 - Typical CAS Configuration

2. TESTING THE CAS FEATURE

2.01 The tests outlined in Chart 2- 1 must be conducted when the CAS feature is installed at a remote PBX to be served by a CAS attendant. These procedures should also be used as a guide to locating a fault in the feature.

2.02 Chart 2-2 describes tests which verify that the CAS attendant can receive and handle calls from any PBX arranged for the CAS feature.

CHART 2-1
TESTING THE CAS FEATURE AT A REMOTE LOCATION

STEP	PROCEDURE
1	<p>Verify that all CAS data is correct in the system memory. Information in the following data blocks is affected by the CAS feature:</p> <ul style="list-style-type: none"> (a) customer data block, (b) SL-1 set data block, if an attendant console is not equipped, (c) trunk data block, (d) route data block. <p>Refer to 553-2001-220 or 553-2yy1-310 and 553-2001-221 or 553-2yy1-311 for data input and retrieval information.</p>
2	<p>Ensure that the QPC250 Release Link Trunks (RLT) are inserted in the correct Peripheral Equipment (PE) shelf slot, enabled, and properly connected at the cross-connect field. Refer to 553-2yy1-215 or 553-2yy1-210.</p>
3	<p>Verify that QPC251 Tone and Digit Switches (TDS) are equipped in all network shelves.</p>
4	<p>From any station, dial the Directory Number (DN) of an RLT. If the RLT is busy, busy tone is heard. If the RLT is idle:</p> <ul style="list-style-type: none"> (a) ringback tone is heard, (b) when the CAS attendant answers, ringback tone stops and a voice path is established. <p>Verify that a 2-way conversation can be carried out. Terminate the call.</p>
5	<p>Repeat Step 4 for each equipped RLT.</p>
6	<p>From any station, dial the DN assigned for the local attendant. This call is presented to the local attendant as a normal dial "0" (zero) call. See Note in Step 7.</p>
7	<p>From any station, dial the DN assigned for the CAS attendant. This call is presented to the local attendant as a normal dial "0" call. See Note.</p> <p style="margin-left: 40px;">Note: If this location is in night service, calls dialed to either the local attendant or the CAS attendant are presented to the night number, or trunk answer from any station (TAFAS) at this location, as applicable.</p>
8	<p>Verify that the CAS key (above the LPK 5 key) on the local attendant console is so designated. If a console is not equipped, this key can be assigned on up to ten SL-1 sets.</p>
9	<p>Momentarily press the CAS key. The associated LED lights to indicate that the CAS feature has been activated. All calls normally presented to the local attendant are now directed to the CAS attendant except as in Step 10.</p>

Chart Continued

CHART 2-1 Continued
TESTING THE CAS FEATURE AT A REMOTE LOCATION

STEP	PROCEDURE
10	From any station, dial the DN assigned for the local attendant. This call is presented to the local attendant as a normal dial "0" call. See Note in Step 7.
11	From any station, dial the DN assigned for the CAS attendant. This call is presented to the CAS attendant, preceded by an identifying tone (two, 100-ms bursts of 440-Hz tone).
12	From any station, dial the access code for an outside line; i.e., 9. When the second dial tone is heard (depends on the tie route and may not be provided), dial the LDN for your location. This call is directed to the CAS attendant.
13	When the CAS attendant answers, request that your call be placed on silent hold. When the specified timeout period has elapsed, your call is directed back to the CAS attendant. The timeout period is service-changeable. See 553-2001-221 or 553-2yy1-311 .
14	Request the CAS attendant to extend your call to a busy station at your location. <ul style="list-style-type: none"> (a) If another call is not already camped-on to the busy station, your call is automatically camped-on to that station. When the busy station becomes idle, your call is presented automatically to that station. (b) If the busy station does not become idle within the specified timeout period, your call is presented to the CAS attendant again.
15	If in Step 14(a) your call was completed to a station that was busy, request that station user to transfer your call to the attendant. This directs your call to the CAS attendant. Proceed to Step 17.
16	If in Step 14(b) the timeout period elapsed, and your call was directed back to the CAS attendant, proceed to Step 17.
17	Disconnect from the call; i.e., replace the handset or press the RLS key.
18	Momentarily press the CAS key on the local attendant console (or SL-1 set). The associated LED goes out, and all calls requiring attendant assistance are presented to the local attendant.
19	Proceed to the CAS attendant location and conduct the tests of Chart 2-2.

CHART 2-2
TESTING THE CAS FEATURE AT THE CAS ATTENDANT LOCATION

STEP	PROCEDURE
1	<p>Verify that all CAS data is correct in the system memory. Information in the following data blocks is affected by the CAS feature:</p> <ul style="list-style-type: none"> (a) customer data block, (b) trunk data block, (c) route data block. <p>Refer to 553-2001-220 or 553-2yy1-310 and 553-2001-221 or 553-2yy1-311 for data input and retrieval information.</p>
2	<p>Verify that the trunk circuit packs (associated with the RLT from each remote location) are in the proper PE shelf slot, enabled, and properly connected at the cross-connect field. See 553-2yy1-215 or 553-2yy1-210.</p>
3	<p>Ensure that the CAS attendant console is equipped with:</p> <ul style="list-style-type: none"> (a) an RLT X ICI key for each group of RLT from remote locations, where X designates a particular remote location, (b) a SIGNAL REMOTE (SIG REM) key (above the LPK 5 key), (c) a RELEASE DESTINATION (RLS DEST) key.
4	<p>From any station, dial the access code for a tie trunk to a remote location. When the second dial tone (optional, depends on tie route) is heard, dial the DN for an RLT. This call is presented to the CAS attendant. Ensure that the call is presented on the correct ICI key, and, if the console is equipped with a digit display, the correct access code and route member number of the RLT is displayed. When the CAS attendant answers, two 100-ms bursts of 440-Hz tone are heard by the attendant (only if defined in customer overlay program 15) prior to a voice path being established. Ensure that a 2-way conversation can be carried out.</p> <p>Note: Calls active on a CAS attendant console which are presented to a remote station (via RLT) are considered as incoming external calls rather than attendant calls for purposes of digit display. This is done because:</p> <ul style="list-style-type: none"> (1) the identity of the CAS attendant cannot be determined from the remote end, and (2) the display of the RLT identification is sufficient to indicate that the call is connected to an attendant; it also enables the user to identify the specific RLT used, in case of a fault.
5	<p>Repeat Step 4 for each RLT at each remote location.</p>
6	<p>Call the attendant at a remote location; i.e., dial the access code for an outside line, then dial the LDN for the remote location. Request the attendant at the remote location to activate the CAS feature by momentarily pressing the CAS key.</p>

Chart Continued

CHART 2-2 Continued
 TESTING THE CAS FEATURE AT THE CAS ATTENDANT LOCATION

STEP PROCEDURE

7 Occupy a CAS attendant position, and direct all CAS-related calls as described in Charts 2-3 through 2-8. The following symbols are used in the charts to signify the state described:

- Ⓜ Operate key momentarily
- Lamp is lit steadily
- Ⓜ₆₀ Lamp ON/OFF at 60 ipm
- _M Lamp lit steadily after momentary operation of key.

Verify that the appropriate identifying tone is received for each type of call. The possible tones are :

CALL	CONDITION	TONE
(a)	Camp-on confirmation, and camp-on' recall	100-ms burst of 440-Hz tone
(b)	Silent hold confirmation , and silent hold recall	four through six, 50-ms bursts of 440-Hz tone
(c)	Timed reminder recall	2-3 seconds of audible ringback tone
(d)	Transfer to attendant, and attendant recall	100 ms burst of 440-Hz tone, 100 ms silence, 100-ms burst of 440-Hz tone.

**CHART 2-3
ANSWERING AN INCOMING CALL FROM A REMOTE LOCATION**

STEP	ACTION	OPERATING KEYS AND INDICATORS																REMARKS									
		ICI	RLS	LPK	SRC	DEST	POS BSY	NITE	HOLD	CONF	RLS DEST	RLS SRC	EXC DEST	EXC SRC	SIG DEST	SIG SRC	DO-NOT-DISTURB		BG IN	BSY VER	AUTO DIAL	SPEED CALL	TIME-OF-DAY	DISPLAY DATE	SIG REMOTE		
1	Start Condition Call Presented to the Attendant	<input type="radio"/>	<input type="radio"/>		<input checked="" type="radio"/>																						<p>Console idle, loops free</p> <p>Tone ringing is heard. If console is equipped with a digit display, the access code and member number of the RLT are displayed.</p> <p>Attendant hears two, 100-ms bursts of 440 Hz tone before the talk path is established with the calling party</p>
2	Attendant Answers by Pressing the LPK Key Associated with the Flashing SRC Lamp	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>																						

CHART 2-5
EXTENDING A CALL TO A BUSY STATION WITH CAMP-ON

STEP	ACTION	OPERATING KEYS AND INDICATORS															REMARKS										
		ICI	RLS	LPK	SRC	DEST	POS BSY	NITE	HOLD	CONF	RLS DEST	RLS SRC	EXC DEST	EXC SRC	SIG DEST	SIG SRC		DO-NOT-DISTURB	BG IN	BSY VER	AUTO DIAL	SPEED CALL	TIME-OF-DAY	DISPLAY DATE	SIG REMOTE		
	Start Condition	<input type="radio"/>		<input type="radio"/>	<input type="radio"/>																						2-Way conversation between attendant and calling party
1	Press SIG REM Key	<input type="radio"/>		<input type="radio"/>	<input type="radio"/>																					Ⓜ Calling party is excluded, attendant hears dial tone	
2	Dial The DN of the Busy Station	<input type="radio"/>		<input type="radio"/>	<input type="radio"/>	Ⓜ																				No busy tone is heard. Camp-on (100-ms burst of 440 Hz tone) is heard at the busy station. Dialed number is displayed. Calling party is camped-on to the busy station. Attendant is recalled if busy station does not become idle in 30 sec. If busy tone is heard, the busy station has a call camped-on.	
3	Attendant Releases		<input type="radio"/>																							Console idle	
4	Recall	<input type="radio"/>			Ⓜ																					Tone Ringing is heard	
5	Attendant Answers	<input type="radio"/>		<input type="radio"/>	<input type="radio"/>																					Attendant hears 100-ms burst of 440 Hz tone before talk path with calling party is established.	



CHART 2-6
PLACING A CALL ON SILENT HOLD

STEP	ACTION	OPERATING KEYS AND INDICATORS															REMARKS									
		ICI	RLS	LPK	SRC	DEST	POS BSY	NITE	HOLD	CONF	RLS DEST	RLS SRC	EXC DEST	EXC SRC	SIG DEST	SIG SRC		DO-NOT-DISTURB	BG IN	BSY VER	AUTO DIAL	SPEED CALL	TIME-OF-DAY	DISPLAY DATE	SIG REM OTE	
1	Start Condition	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																						2-Way conversation between attendant and calling party
2	Press SIG REM Key	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																				<input checked="" type="radio"/> M	Calling party is excluded, attendant hears dial tone
3	Dial Silent Hold DN	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																					Hold confirmation tone (four to six, 50-ms bursts of 440-Hz tone) is heard by the attendant. Calling party is placed on hold. Attendant is recalled when hold timer has elapsed.
4	Attendant Releases Recall	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/> M																					Console idle
5	Attendant Answers	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																					Tone ringing is heard Attendant hears hold confirmation tone before being connected to the calling party

CHART 2-7
REMOTE STATION DIALS THE CAS ATTENDANT DN

STEP	ACTION	OPERATING KEYS AND INDICATORS																REMARKS									
		ICI	RLS	LPK	SRC	DEST	POS BSY	NITE	HOLD	CONF	RLS DEST	RLS SRC	EXC DEST	EXC SRC	SIG DEST	SIG SRC	DO-NOT-DISTURB		BG IN	BSY VER	AUTO DIAL	SPEED CALL	TIME-OF-DAY	DISPLAY DATE	SIG REMOTE		
1	Start Condition Incoming Call is Presented to the CAS attendant.	<input type="radio"/>	<input type="radio"/>		<input checked="" type="radio"/>																						Console idle loops free
2	Press the LPK key associated with the flashing SRC lamp.	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/> M	<input type="radio"/>																					Tone ringing is heard. If console is equipped with digit display, the access code and member number of the RLT being used are displayed. Tone ringing stops. attendant hears the appropriate identifying tone. 2-way conversation between attendant and calling party.	

**CHART 2-8
REMOTE LOCATION TRANSFERS CALL TO CAS ATTENDANT**

STEP	ACTION	OPERATING KEYS AND INDICATORS																REMARKS									
		ICI	RLS	LPK	SRC	DEST	POS BSY	NITE	HOLD	CONF	RLS DEST	RLS SRC	EXC DEST	EXC SRC	SIG DEST	SIG SRC	DO-NOT-DISTURB		BG IN	BSY VER	AUTO DIAL	SPEED CALL	TIME-OF-DAY	DISPLAY DATE	SIG REMOTE		
1	Start Condition	<input type="radio"/>																									Console idle, loops free
2	Call Presented to the Attendant	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>	<input type="radio"/>																					Tone ringing is heard	
1	Attendant Answers	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>																					Attendant hears two, 100-ms bursts of 440 Hz tone and talk path is established	
1	To Transfer the Call to Another Station at the Remote Location	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																					Called station is disconnected. Attendant and calling party are connected	
2	Press RLS DEST Key	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																					Dial tone is heard, calling party is excluded.	
3	Press SIG REM Key	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																					Attendant hears ringback tone	
4	Dial Station DN	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																					Console idle, calling party hears ringback tone.	
4	Attendant Releases	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>																						

BUSINESS COMMUNICATIONS SYSTEM

SL-1*

MESSAGE CENTER DESCRIPTION AND OPERATION

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Reason for Reissue: This practice is reissued to incorporate information concerning the **QSY22** message waiting power supply, and the Audible Message Waiting (AMW) feature for Generic **X11** Release 2 and subsequent. Due to the extensive nature of the changes, bracketing arrows and arrow heads indicating changes are omitted.

* SL-1 is a trademark of Northern
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1. INTRODUCTION

1.01 In certain PBX applications, it is convenient to have a centralized facility for the taking, holding and retrieval of messages. The Message Center feature of SL-1 allows one or more **SL-1** sets or attendant consoles to be designated a “message center”. Calls are forwarded to this message center, which takes and holds messages and activates and deactivates the message waiting indication at a user telephone.

1.02 Message Waiting indication can be provided by message waiting lamps at the user station. However, for Generics **X1** 1 Release 2 and subsequent, the message waiting indication at NE-500 or NE-2500 type user stations can take the form of Audible Message Waiting (AMW), which utilizes interrupted dial tone instead of a message waiting lamp. This reduces the power consumption and hardware requirements of an SL-1 system equipped with the Message Center feature.

1.03 This practice describes the installation and operation of the Message Center hardware, as well as the equipment requirements of an SL-1 system that incorporates Message Center (station sets, service change, hardware, etc.).

1.04 *Reference Documentation.* The following information for Message Center can be found in the indicated NTP:

- 553-2YY1-105: Feature Availability
- 553-2001-110: SL-1 Sets Description
- 553-2YY 1-1 50: Equipment Identification and Ordering
- | 553-2001-1 51: Memory and Real Time Requirements
- | 553-2001-183: 500/2500 Line Circuit Description
- | 553-2YY 1-220: Feature Implementation
- 553-2YY1-221: Service Change
- 553-2YY1-300: Attendent Console Operation
- 553-2YY1-305: Sk-1 Set Operation
- 553-2671-100: Automatic Call Distribution (ACD)
- 553-2781-100: Integrated Messaging System (IMS).



2. FUNCTIONAL DESCRIPTION

INCOMING CALLS TO THE MESSAGE CENTER

2.01 Calls coming in to a message center are of two basic types: direct and indirect. This classification refers more to how the calls originate than to how they are handled by a message center operator.

Direct Message Calls

2.02 The message center can also be accessed by a 'direct message call' which can take any of the following forms:

- An attendant extends a call to the message center.
 - l A 500/2500-type starter dials the message center DN.
 - l An SL-1 type station dials the message center DN or depresses the MESSAGE WAITING key.

Indirect Message Calls

2.03 A telephone call can be rerouted to the message center if it is not answered at the original terminating station for a specified length of time. Such a call is referred to as an 'indirect message call' and can take any of the following forms:

- *Non-Direct Inward Dial (DID) Calls.* A non-DID call encounters a no-answer condition so that Call Forward No Answer (CFNA) is invoked, rerouting it to the message center.
- *DID Calls.* A DID call encounters CFNA or (optionally) Call Forward Busy (CFB), and is rerouted to the message center.
- *Hunting and Call Forward (All Calls).* The Message Center Directory Number (DN) can be specified as a Hunt or Call Forward DN for any station.

MESSAGE CENTER OPERATION

2.04 A message center can be organized in one of three ways:

- an Automatic Call Distribution (ACD) message center with SL-1 sets (IMS optional)
 - l a DN-type message center with SL-1 sets; in this case, IMS is allowed only if the message center has only one set
- an attendant-type message center with attendant consoles.

ACD-Type Message Center

2.05 An ACD message center consists of up to 240 SL-1 sets (with 16-digit display) arranged as an ACD, the message center DN (MCDN) being the ACD-DN (see 553-2671-100 for a general description of ACD). Calls to this type of message center (whether direct or indirect) are placed in an 'incoming calls' queue in their order of arrival. The message center operators are placed in an 'available agents' queue in the order they become available. The two queues are matched to ensure an equitable distribution of incoming calls among operators. If all operators are busy, incoming message calls are placed in a queue to wait for a free operator.

2.06 When an indirect message call is presented to a message center operator's SL-1 set, the MSG IN-CALLS key flashes, and the calling station DN and called station DN are displayed on the digit display. If the call was originated from a trunk, the route and member number of the trunk are shown instead of the calling station DN. The MSG INDIC lamp reflects the condition of the called station's message waiting lamp (or interrupted dial tone, if AMW is used in place of a message waiting lamp) as* follows:

- *Steadily Lit*: The message waiting indication at the called station is inactive.
- *Flashing (60 ipm)*: The message waiting indication at the called station is active.
- ! *Slow-Flashing (30 ipm)*: The message waiting lamp at the called station is either disabled or not equipped. This state does not occur if message waiting indication is by interrupted dial tone.

2.07 To answer the call, the message Center operator presses the MSG IN-CALLS key. After this point, the operator can release or transfer the call without leaving a message if required.

2.08 To leave a message, the operator first verifies that the number shown on the digit display is indeed the number at which the user wishes to leave a message. The operator takes down the message from the caller. If the caller wants to leave a message at the number shown on the digit display, the operator presses the MSG INDIC key to activate message waiting indication at the terminating station.

2.09 To terminate the sequence, the operator presses the RLS key. If the operator does not wish to rejoin the available agents queue, the NOT READY key or some other feature key can be depressed; otherwise the operator is presented with the next call.

2.10 If the terminating DN shown on the digit display is not the DN at which the message is to be left, the operator can press the RLS key to release the call, then manually access the required DN and activate its message waiting indication.

DN-Type Message
Center

2.11 A DN-type message center consists of one or more 16-digit display SL-1 sets sharing a DN which is designated the message center DN. This arrangement is available only on Generic X09 and subsequent. An incoming call to the message center is presented to all message center stations at once. The first operator to press the flashing MC-DN key is connected to the call. If all operators are busy, incoming message calls get whatever treatment is specified for a busy DN.

2.12 Upon pressing the flashing MC-DN key, the message center operator is connected to the calling party. The digit display shows the originating DN (or trunk route and member number) and terminating DN, and the MSG INDIC lamp shows the state of the called station's message waiting lamp (or interrupted dial tone) in the same way as for the ACD-type message center.

2.13 The procedures for activating a message waiting indication at a user station are the same as for an ACD-type message center.

Attendant-Type Message Center

2.14 An attendant-type message center consists of up to 15 attendant consoles in an **SL-1** system which handle messages in addition to their normal functions as attendants. Incoming message calls can access an attendant type message center in one of two ways: a phantom message center DN or normal attendant access.

2.15 *Phantom Message Center DN.* This method requires the presence of the ACD package in the SL-1 system. The ACD-DN of the ACD feature has no agents assigned to it. Any message calls coming into it are automatically transferred to the attendant and appear on a MSG CENTER Incoming Call Indicator (ICI) key. The message center DN established in this way is thus a 'phantom' message center DN. (See 553-2YY1-220 and 553-2YY1-221 for more information.)

2.16 When a message call appears at an attendant console via a message center DN, the MSG CENTER ICI lamp flashes, and the digit display shows the originating DN (or trunk route and member number) and terminating DN. The MSG INDIC lamp on the attendant console shows the status of the terminating party's message waiting indication in the same way as for an ACD-type message center. Upon pressing a free LPK key or the flashing MSG CENTER ICI key, the attendant is connected to the incoming call. The procedure for activating a message indication at a user station are the same as for an ACD-type message center.

2.17 *Normal Attendant Access.* If a phantom message center DN has not been established on an attendant console ICI key, the attendant must determine verbally that this is a message call. In this case the MSG INDIC lamp does not show the state of the user's message waiting indication. After the call is released, the attendant can provide message waiting indication by directly accessing the user's station.

MESSAGE RETRIEVAL CALLS

2.18 Message indication at a user's station set can take one of the following forms:

- **SL-1 Sets With a Message Waiting Key/Lamp Pair.** The flashing lamp associated with the MSG WAITING key indicates that a message is waiting at the message center.
- **500/2500 Sets With a Message Waiting Lamp.** A flashing message waiting lamp on the station set indicates that a message is waiting at the message center.
- 1 **Sets Without a Message Waiting Lamp.** As an option in Generic XI 1 Release 2, message-waiting indication can be provided by AMW (a 120 ipm interrupted dial tone) instead of a message waiting lamp or key/lamp pair when the user goes off-hook at the station. A faulty message waiting lamp does not result in message waiting indication by interrupted dial tone; in this case, it is up to the user to call the message center to see if there is a message. Station sets (**SL-1** or **500/2500**) equipped with neither message waiting lamps nor AMW have no visual indication that a message is waiting. It is up to the user to call the message center to see if a message is waiting.

2.19 Regardless of whether message indication is by lamp or AMW, the user can still make calls and operate features normally. The message indication remains active until cancelled by a message center operator or by the system during a night routine.

Placing a Message
Retrieval Call

2.20 *SL-1 Sets*. The user goes off-hook and presses the MESSAGE WAITING key. The digit display shows the message center DN. The message center answers and gives the user the message, then extinguishes the user's message indication lamp. The user then goes on-hook. The user can also access the message center by going off-hook and pressing the MESSAGE WAITING key even if the associated lamp is not flashing.

2.21 If message indication is by AMW, the user goes off-hook and dials the message center DN. The message center operator cancels the AMW in the same way as a message waiting lamp.

2.22 *500/2500 Sets*. Whether or not a message indication lamp is provided with the set, the user goes off-hook and dials the message center DN. The message center answers and gives the user the message, then extinguishes the user's message indication lamp. If message indication was provided by interrupted dial tone, this is cancelled by the message center operator in the same way as a message waiting lamp. The user then goes on-hook.

Answering a Message
Retrieval Call at the
Message Center

2.23 *ACD-Type Message Center*. The message retrieval call is presented on the MSG IN-CALLS key of a message center SL-1 set, and the digit display shows the DN of the calling station. The MSG CANC lamp reflects the state of the caller's message waiting indication as follows:

- 1 *Steadily Lit*: The message waiting indication at the calling station is inactive.
- 1 *Flashing (60 ipm)*: The message waiting indication at the calling station is active.
- 1 *Slow-Flashing (?O ipm)*: The message waiting lamp at the calling station is either disabled or not equipped. This state does not occur if message waiting indication is by interrupted dial tone.

2.24 To answer the call, the Message Center operator presses the MSG IN-CALLS key. If the message center is equipped with Call Forcing, the call is answered automatically. The operator gives the caller the message. If the MSG CANC lamp is flashing, the operator presses the MSG CANC key to de-activate the caller's message waiting indication. The operator then presses the RLS key to release the call.

2.25 *DN-Type Message Center*. The message retrieval call is presented on the MC-DN key of a message center SL-1 set. The message center operator answers the call by pressing the MC-DN key. The digit display shows the calling DN and the MSG CANC lamp indicates the state of the calling station's message waiting indication in the same way as for an ACD-type message center. The procedures for deactivating message waiting indication are the same as for an ACD-type message center.

2.26 *Attendant-Type Message Center*. The presentation of a direct message call to an attendant console depends on whether a phantom message center DN has been assigned to an ICI key or the station user must dial the SL-1 attendant as in a normal attendant call.

- (a) *Message Center DN Access.* In this case, the phantom message center ACD-DN is assigned to overflow to the MSG CENTER ICI key on the attendant console. When a direct message center call comes into the console, the MSG CENTER ICI key flashes. To answer the call, the message center operator presses the MSG CENTER ICI key or a free LPK key. The procedures for deactivating a message waiting indicator are the same as for an ACD-type message center.
- (b) *Normal Attendant Access.* If a phantom message center DN has not been established on an attendant console ICI key, the attendant must determine verbally that this is a message retrieval call. During the call, the MSG CANCEL key does not show the state of the user's message waiting lamp. After the call is released the attendant must deactivate message waiting indication directly outlined in 2.31.

DIRECT MESSAGE WAITING LAMP CONTROL

2.27 The message center operator must directly access a user station to activate or de-activate message waiting indication if:

- | the user station is calling an attendant message center by direct access
- the user station is not currently in contact with the message center.

2.28 The message center operator must first 'release' or place on 'hold' all calls; in an ACD-type message center, the NOT READY key must be activated. The operator then decides whether the station user's message waiting indication is to be activated (if not already activated) or canceled (if not already canceled).

Query and/or Activate

2.29 The operator presses the MSG INDIC key, and the associated lamp lights. The operator then dials the user station DN: No **ringback** is heard, but the state of the MSG INDIC lamp reflects the state of the user's message waiting lamp (or interrupted dial tone) as follows:

- (a) *Steadily Lit:* The user's message waiting indication is inactive.
- (b) *Flashing (60 ipm):* The user's message waiting indication is active.
- (c) *Slow-Flashing (30 ipm):* The user's message waiting lamp is either disabled or not equipped. This state does not occur if message waiting indication is by interrupted dial tone.

2.30 At this point, the operator can change the state of the user's message waiting indication from inactive to active by pressing the MSG INDIC key, or the indication can be left as is. The connection is released when the Message Center operator depresses the RLS key.

Query and/or De-activate

2.31 The operator presses the MSG CANCEL key, and the associated lamp lights. The operator then dials the user station DN. No **ringback** is heard but the state of the MSG CANCEL lamp reflects the state of the user's message waiting indication in the same way as the MSG INDIC lamp above.

At this point, the operator can change the state of the user's message waiting indication from active to inactive by pressing the MSG CANCEL key, or the indication can be left as it is. The connection is released when the message center operator depresses the RLS key.

3. EQUIPMENT REQUIREMENTS

USER STATION SETS	<p>3.01 A message center operator can activate a message waiting indication at any of the following types of station set:</p> <ul style="list-style-type: none"> ● SL-1 set with MESSAGE WAITING key/lamp pair assigned ● NE-500YR set (equipped with message waiting lamp) ● NE-2500YQA set (equipped with message waiting lamp) <p>a Standard SL-1, NE-500 or NE-2500 set with AMW (Generic X11.2 and subsequent only).</p>
SL-1 Sets With Message Waiting Key/Lamp Pair	<p>3.02 Message waiting indication is provided by a MESSAGE WAITING key lamp on the SL-1 set. The set can be a non-digit display set or a digit display set.</p>
NE-500YR Sets	<p>3.03 The NE-500YR station set is similar to the standard NE-500 dial telephone. except that the YR version has a neon lamp above and to the left of the dial. This light comes on during ringing in any type of call, but the message center operator can cause the neon lamp to flash in order to indicate that a message is waiting.</p>
NE-2500YQA Sets	<p>3.04 The NE-2500 station set is similar to the standard NE-2500 key pad telephone, except that the YQA version has a neon lamp above and to the left of the key pad. This lamp functions in the same way as that of an NE-500YR set.</p>
Standard 500/2500 and SL-1 Sets	<p>3.05 Message waiting indication cannot be provided visually at these sets. The user must call the message center by dialing the message center DN and check to see if there are any messages.</p> <p>3.06 If the SL-1 system is equipped with Generic X11 Release 2 or subsequent, AMW can be provided when the user goes off-hook in the form of an interrupted dial tone (120 ipm). The user can then dial the Message Center DN and collect the message. In the case of the 500/2500 sets, the interrupted dial tone form of message indication reduces the hardware requirements of the SL-1 system. In the case of the SL-1 set it frees an extra feature key for other uses.</p>
Message Waiting Lamp Requirements	<p>3.07 Should the neon bulb of an NE-500YR or NE-2500YQA station set require replacement, it is recommended that customers order replacements from Northern Telecom. The bulbs may be ordered using number A0250554. To avoid potential problems, bulbs must meet the following criteria:</p> <ul style="list-style-type: none"> ● Ignition Voltage. The bulb must light when a voltage of 90 ± 2 V dc is applied. ● Extinguishing Voltage. The bulb must extinguish when the applied voltage is reduced to 60 ± 2 V dc. ● Power Consumption. When a resistance of 30KΩ is placed in series with the bulb, the power consumption of the bulb must be no more or less than 0.25 W when operated at 90 ± 2 V dc.

3.08 If the bulbs installed in an NE-500YR or NE-2500YQA set do not meet the criteria listed above (e.g., resistance is higher than recommended), the system may see the bulb as being faulty and give a misleading indication to the Message Center operator, and/or disable the feature.

MESSAGE CENTER
STATION SETS

3.09 The following types of station apparatus are suitable for use in a message center:

- | 16-digit display SL-1 sets
- 16-digit display attendant consoles.

SL-1 Sets

3.10 *ACD-Type Message Center.* 16-digit display SL-1 sets are recommended for use in an ACD-type message center. The ACD SL-1 set is similar to the standard 16-digit display SL-1 set, but has provision for plugging one or two headsets into the set in place of a handset. The sets should be equipped with MSG IN-CALLS, MSG INDIC, MSG CANC and NOT READY keys, in addition to other keys as required.

3.11 A plug-in 24 V transformer or QUT1 centralized power supply is required to power the digit display and/or any add-on modules (see 553-2001-110 for more information).

3.12 Other ACD-type features are described in 553-2671-100.

3.13 *DN-Type Message Center.* The same SL-1 and ACD SL-1 sets used for an ACD-type Message Center can also be used for a DN-type Message Center. In this type of installation, the set should be equipped with an MC-DN key instead of a MSG IN-CALLS key. The MSG INDIC, MSG CANC, and NOT READY keys are also required.

Attendant Consoles

3.14 The 16-digit display console is recommended for message center use, since with the 8-digit console, some truncation of digit display information can occur. The console(s) can be equipped with a MSG CENTER ICI key to facilitate the control of a message indication at user stations (although this is not required, since message waiting indication can be performed by direct access), as well as the MSG INDIC and MSG CANC keys.

4. SL-1 SYSTEM REQUIREMENTS

4.01 The Message Center feature is available with Generics X0.5, X09, and X1 1, and X37. The feature is available with all issues of the above generics. The interrupted dial tone form of message indication is available on Release 2 of Generic X1 1.

QSY22 POWER SUPPLY

4.02 Power to operate the neon message waiting lamps for NE500YR and NE-2500YQA station sets is supplied by the QSY22 message waiting power supply unit. (SL-1 type sets do not require this power supply.) This unit replaces the earlier QSY 19 power unit.

4.03 The power supply can be mounted in the following cabinets: QCA23 (SL-1LE), QCA28 (SL-1A), QCA37 (SL-1M), and QCA8. The power supply is capable of providing power to the message waiting line cards of two full PE cabinets (QCA7 and QCA8).

Description

4.04 The QSY22 (Fig. 4-1) is a -48 to -150 V dc converter with 1.0 A output capacity. The input (TB1-1) is fused at a 6.25 A (slow blow) and accommodates the standard 48 V dc power from the SL-1. The power ground to the SL-1 is connected via TB1-2. The logic ground to the SL-1 logic ground bus bar is connected via TB1-3. The outputs consist of:

- seven separately fused -150 V dc feeds at connector J14 with a common alarm bar, and seven separately fused -150 V dc feeds at connector J15 with a common alarm bar. This provides -150 V dc power to the cabinet in which the QSY22 is mounted. The fuses are rated at 0.25 A each and are numbered 1 to 7 (from connector J14, and 10 to 16 (from connector J15). In the event of a fuse failure, a front panel Light Emitting Diode (LED) lights, indicating that failure has occurred. There is no system alarm to detect loss of output.
- an additional -150 V dc output (TB1-4) fused at 1.33 A (fuse No. 8 on the front panel fuse block) which provides power to the QBL16 power distribution unit on an adjacent PE cabinet.

Mounting

4.05 The unit mounts between the uprights where the terminal block (TBC) is on a normal SL-1 cabinet. To accommodate the power supply, the TBC terminal block is moved up if necessary to facilitate the installation of the QSY22 unit (see Fig. 4-1). The power supply is then installed below the terminal block.

QSY19 POWER SUPPLY

4.06 The QSY 19 power supply, which is no longer available (Fig. 4-3), is any early version power supply used to power the message waiting lamps. It can be mounted in the QCA8, QCA23, and QCA28, and QCA37 cabinets in the same way as the QSY22. The inputs and outputs are fused in the same way as the QSY22, however there is no alarm LED on the front panel.

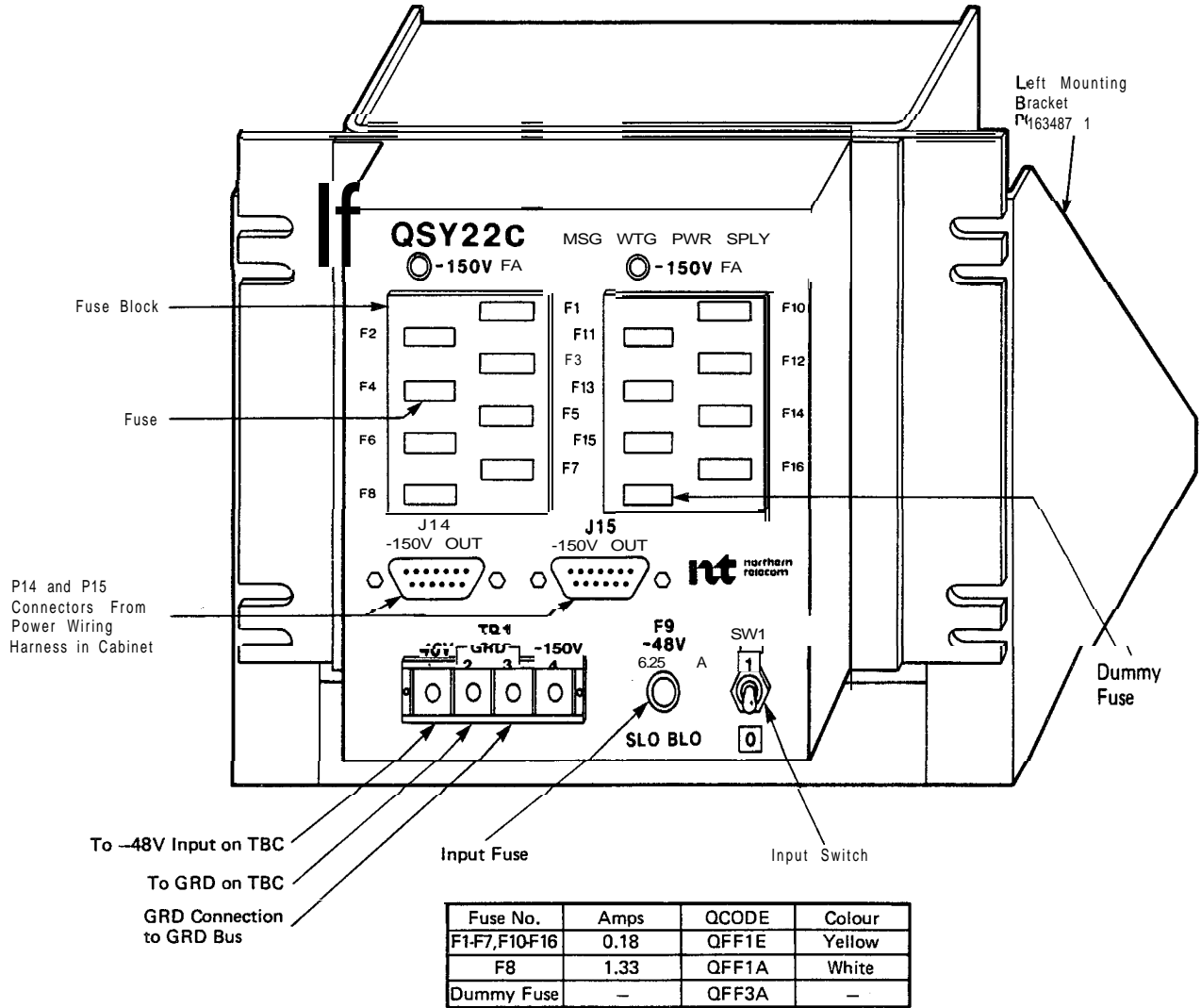


Fig. 4-1
QSY22 Power Supply Connections

4.07 The -48 V dc from the SL-1 system (Fig. 4-4) is connected to the **TB-1** terminal terminal of the **QSY 19**. The power ground connection to the SL-1 is made via pins 8-15 of P14. The -150 V dc output for the QBL16 on the **QCA7** cabinet is taken from **TB1-3**. A load strap is connected from **TB1-2** to **TB1-3**.

*Caution: The installer must insure that the load strap (Fig. 4-3) is installed before connecting -48 V dc power to the **QSY19** power supply, otherwise the power supply could be damaged. If a field installation of the load strap is required, 16 A WG red stranded wire is recommended.*

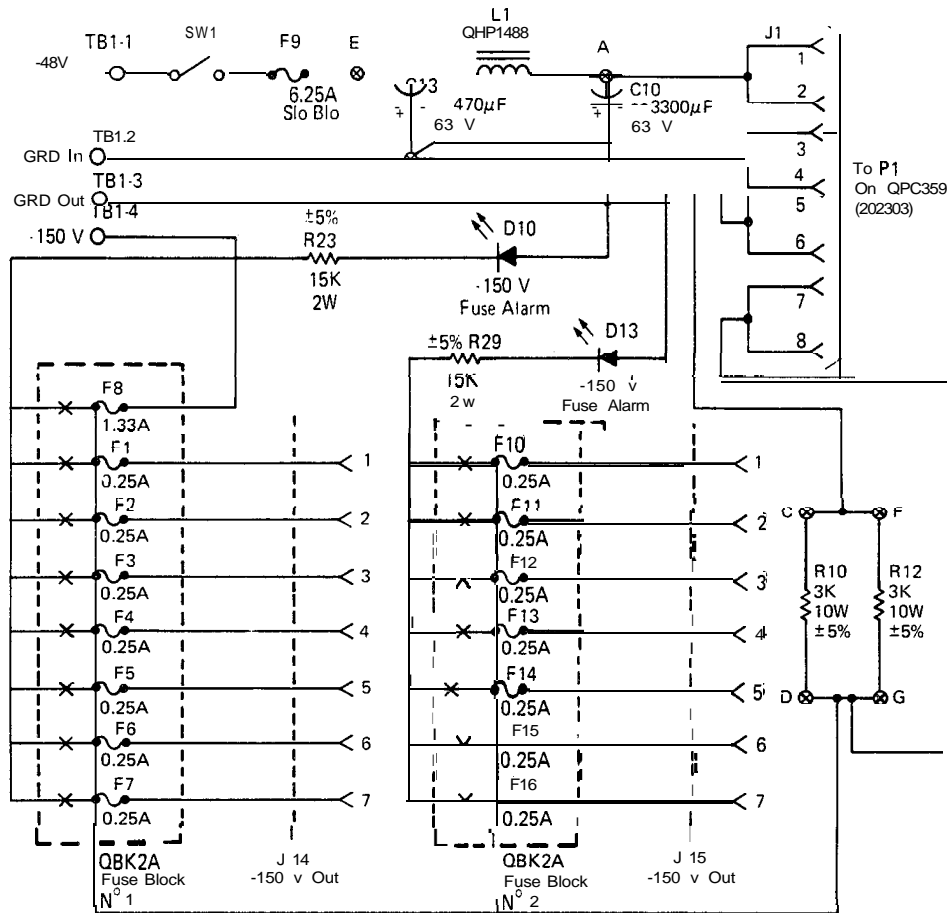


Fig. 4-2
QSY22 Power Supply Circuit Diagram

QBL16 POWER DISTRIBUTION UNIT

4.08 Power distribution in the associated PE cabinet is done by the **QBL16** Power Distribution Unit (Fig. 4-5). The -150 V dc input for the **QBL16** comes from the **QSY22/QSY19** via a **QCAD2** 10 AWG power cable which connects to the **TB1** terminal of the **QBL16** (Fig. 4-6). The power is divided into one lead for each shelf, each protected by one fuse (0.18 A) in the **QBL16**. This power goes out to the shelves via the **P14** connector into which plugs the cable from the cabinet wiring harness.

Mounting

4.09 The **QBL16** is mounted between the center uprights in the bottom of the **QCA7** cabinet in the same relative location as the power unit.

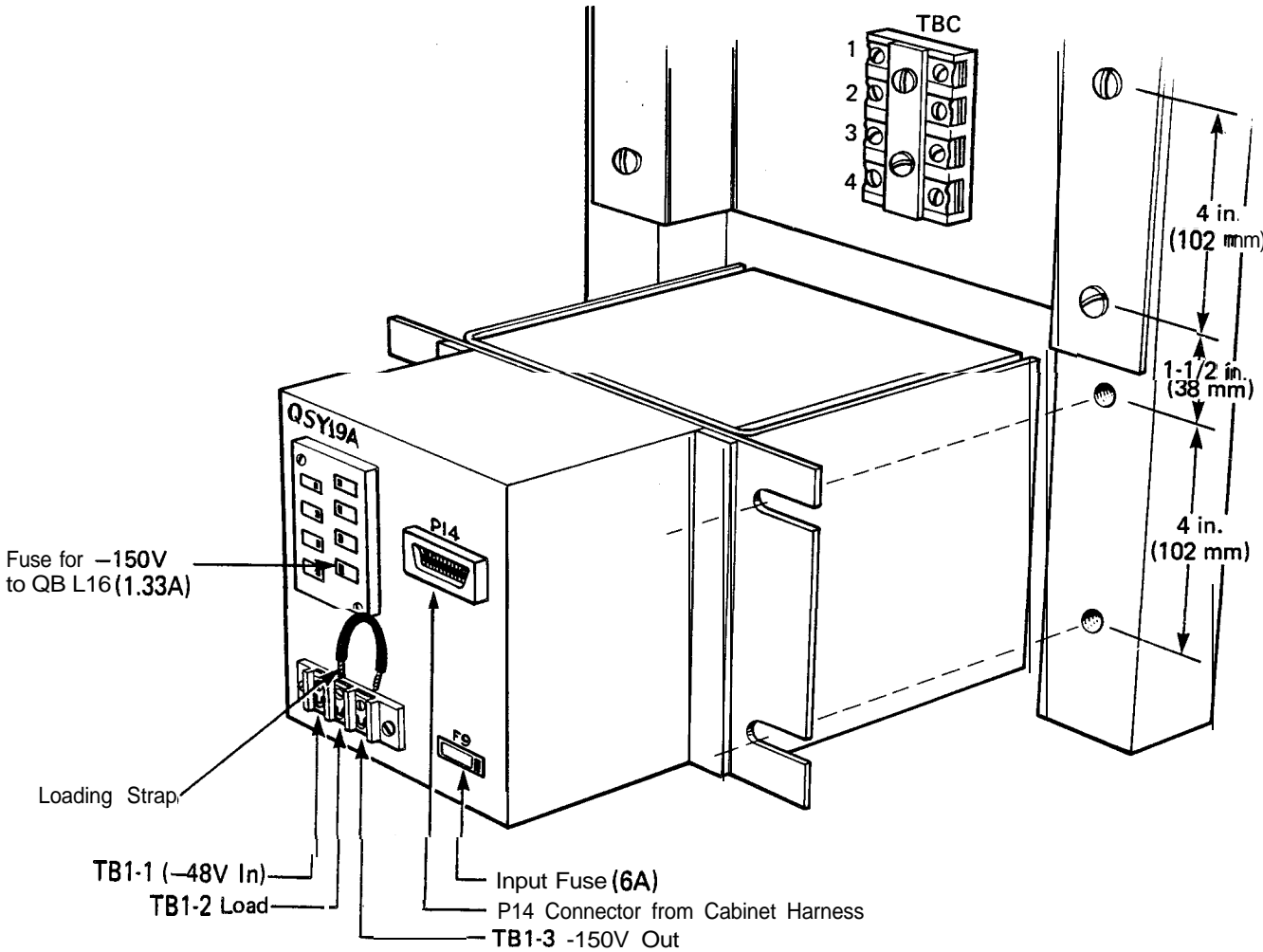


Fig. 4-3 QSY19 Power Supply Connections (This unit is no longer available)

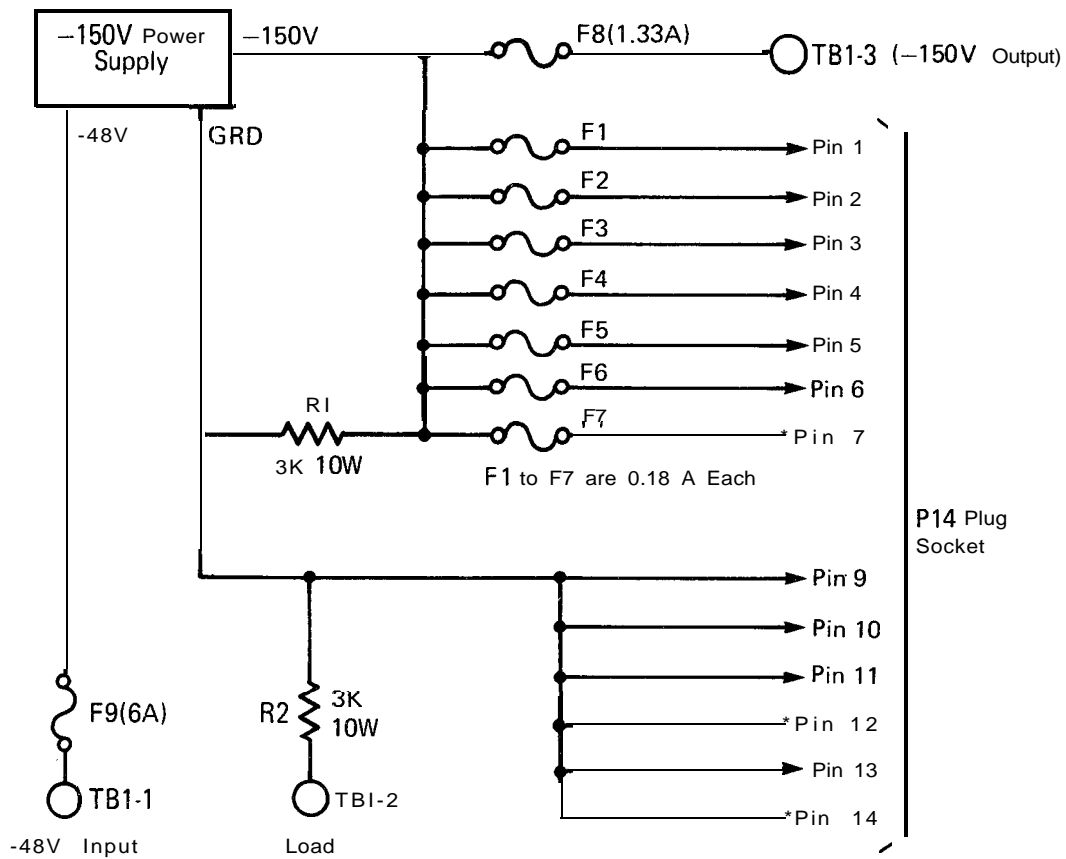


Fig. 4-4
QSY19 Power Supply Circuit Diagram

PE Cabinets and Shelves

4.10 The power unit and **QBL16** distribution unit can be used with PE cabinets of the following vintages:

- | QCA8: vintage E or subsequent
- | QCA7: vintage D or subsequent
- | QCA23, QCA28, QCA37: all vintages.

4.11 These cabinets are equipped with wiring harnesses that incorporate a cable and P14 connector. The connector plugs into the **P14** socket of the power unit or distribution unit, and the wiring harness provides -150 V dc power distribution to the PE shelves in the cabinet.

4.12 The cabinets used with the power unit or power distribution unit must contain shelves of the following vintages:

- . QSD3, QSD7: vintage B or subsequent
- | all other PE shelves: all vintages.

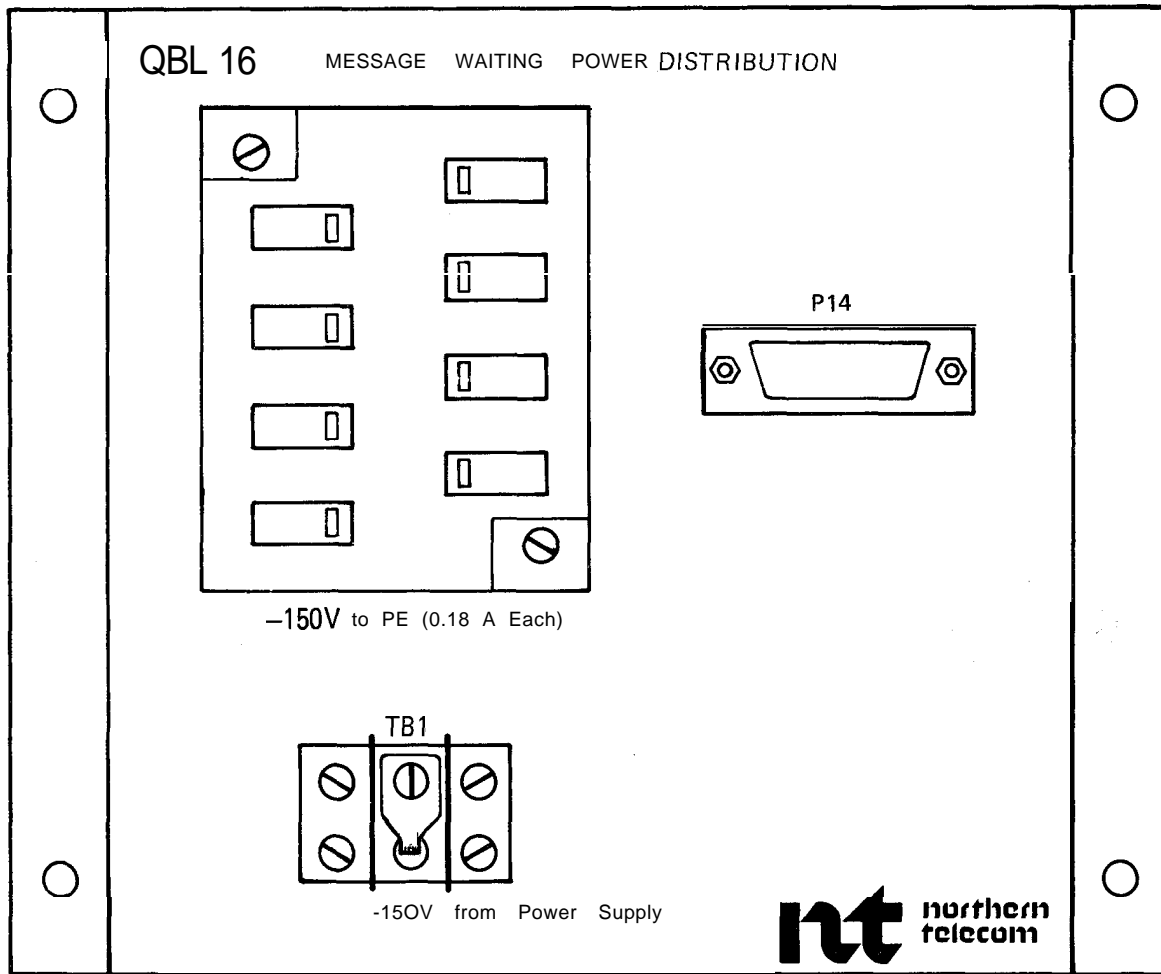


Fig. 4-5
QBL16 Power Distribution Unit Connections

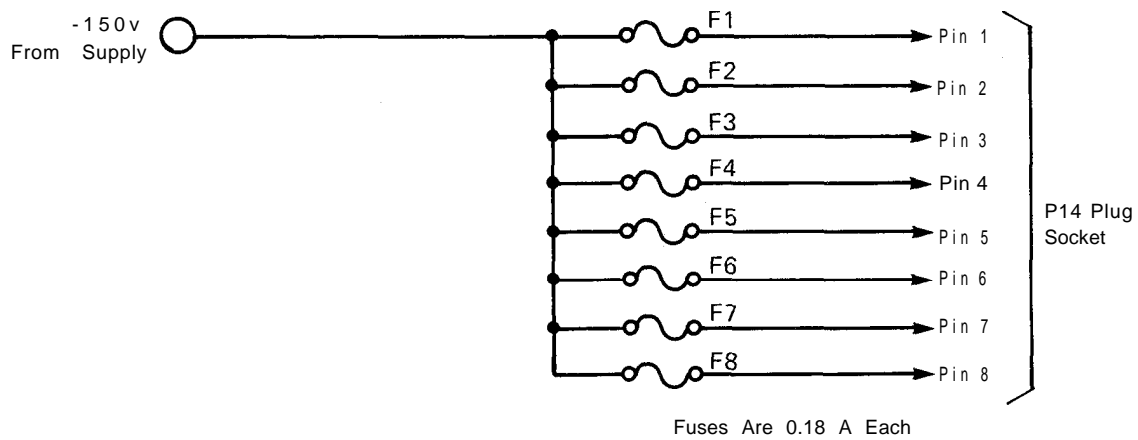


Fig. 4-6
QBL16 Power Distribution Unit Circuit Diagram

5. POWER SUPPLY INSTALLATION

- QSY22** 5.01 The Message Waiting power supply is installed according to the procedures in Chart 5-1. The QSY22 Message Waiting power supply supercedes the QSY19 and can be mounted in any of the following cabinets: QCA8, QCA23, QCA28, or QCA37. The unit is mounted between the uprights immediately below the TBC (Fig. 4-1 and 4-3). To facilitate the installation of the Message Waiting power supply, the TBC may have to be moved up. If re-location of the TBC is required, this should be done before the procedures of Chart 5-1 are started.
- QBL16** 5.02 The QBL16 Message Waiting Power Distribution Box is installed according to the procedures of Chart 5-2. This unit is mounted in the QCA7 companion cabinet next to the cabinet containing a QSY22 or QSY19, and in the same relative location within the cabinet as the QSY22/QSY19.

Chart 5-1 INSTALLING THE QSY22 MESSAGE WAITING POWER SUPPLY

Note: The QSY22 includes the cables required for the -48 and $\bar{\text{—}}$ 150 V power leads. The cabinet wiring harness may or may not contain a P14 plug, depending on whether or not PE is provided in the cabinet.

STEP	PROCEDURE
1	Set the input switch on the QSY22 power unit to OFF.
2	Remove the two flat-head screws securing the plastic shield over TB1 of the QSY22 unit.
3	Disconnect the lead assembly supplied with the QSY22 unit.
4	Position the QSY22 unit in the cabinet and secure it with four mounting screws (Fig. 4-1).
5	Connect the BLACK lead (with two lugs) between TB1 terminal 3 of the QSY22 unit, and the ground bus in the center of the cabinet.
6	Connect the other BLACK lead (lug end) to TB1 terminal 2 of the QSY22 unit, and to the ground terminal (1) of the TBC.
7	If the Message Waiting shelves and harness are provided in this cabinet, connect the P14 connector of the cabinet harness to connector P14 of the QSY22 unit.
8	Connect the RED lead (lug end) to TB1 terminal 1 of the QSY22 unit, and the other end to the -48 V terminal (3) of the TBC.

Caution: This connection is a LIVE connection. Care must be used when making this connection, since a short to ground could cause a complete loss of power to this cabinet.

Chart continued

PRACTICE 553-2691-100

Chart 5-1 Continued
INSTALLING THE **QSY22** MESSAGE WAITING POWER SUPPLY

STEP	PROCEDURE
9	Set the INPUT switch of the QSY22 to ON.
10	Test for -150 V at TB1 terminal 4.
11	If a QBL16 distribution box is to be provided in a companion QCA7 cabinet, perform the steps of Chart 5-2, then return to Step 12 of this chart.
12	Secure the plastic shield in place over TB1 with two flat-head screws.

Chart 5-2
INSTALLING THE **QBL16** MESSAGE WAITING DISTRIBUTION BOX

STEP	PROCEDURE
1	Position the QBL16 unit in the QCA7 cabinet and secure it with four mounting screws.
2	Remove the -150 V dc output fuse from the QSY22/QSY19 power supply (F8, 1.33 A).
3	Run the QCAD2 power cable between the QSY22 (or QSY 19) power supply and the QBL 16 distribution box.
4	Connect one end of the QCAD2 cable to the terminal on the QBL16 (Fig. 4-5).
5	Connect the other end to the terminal TB1-4 of the QSY22 power supply (Fig. 4-1). In the case of the QSY19 power supply, connect it to terminal TB1-3 (Fig. 4-3).
6	Connect the P14 plug from the QCA7 wiring harness to the P14 socket on the QBL16 .
7	On the QSY22/QSY 19 power unit, replace the -150 V dc output fuse.
8	Test for -150 V dc at TB1 of the QBL16 power distribution box.

6. FEATURE INTERACTION

All Generics

6.01 *ACD-Message Center*. The operation of ACD-type message center SL-1 sets is basically the same as an ACD system with incoming call queues and available agent queues. The ACD-type message center cannot operate in combination with an attendant-type message center. However, if all SL-1 sets are in the 'Make busy' mode (not logged in), message center calls can be routed to the attendants who can then function as the message center. Queue overflow features are allowed for a message center ACD-DN in the same way as for any other ACD system with the properly equipped package. Other ACD features such as RAN, music, etc. operate as for a normal ACD system with the appropriate packages. See ACD NTP for more details.

6.02 A message center operator cannot originate calls on the MSG IN-CALLS key; therefore originating features are not applicable on this key. Separate DN keys must be provided for these functions.

6.03 *DN-Type Message Center*. The message center DN must be the prime DN, otherwise all normal features can be assigned to this DN.

6.04 *Attendant-Type Message Center*. Once a call is extended to an ACD message center by an attendant, it is released completely from attendant operation; recall, camp-on, etc. cannot be activated. For calls extended to a DN-type message center, normal attendant functions, such as recall and camp-on, can be used. Other attendant functions operate normally.

6.05 *Station Features*.

- (a) *Call Forward (All Calls)*. Call Forward should be denied at stations serving as the message center. On a station basis, Call Forward takes precedence over the message center. If a call is forwarded to another set, activation of message waiting depends on whether or not the second set has 'message waiting allowed'.
- (b) **Call Forward Busy (CFB)**. CFB should be denied at stations serving as the message center. An option is provided to allow DID calls to a busy station to be routed to the message centre. If this option is selected by the customer, message waiting takes precedence over the customer-defined path for CFB.
- (c) *Call Forward No Answer (CFNA)*. CFNA should be denied at stations serving as the message center. On a station user basis, message waiting takes precedence over the customer-defined path of CFNA.

The capability to light and extinguish message waiting lamps can be used in conjunction with CFNA to simulate a multiple message centre. Any station equipped with message lamps, but without 'message waiting allowed' class of service, can CFNA to specified DNs on the SL-1 stations equipped with MSG INDIC and MSG CANC key/lamp pairs.

These stations have the capability to light or extinguish message waiting lamps by manually entering the DN of the station for which a message was taken. Call processing is the normal call processing for CFNA, not the message center call processing. When a call is forwarded, the MSG INDIC lamp does not light since this is not true message center operation.

- (d) *Hunting*. Hunting should be denied at stations serving as the message center. On a station user basis, hunting takes precedence over message waiting. However, message waiting can be activated after hunting provided the hunted station is 'message waiting allowed' and does not answer the call. If desired, the MC-DN can be specified as the hunt number.
- (e) *Listed Directory Numbers*. A message center can be assigned to a Listed Directory Number (LDN) and behaves in a similar manner to an attendant message center. The calls come in on an LDN ICI instead of the MSG CENTER ICI, and direct message calls do not activate the MSG CANC key. The operator must access the user station directly to cancel that station's message indication.

Generic **X05**

6.06 Generic **X05** does not have provision for DN-type message centers. In addition, delay announcements and other ACD features are not permitted on ACD-type message centers.

Generic **X09/X37**

6.07 The restrictions given for Generic **X05** are lifted for Generic **X09**.

Generic X11.2

6.08 Generic XI 1.2 and subsequent versions of Message Center have all the feature interactions of Generic **X09**, in addition to the following.

6.09 Flexible CFNA to any DN. Flexible CFNA to any DN forwards unanswered calls to a pre-designated CFNA DN. All sets with 'message waiting allowed' have the CFNA DN assigned to the message center regardless of whether Flexible CFNA has been selected by the customer or whether CFNA is allowed or denied for the set.

6.10 Call Transfer/Conference from 500/2500 Sets. Message waiting interrupted dial tone is not provided when the user flashes the switchback to activate Call Transfer or Conference. The normal dial tone for this purpose is provided.

6.11 Ring Again for 500/2500 Sets. Message waiting interrupted dial tone is not provided when the user flashes the switch back to activate Ring Again. The normal dial tone for this purpose is provided.

7. MESSAGE CENTER PACKAGING

7.01 If an ACD-type message center is not required, the customer must order only the message center package. However, if an ACD-type message center is required, the appropriate ACD package should be ordered in addition to Message Center. An option for the ACD-type message center is the Integrated Messaging System (IMS). See 553-2781-100 for IMS requirements.

7.02 Note that an ACD package is required if an attendant-type message center is accessed by a MSG CENTER ICI key. This is required for the phantom ACD-DN associated with the MSG CENTER ICI key.

7.03 See 553-2671-100 for available ACD packages. In any case, a Northern Telecom sales office should be consulted for further information.

