Gentner

GENTNER BROADCAST SYSTEMS
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SPH-4

TABLE OF CONTENTS

1. Introduction.

This manual provides all information needed for proper installation and operation of the Gentner Telephone System **Model SPH-4.** This manual should be totally read and understood before the unit is installed.

The SPH-4 is a full-featured telephone hybrid that offers easy operation and clean, dependable phone audio. The SPH-4 features include:

* **A hybrid insensitive to load changes.**

* **A "Caller Control" feature that automatically reduces caller volume (by an adjustable amount) when talent is speaking. This feature also compensates for hybrid leakage.**

* **Caller-only outputs as well as a "Mix" output of caller and talent (good for feeding a tape recorder).**

* **A monitor speaker amplifier with program-controlled speaker dimming or muting.**

* **Caller mute without line disconnect.**

* **A built-in test generator for easy hybrid null adjustment.**

* **Direct connection to the telephone line.**

* **A back-fed modular jack to allow hook-up of an attendant telephone set.**

* **Active inputs and outputs, RFI-protected.**

* **Remote control capability.**

* **A click-mute which suppresses switching transients during termination.**

* **Send EQ and level control which conforms with FCC Rules, Part 68.**

2. Specifications and Warranty Information.

SEND INPUT:	Balanced active 600 ohms. Minimum input level -25dBm Maximum input level +18dBm
OUTPUTS:	
"Caller Only" and "Mix":	600 ohms balanced active. Nominal load 600 ohms; minimum 50 ohms. Nominal output 0dBm; maximum output +22dBm.
"Mix" (unbalanced):	10k ohms -6dBm nominal output
Speaker (unbalanced):	8 ohms; minimum load 4 ohms.
Conference output:	600 ohms unbalanced; nominal 0dBm.
TRANSMIT LEVEL TO TELEPHONE:	Adjustable; -9dB maximum.
HYBRID NULL:	-20dB, minimum -15dB.
FREQUENCY RESPONSE:	
Send input to "Mix" output:	$\pm$ 0.2 dB 20-20KHz.
Telephone input to system output:	300 to 3200Hz $\pm$2dB typical
DISTORTION:	
Send input to "Mix" output:	.1% THD maximum, 20-20KHz.
Telephone input to system output:	.1% THD maximum
SYSTEM HUM AND NOISE:	greater than -60dBm
OTHER:	
"Caller-Control" range:	0dB to -50dB maximum override, internally expandable.
"Beep" frequency:	Nominally 1200Hz (modified square wave)
Squelch sensitivity:	Adjustable to 30dB below nominal send level.
Speaker Dimming:	0-22dB, internally adjustable.
SquelchRelease Time:	100-400 milliseconds, internally adjustable.
LOGIC INPUTS/OUTPUTS:	
All On, Off, Click-Mute, etc. commands:	Apply Common
Lamps for remote control switches:	Open-collector against +24VDC

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CORPORATE WARRANTY POLICY
AS OF APRIL 15, 1989

1. ALL SINGLE LINE PHONE SYSTEMS WILL CARRY A ONE YEAR LIMITED WARRANTY COVERING PARTS AND LABOR.

2. ALL PREWIRED PATCH PANELS WILL CARRY A ONE YEAR LIMITED WARRANTY COVERING PARTS AND LABOR.

3. ALL MULTI-LINE TELEPHONE SYSTEMS WILL CARRY A 90 DAY WARRANTY ON PARTS ONLY AND A ONE YEAR LIMITED WARRANTY ON LABOR.

4. ALL OTHER PRODUCTS WILL CARRY LIMITED WARRANTIES OF AT LEAST 90 DAYS ON PARTS AND LABOR.

LIMITED WARRANTY SHALL MEAN THE FOLLOWING:

A. GENTNER ELECTRONICS ASSUMES THE RESPONSIBILITY OF MANUFACTURING A PRODUCT AS ADVERTISED FOR THE PERIOD OF THE WARRANTY. ANY DEFECTS IN THE PRODUCT CAUSED BY POOR MANUFACTURING TECHNIQUES, INFERIOR COMPONENTS AND/OR DESIGN ERRORS WILL BE REPAIRED BY GENTNER.

B. ALL RETURNED SHIPMENTS FOR REPAIR SHOULD BE SENT **PREPAID** UNDER A GENTNER RETURN AUTHORIZATION NUMBER.

C. GENTNER WILL REPAIR THE PRODUCT IN A REASONABLE AMOUNT OF TIME.

D. GENTNER MAY, BUT IS NOT REQUIRED TO, PROVIDE THE CUSTOMER WITH A LOANER PRODUCT DURING THE REPAIR PERIOD.

E. GENTNER ASSUMES NO RESPONSIBILITY OR LIABILITY IN ANY FORM FOR LOSSES DUE TO FAILURE OF ANY OF ITS PRODUCTS.

3. Installation and Adjustment.

The Gentner SPH-4 Telephone System was designed for easy installation and use. Diagram 3-A shows the front panel controls while diagram 3-B illustrates the rear panel connections:

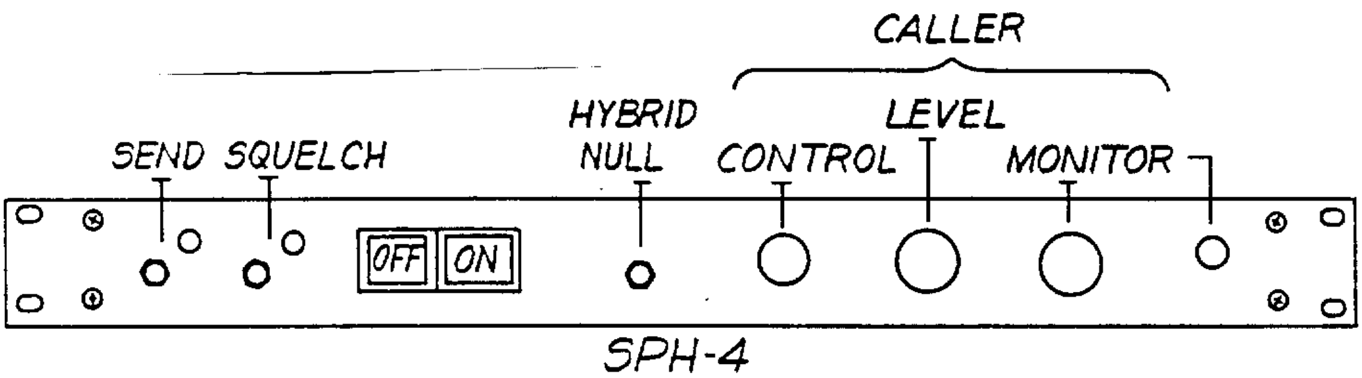


Diagram 3-A. SPH-4 Front Panel Controls.

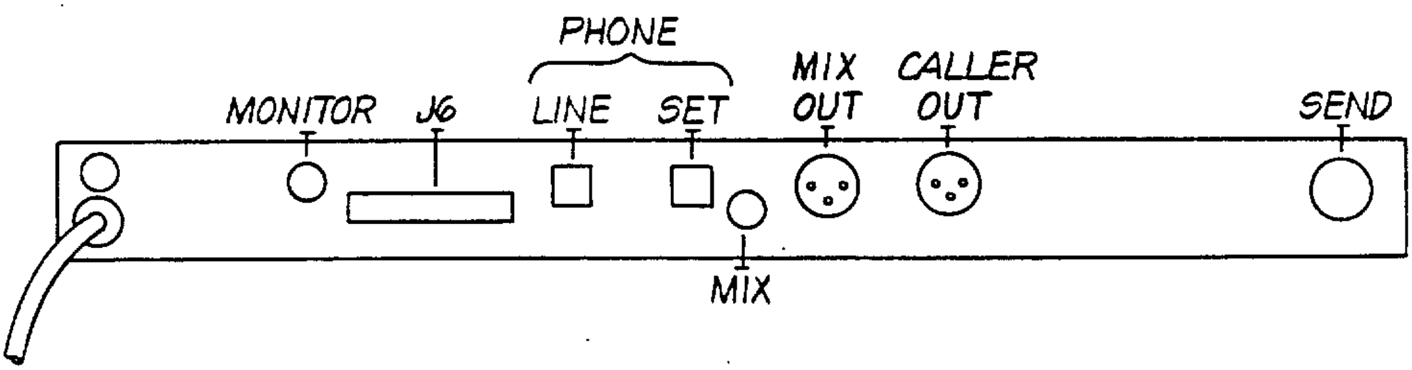


Diagram 3-B. SPH-4 Rear Panel Connections.

A. FRONT PANEL CONTROLS:

The front panel controls on the SPH-4 do the following:

1. **ON AND OFF SWITCHES.** These are not power switches. Pushing the "ON" switch will:

   a. Mute switching transients so that "clicks" and "pops" associated with line connection are suppressed.
   b. Send a "beep" down the phone line so the caller will know he is on the air.
   c. As the switch is released, the phone line is unmuted and the caller is on the air.
   d. If the "ON" switch is depressed during a telephone call, the caller's audio is muted for the duration.

Pushing the "OFF" switch will:

a. Disconnect the caller.
b. Mute the phone line audio going to your equipment.

2. **CALLER LEVEL.** This knob adjusts the volume of the incoming caller's voice.

3. **MONITOR VOLUME CONTROL.** This adjustment controls the volume of the incoming caller's voice at the speaker only. Keep this level low enough so the caller's voice doesn't echo or sound "hollow".

4. **CALLER CONTROL.** THIS IS AN EXCLUSIVE FEATURE OF THE GENTNER SPH-4. This knob adjusts the volume of the caller's voice relative to send audio; as talent speaks, caller volume is reduced. This control can be adjusted from no caller reduction to a full "speakerphone" effect.

4A. The **CALLER CONTROL** knob in the detent position generates a test tone for hybrid nulling.

5. **NULL.** This is a one-time set-up adjustment that should never be changed under normal operating conditions. It adjusts the amount of rejection of the send audio heard at the caller output. It is typically adjusted for minimum send audio at the caller output or the monitor output.

6. **SEND.** This control adjusts the volume of the send audio that is heard by the caller. This too is a one-time set-up adjustment and should not be changed under normal operating conditions. It is typically adjusted for just less than normal volume heard by the caller.

B. REAR PANEL CONNECTIONS.

1. **MONITOR.** This is a 1/4" tip-ring-sleeve phone jack. Connect a speaker or a headset between the tip-ring and the sleeve. This output will drive a 2 watt speaker and is used to listen to the caller without having to wear a headset.

2. **J6.** This 26 pin header comes with the unit. It provides remote control, unbalanced inputs and outputs, and brings out the regulated power supply.

3. **LINE.** This is where you connect a standard modular jack for the telephone line. RED and GREEN are tip and ring; YELLOW and BLACK are A-Line.

4. **SET.** Connect a standard telephone here as an attendant set. When the front panel "ON" button is activated, the telephone will not work. When the "OFF" switch is depressed, the telephone will work normally.

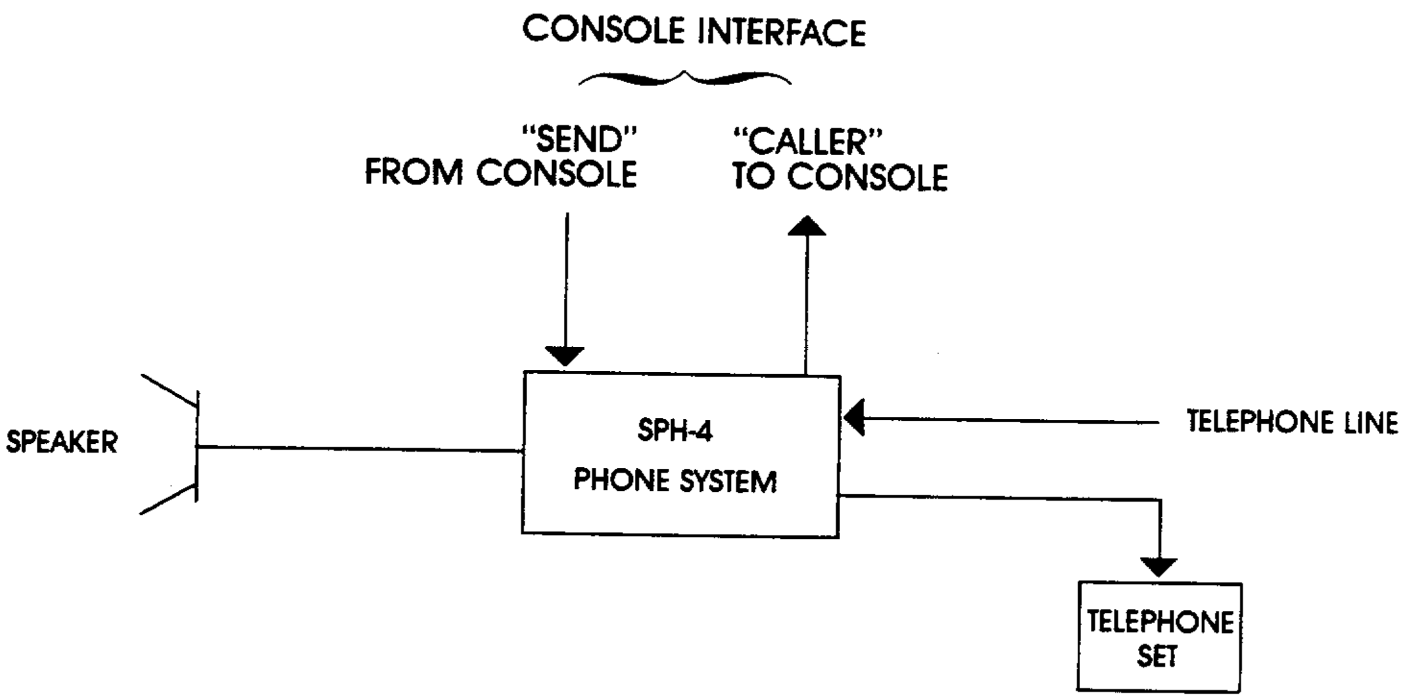
5. **MIX.** These two outputs (one is an RCA phono and the other is an XLR) contain a mixture of SEND and CALLER audio. Even when the "OFF" button is activated, send audio will appear here.

6. **CALLER.** This is the output of the caller's voice (it is an XLR as well).

7. **SEND.** This XLR input is the audio you want the caller to hear.

HOW TO INSTALL THE SPH-4 TELEPHONE SYSTEM:

The system block diagram on the next page shows the overall operation of the SPH-4; the basic set-up is shown below:

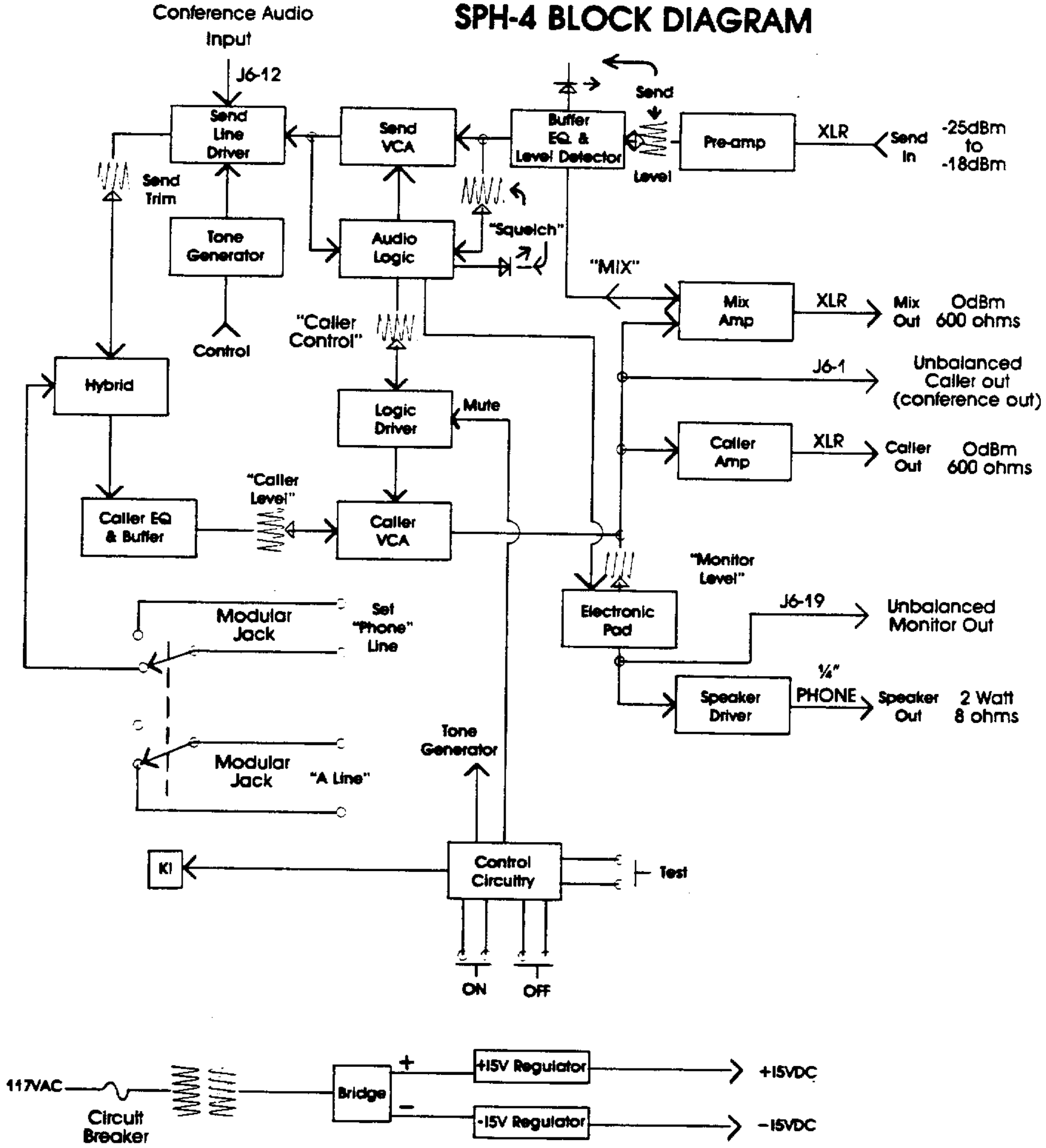


SPH-4 PHONE SYSTEM
BASIC SET-UP

1. Connect the send audio into rear panel J1. Verify that the source (send) level from console or mixer to be fed to the SPH-4 is between -25dBm and +8dBm.

2. Connect a speaker or headphones to either front or rear-panel phone jacks. Connect an SPH-4 output, either "Caller Out" or

GENTNER PHONE SYSTEMS
SPH-4 BLOCK DIAGRAM



"Mix" out to your console or amplifier. Note these two rear-panel XLR outputs are active-balanced. They will directly feed a load of 150 ohms or more, balanced. **IF AN UNBALANCED LOAD IS TO BE CONNECTED, WIRE IT BETWEEN XL PIN 1 (GROUND) AND EITHER XL PIN 2 OR XL PIN 3.**

3. Connect the unit to power. The OFF lamp and the green LED should light. Adjust the SEND control (front panel) until the red LED barely lights in the presence of normal send audio.

4. Now adjust the SQUELCH control until the green LED is on most of the time but turns off during program peaks.

5. Connect your telephone line to the "line" jack via a standard modular cord. If you are ordering a modular jack from the telephone company for use with the SPH-4, specify "USOC RJ11C." If desired, plug a standard telephone into the rear-panel "Set" jack.

The phone line should be connected to the green/red pair of the modular phone jack. If you cannot easily locate the phone line in your telephone, we suggest the following:

A. Make sure all the audio connections are wired in. Listen either to the speaker output or a caller output port by turning the "caller" pot to 12 o'clock and depressing the "on" switch.

B. Connect clip leads to the SPH-4 input phone line (green/red pair). By trial and error try different combinations of connections in the phone. When you locate a phone line the SPH-4 will terminate the line and you will hear dial tone. For call directors, try punching different lines to ensure you're on the output of the call director. **For digital systems, refer to Appendix B, Digital Telephone Connections.**

A-LINE CONNECTIONS-For single line applications you won't need to worry about connecting the A-line (or A-lead). For multi-line phones with call directors it is important to properly connect the A-line.

A-lines operate by a closure provided on the black/yellow pair of the modular jack. However, since the A-line is a "dry" closure, it may be used to operate a relay to do a number of external functions, including operation of a digital telephone (see Appendix B). To find the A-line we suggest the following:

A. Punch up any line on the call director. Hang up the phone either by placing the handset on the cradle or taping the hook switch closed.

B. With the phone cover removed, short different combinations of terminals on the phone until the light on the line you punched up illuminates. Once you have found the connection, try punching different lines to ensure the A-line follows. If it doesn't, then you have located the A-line for that particular line only. What you want is the A-line connection for the output of the call director.

C. Connect the terminals you found on the phone to the black/yellow pair.

To ensure the phone has been properly connected, do the following:

A. For single line phones - make sure the handset is on the cradle. When you depress the "on" switch, you will hear dial tone thru the speaker or at the caller output port.

B. For multi-line phones - make sure the handset is on the cradle. Punch any line on the call director and depress the "on" switch. You will hear dial tone thru the speaker or at the caller output port. Additionally, the light on the phone will illuminate. Now punch up all the different lines, making sure that dial tone follows and the light follows.

As you can see, the SPH-4 takes direct control of the phone line. **When using the SPH-4 make sure the handset is always on the cradle.** If it is not, the unit will not null properly.

6. Shut off the send audio and push the ON switch. You should hear a dial tone in any of the three SPH-4 outputs, adjustable by the front-panel "Caller" control. If you are monitoring via the headphone jacks, the level is also determined by the "Monitor" control. Adjust these two controls for the desired dial tone level. Note that the dial tone level is likely to be at least 10 dB higher than a typical caller.

7. Call the SPH-4 from another phone and set the other phone down. rotate the "Caller Control" all the way counterclockwise to enable the internal test generator. You should hear a tone in the SPH-4 output. Adjust the front panel "Null" control for the minimum tone.

8. Turn off the test tone.

9. Verify the ON logic by holding the ON button down. You should hear a continuous beep in the other telephone but nothing at the outputs of the SPH-4.

10. Return normal send audio to the SPH-4. You should hear it comfortably in the other telephone.

To hook up the SPH-4 to a digital telephone system, refer to Appendix B in the back of this manual.

4. Operation.

1. TO PUT A CALLER ON-AIR, SIMPLY PUSH AND RELEASE THE ON BUTTON.

As the "ON" button is depressed, the clicks and pops associated with punching up a telephone line are muted. At the same time, the caller hears a "beep" to let him know he is on the air.

2. TO MUTE THE CALLER WITHOUT DISCONNECT, push and hold the "ON" button.

Caller audio is muted during the time the button is held down, and the caller hears a continuous "beep" during this time. When you release the ON button, caller audio comes back.

3. TO DISCONNECT THE CALLER, push the OFF button.

The SPH-4 mutes all audio coming from the telephone line and drops the caller.

4. TO ADJUST MONITOR VOLUME, turn the "Monitor" knob.

This controls the volume of the incoming caller's voice at the speaker only. Keep this level low enough so the caller's voice doesn't echo or sound hollow.

5. TO ADJUST THE CALLER VOLUME, turn the "Caller" knob.

This controls the volume of the incoming caller's voice at your equipment.

6. USE OF THE "CALLER CONTROL" KNOB:

This is a variable control and may be set up to individual needs. It reduces the caller's volume relative to talent audio. As the knob is rotated clockwise, REDUCTION of the caller volume INCREASES when talent speaks. The net effect is that talent can "shout down" an unruly caller by speaking normally.

5. Technical Description.

The SPH-4 telephone hybrid system can be divided into three smaller systems for technical analysis. The three systems are: 1-Send or transmit audio circuits, 2-Telephone audio receive circuits, and 3-Logic, timing and control circuits.

1. SEND CIRCUITS:

Audio to be sent to the caller is applied to J1. Op-amp U1D converts the balanced audio to unbalanced for processing. 1% resistors are used in this circuit to keep common mode signals to a minimum. The unbalanced audio is fed to the send level pot. Then audio is futher amplified by U1B and low-pass filtered by U1C and its associated reactive components. The equalized audio is then applied to the input of VCA U3B where it is expanded or limited depending upon the audio level. The audio out of the VCA, the beep tone, and conferencing audio via J6 pin 12, are all combined by op-amp U4B. U4B's feedback loop employs a 10K trimmer allowing for adjustment of the audio level that is applied to the phone line. U4B drives the telephone load coil T2, via a 620 ohm resistor.

The secondary of T2 provides both A.C. and D.C. characteristics for proper loading and supervision of the telephone line. The line is connected to T2 via the relay when the "on" button is depressed.

2. RECEIVE AUDIO CIRCUITS:

Incoming audio from the telephone line is coupled through the load coil T2 and fed to the first receive amp U4A. U4A is employed as a differential amplifier with one input being fed with the sum of both transmit and receive audio from the phone line and the other input fed with a sample of transmit audio only. When the 100k null trimmer is properly adjusted, only the received or caller audio will be present at the output of D1.

The receive audio is bandpass filtered by U2A, U2B, U2C, U2D, U9C, and their associated circuitry. This filtering eliminates unwanted hum induced on the phone lines along with high frequency tone caused by the phone company's multiplexing schemes. If desired, the filtering can be bypassed by changing an internal jumper.

Filtered caller audio is applied to a 10K pot for caller level control. Caller audio is futher amplified by U9B and fed to the caller VCA U3A. This VCA is employed to mute the receive for 600 ms whenever the "on" button is depressed, and it provides the level control for the caller control feature.

The audio from the VCA drives the monitor amplifier circuit (U12) and the caller balancing amplifier implemented with op-amps U5A and U5B. The balanced caller audio is made available at J2. A sample of caller audio is mixed with a portion of send audio, and is then balanced by U6A and U6B. The balanced "MIX" audio is available at J3.

3. AUDIO LOGIC AND CONTROL:

"PRE" send VCA audio is fed to U1A where it is amplified, rectified, filtered, and compared to a constant D.C. level by U9D which in turn drives the red send LED. The LED, when lit, indicates that an adequate audio send level is present at the input of the send VCA. "PRE" send VCA audio is also applied to the squelch trimmer and is in turn amplified by U7A. This audio is then rectified, filtered, timed, and compared to a constant D.C. by U7B. The squelch decay time constant is adjustable with the 100K trimmer that follows the rectifier. Thus when the green LED winks out the squelch function is active.

The squelch comparator drives the base of U8A (U8 is a transistor array) which forces the FET to open and dim the audio going to the monitor amplifier. This provides for better acoustic isolation between the microphone and the monitor speaker. The amount of dimming can be adjusted via the 100K trimmer that parallels the FET.

The squelch comparator U7B drives a D.C. amplifier U9A via the 50K caller conquer pot. The output of U9A supplies the control voltage for the receive VCA.

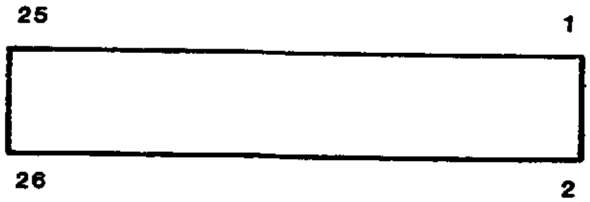
"POST" send VCA audio is amplified by U7D, rectified, filtered, and applied to a D.C. servo amp U7C. U7C is also supplied with a D.C. level from the squelch comparator. The output of U7C is a voltage which causes the send VCA (U3B) to either expand or compress the send audio depending on the levels of the PRE and POST audio samples. This is done to keep a consistent level of send audio going to the caller. It also ensures that the phone line will not be driven with an audio level greater than -9 dbm average over a three second period.

The relay and the "OFF" and "ON" lamps are driven by two 2N3904 transistors.

I.C.'s U10 and U11 are both 555 timers. U10 is the 600 ms mute timer. U11 generates the 1200 Hz caller beep tone. Transistor U8E saturates to turn on the tone continuously when the caller control pot is rotated to the test posistion. The mute timer I.C. u10 drives transistors U8B, U8D and U8C which are used as voltage level converters, to clamp the control voltage to the receive VCA (U3A) to the minus 15 Volt supply during the mute time.

6. Pinouts.

Pins	Function
J6:	
1	Auxiliary Input - Unbalanced
7	"OFF" Lamp (Open Collector)
8	"ON" Lamp (Open Collector)
9	"ON" Switch (Apply Ground)
10	"OFF" Switch (Apply Ground)
11	External Mute (Apply Ground)
12	Aux. Unbalanced In (Post EQ)
13 - 18	Ground
19, 20	+24VDC (Lamp Common)
21, 22	-15VDC
24 - 26	+15VDC



J6 PIN CONFIGURATION

XLR Connections:

Pins	Function
1	Ground
2	Low
3	High

For unbalanced connections, always connect between either 2 or 3 and pin 1.

Modular Phone Jacks:

Pins	Function
1 & 4	A-Line (Black and Yellow Wires)
2 & 3	Phone Line (Red and Green Wires)

7. Appendix A - Telephone Line Basics.

I. Definition: Telephone Line.

A simple way to define a telephone line is to describe it as a two-way, two-wire communication transmission line. It gives the user the ability to "send" and "receive" information simultaneously (a great advantage over one-way systems). Diagram A-1 shows how a telephone line permits two-way transmission of information:

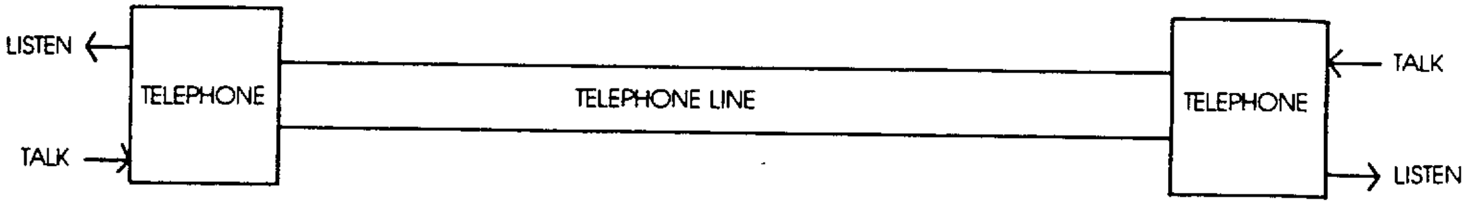


Diagram A-1. A telephone line.

II. How a telephone line is processed:

The two wires making up the telephone line are called "Tip" and "Ring", abbreviated "T" and "R"; these names are carried over from the early days of the telephone when operators used physical patch cords to tie calls together. "Tip" and "Ring" refer to the physical position on the patch plug; since the tip was the part most likely to be touched by the operator, it was made the grounded part of the circuit. The term "Ring" in this usage has nothing to do with the ringing mechanism of the telephone.

Tip and Ring are brought from the local telephone company Central Office (C.O.) to each user's telephone equipment (homes, businesses, etc.). When the user's tip and ring reaches the C.O., it is usually "two to four wire" converted. This means that the "send" and "receive" audio is separated so the C.O. can process the two sources of audio independently. The information is then sent to other C.O.'s by a variety of methods including (1) Cable; (2) Microwave and (3) Satellite. Diagram A-2 illustrates this two to four wire conversion:

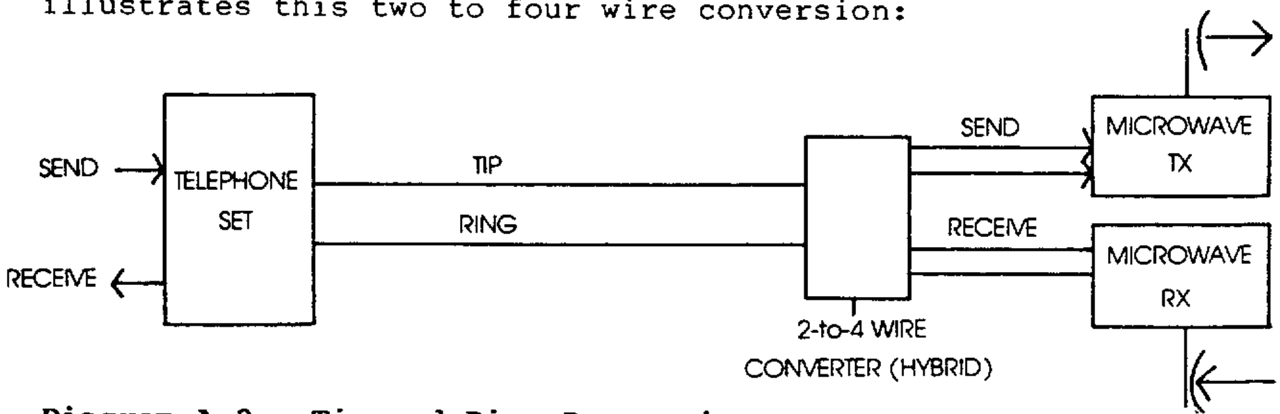


Diagram A-2. Tip and Ring Processing.

Diagram A-2 shows how send audio is taken from the two to four wire converter (hybrid) and is transmitted via a microwave transmitter. In addition, receive audio is taken from the microwave receiver and is applied to the input of the hybrid. The combined communications become tip and ring for processing at the user's equipment.

It is important to note the totally separate talk and listen paths provided by the microwave transmitter and receiver. The entire telephone transmission path is often thought of as a single path, when in reality it is a dual system. The entire system is illustrated in diagram A-3:

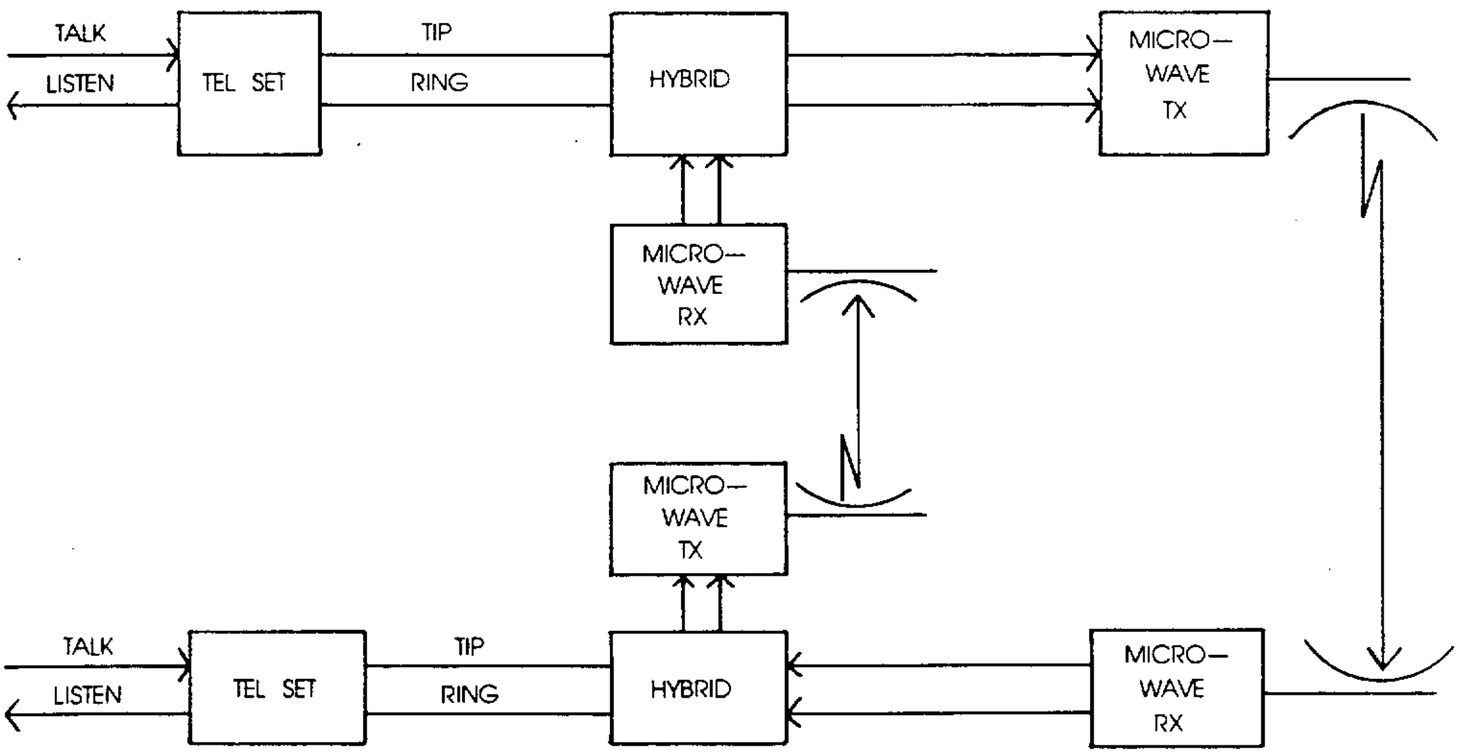


Diagram A-3. A Telephone Transmission System.

III. Tip and Ring - A More Detailed Look.

Since the customer has little control over what the telephone company does with calls once they reach the C.O., a more detailed look at tip and ring will help explain the telephone's operation. This will be discussed as follows:

A. Battery. At the C.O., the telephone company (Telco) places a D.C. voltage on tip and ring. This D.C. voltage is usually provided by a trickle-charged battery (the battery keeps the system working during power outages). The voltage varies from 20 VDC to about 48 VDC with the tip and ring on hook (or no load). This voltage provides D.C. for the telephone instrument to operate.

B. Ringing. When someone calls, a ring voltage of 105 VAC is provided to drive the ringer inside your telephone. When you pick up the handset, the hook switch does the following: 1) it disconnects the ringer; and 2) it connects the telephone to the telephone line. This process is known as "terminating the line." Anytime a D.C. path of 600 ohms or less is provided to the tip and ring, the line will be terminated and the call will be "answered."

C. Disconnect. When a call is over, the reverse sequence of events occurs. As the C.O. senses disconnect (change in load on tip and ring) it usually reverses the D.C. voltage on the receiving phone's tip and ring (some systems simply provide dial tone without D.C. reversal). This stops the long distance billing (if applicable) and gives a dial tone to the receiving caller.

IV. Single line telephone systems.

A single line telephone system simply refers to a standard telephone set. Connection to Telco is usually provided through a modular phone jack (USOC RJ-11C). This jack has four wires going into it; the inner two conductors are tip and ring and the outer conductors are not used.

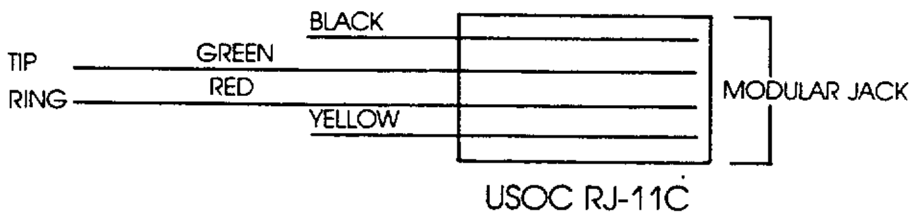


Diagram A-4. USOC RJ-11C.

Note that tip is usually a green wire and ring is red.

Inside the telephone, tip and ring go to the hook switch. When you pick up the telephone, tip and ring are routed to a hybrid that provides separation for the talk and listen paths going to the handset. In addition, the ringer is disconnected:

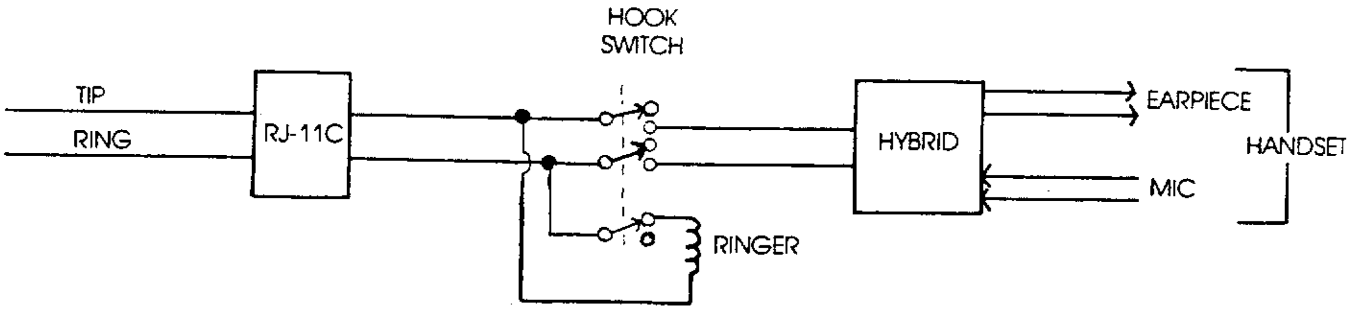


Diagram A-5. Single line telephone set - simplified.

V. Multi-line telephone systems.

In multiple line usage, the various lines enter the customer's Key Service Unit (KSU), providing the necessary routing and control functions required by the customer. There are two types of KSUs: fat wire and slim wire.

The fat-wire phone system uses at least one 25 pair cable going to each telephone. The telephones act like a rotary switch. When a line button is depressed, the appropriate tip and ring is routed to the telephone instrument and the button light is turned on. The following diagram illustrates this operation:

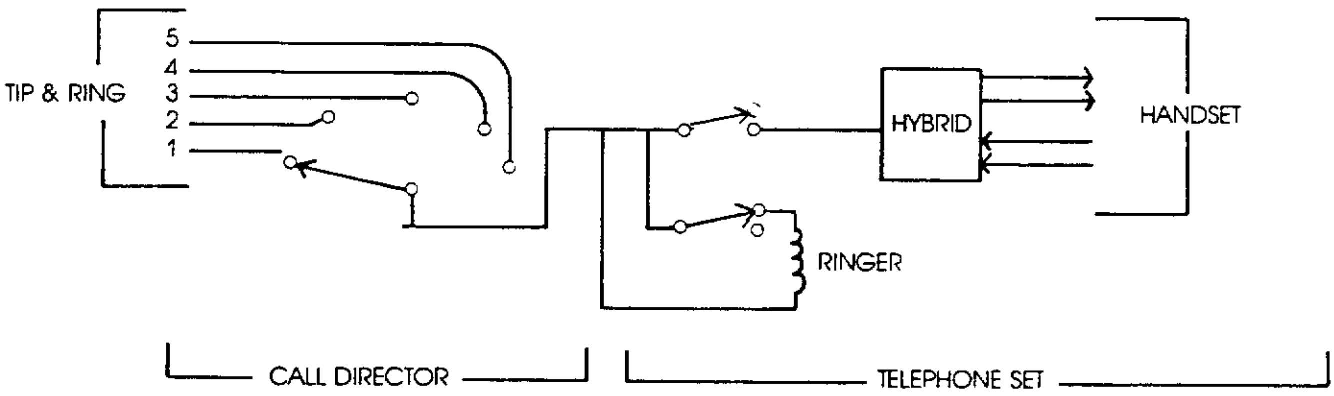


Diagram A-6. Multiple line call direction.

This "rotary" switch is commonly called a "call director" because it directs the appropriate phone line to the telephone instrument. The closure that turns the light on in the telephone is called the "A-lead" or "A-line."

As each button is depressed, the following occurs:

1. The telephone line is routed to the telephone instrument.

2. The button lamp is illuminated.

<u>Thus, the function of the KSU is to:</u>

1. Route the phone lines to all of the telephones.

2. Provide control functions to light the button lamps.

3. Place calls on "hold."

The KSU may also be used as an intercom and for other accessory functions.

The Gentner SPH-4 easily adapts to the fat-wire phone system by connection the tip and ring output of the call director to the tip and ring input of the SPH-4. In addition, the SPH-4 provides an A-line closure to turn on the lamp (refer to Installation section for more details). The following diagram shows how this is accomplished:

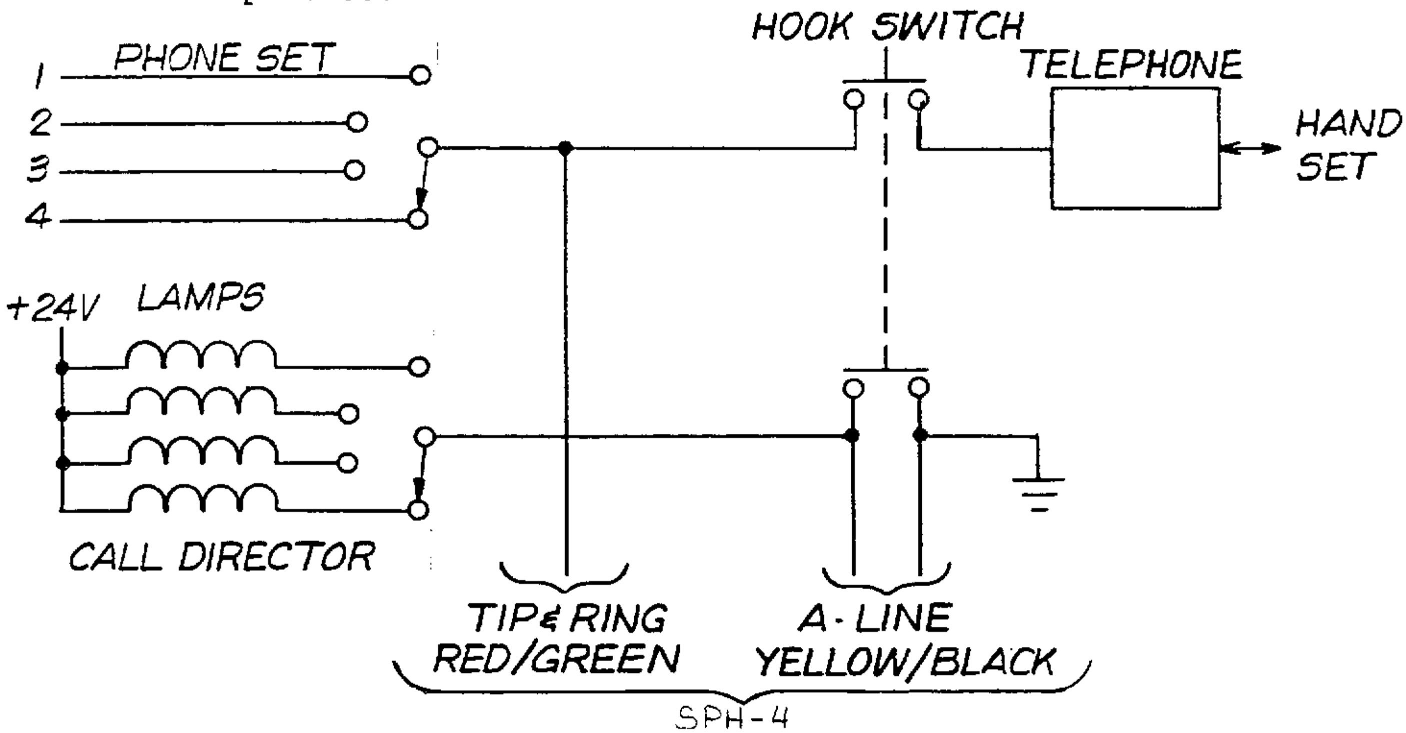


Diagram A-7. A-lead closure: SPH-4 telephone system.

A slim-wire phone system (usually referred to as digital) accomplishes the same task with 4 pairs of wire or less. This is cone with the help of a microprocessor at the EKSU (Electronic Key Service Unit). When a line is requested by the telephone, the following occurs:

1. The telephone sends a digital signal to the microprocessor telling it which line to route to the phone.

2. If the line is available, the EKSU routes the tip and ring to the phone.

This type of phone system can be represented as follows:

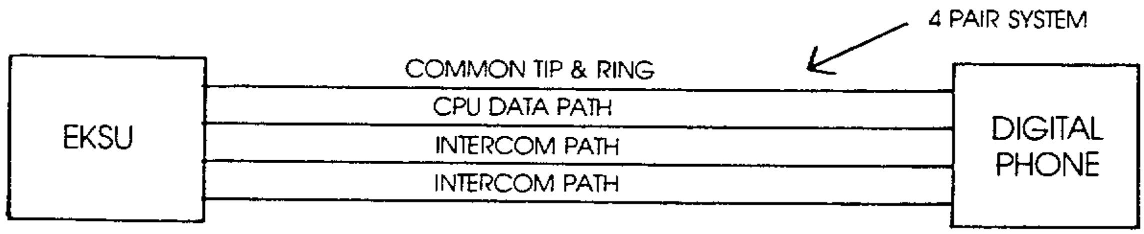


Diagram A-8. Slim-wire (digital) phone system.

The SPH-4 can be installed by routing tip and ring through the unit:

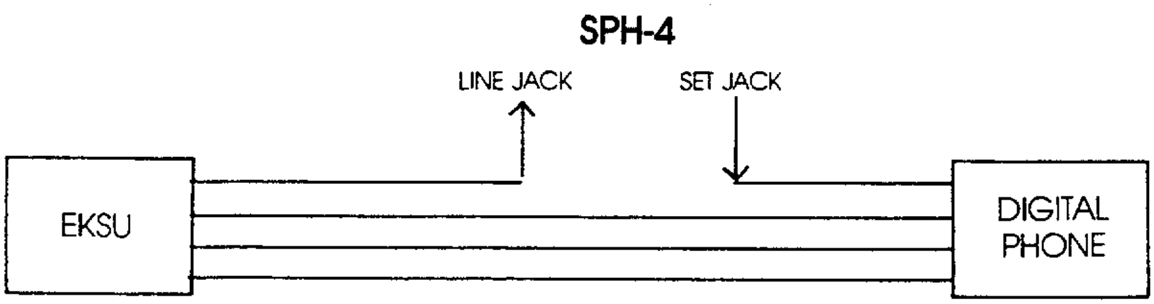


Diagram A-9. SPH-4 connection in digital system.

To install the SPH-4 in a digital telephone system, refer to Appendix B (next page).

9. Appendix B: Digital Telephone Installation.

All Telephone Systems can be grouped into two categories:

1. Fat Wire and

2. Slim Wire.

1. The Fat Wire Telephone System. This system is named because of its 25 pair cable or "fat" wire that comes into the back of the set. This system simply switches the appropriate telephone line to the telephone set. The Key Service Unit (KSU) provides signaling (turns on the lamps on all the phones), holding of lines, and ringing of telephones.

It is important to note that all telephone lines are brought to each telephone and the appropriate line is simply selected (or call directed) to a telephone set. The KSU is only secondary to the telephone; that is each telephone really functions as its own system and is only supported by the KSU. The following diagram will illustrate this:

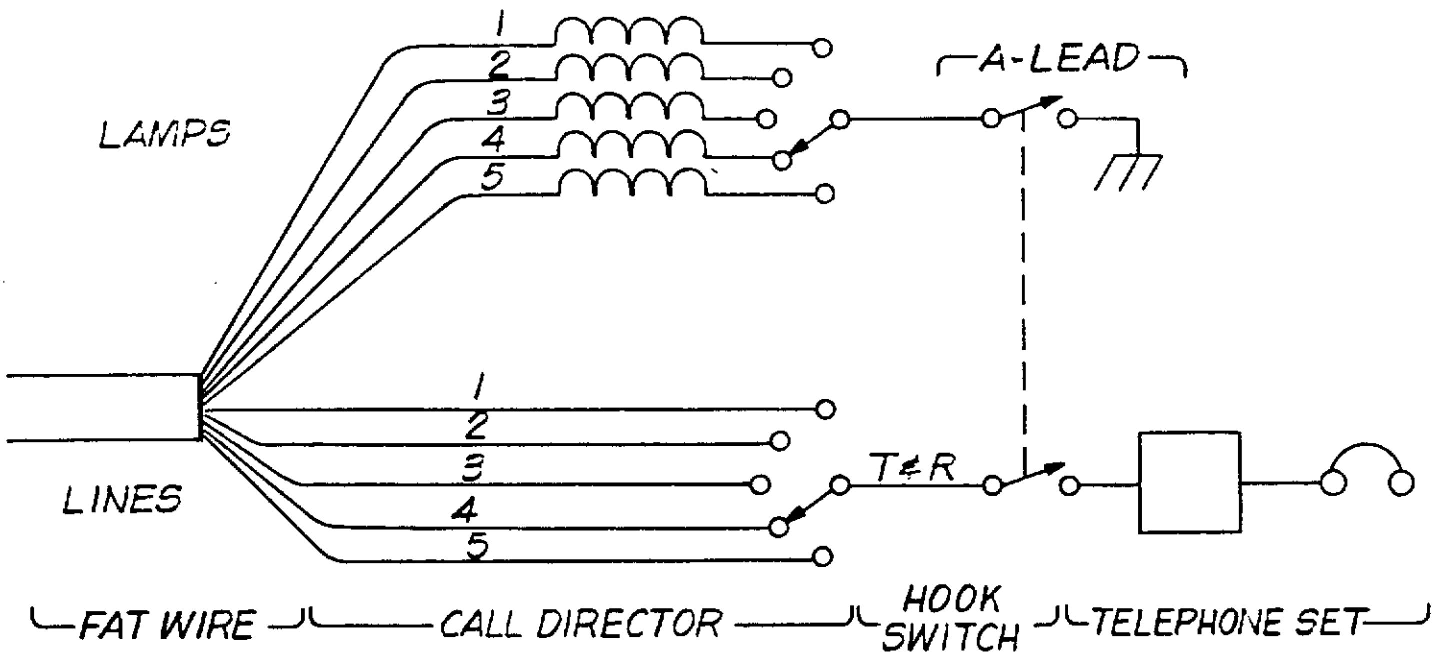


Diagram B-1. Fat Wire telephone system.

2. Slim Wire Telephone System. In contrast, the slim wire telephone system is a fully integrated system; the telephone and KSU are completely dependent on one another. The slim wire system is named for its small or "slim" cable that enters the rear of the set. This cable contains two to four cable pairs. These pairs are used by the system as follows:

1. CPU serial data
2. Common Tip and Ring (Telephone Line)
3. Intercom.

4. Intercom.

The first pair sends data back and forth between the KSU and the telephone set. At the KSU the CPU processes the data and responds as required by the user. The function of the KSU is to route the appropriate tip and ring to the telephone set.

Thus, the slim wire system doesn't need a lot of wires going to the phone system. The following diagram illustrates the slim wire telephone system:

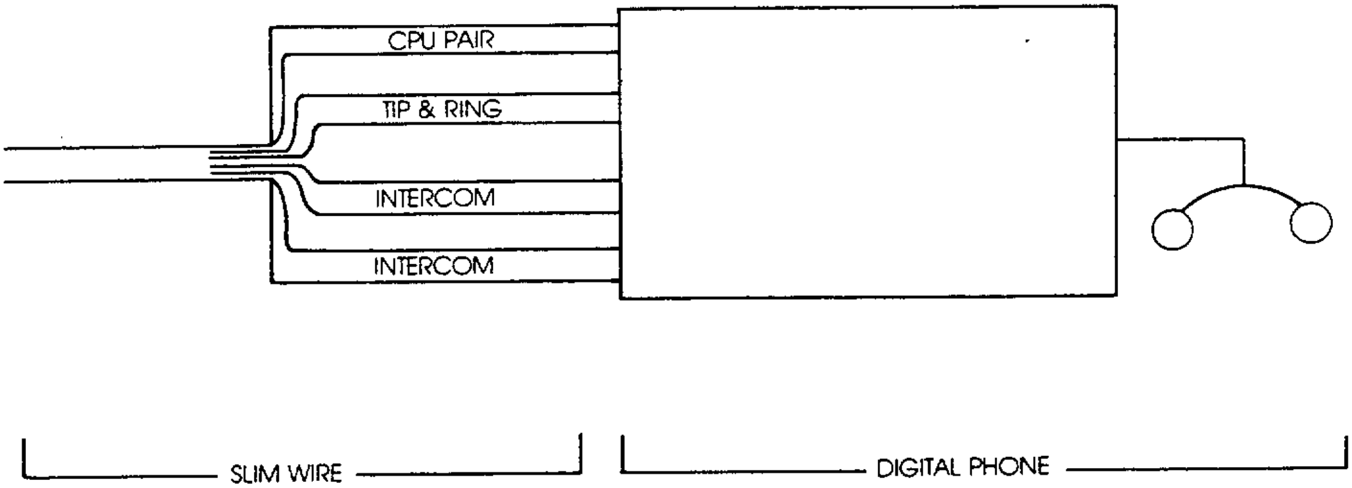


Diagram B-2. Slim Wire telephone system.

How to connect your SPH-4 Telephone System to a "slim" wire Telephone System: (Refer to Diagram B-3)

1. Locate the common tip and ring pair coming from the KSU. The best place to locate it is inside the telephone set. You may want to use your SPH-4 to accomplish this by connecting a speaker up to the monitor out, turning the unit on, and using the "LINE" jack input to search for the line.

2. Disconnect tip and ring going to the set.

3. Connect the "LINE" tip and ring on the SPH-4 (red and green wires) to the common tip and ring coming from the KSU.

4. Connect the "SET" tip and ring on the SPH-4 (red and green wires) to tip and ring going to the telephone set.

With the SPH-4 in the OFF position your telephone will work properly. That's because the tip and ring are simply routing through the unit. However, when the SPH-4 is turned ON, the telephone line is routed to the SPH-4 and the telephone set will no longer be able to take a call.

Remember that to take a call on your telephone you must pick up the handset. Taking the phone "off hook" switches the hook switch telling the KSU your ready to take a call. You must then make a decision on whether you want to pick up the handset every time you want to take a call on the SPH system. (Remember, the KSU only has one way of knowing it's time to send tip and ring.)

It is possible to use the A-lead closure on the "LINE" jack to simulate off hook. **HOWEVER, THE SPH-4 WILL FUNCTION JUST FINE WITHOUT ANY A-LEAD HOOK-UP EXCEPT THAT YOU WILL HAVE TO TAKE THE HANDSET OFF-HOOK TO MAKE THE TELEPHONE COMMUNICATE WITH THE KSU.**

To use your SPH-4 system without having to take the handset off-hook each time you take a call, do the following:

1. Locate the hook switch inside the phone.

2. If it is a normally open single pole single throw (SPST) switch, the A-Lead closure can just parallel the switch.

3. If it is a multi-pole and/or a multi-throw switch, you will have to use a relay inside the set to simulate off hook. You can use the A-lead closure to pull the relay in and the +15VDC regulated supply on J6 to power the relay.

4. If you have any problems please call for service at (801) 268-1117.

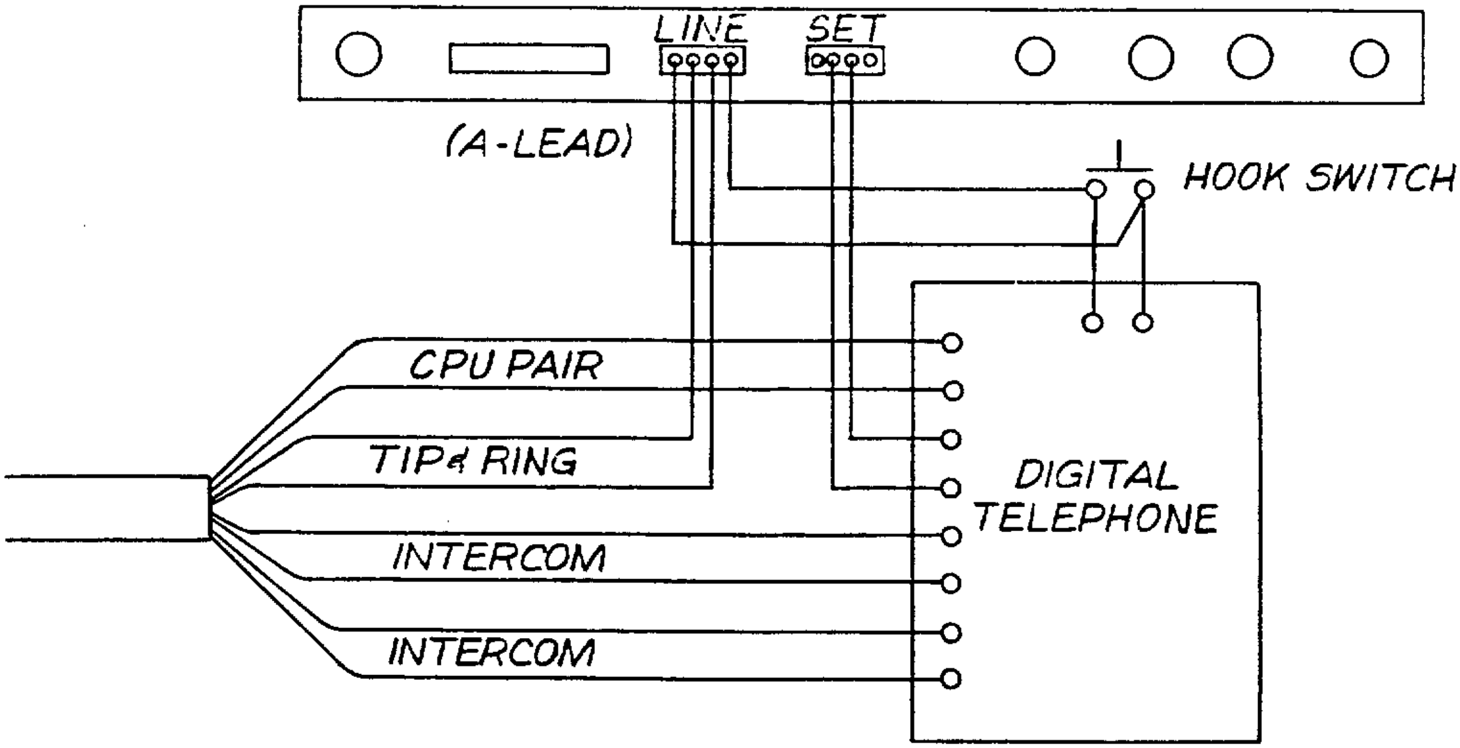


Diagram B-3. SPH-4 Installation in slim wire system.

Wire connections in slim wire system:

LINE JACK

RED } TIP AND RING FROM KSU
GREEN }

YELLOW } CLOSURE TO HOOK SWITCH*
BLACK }

SET JACK

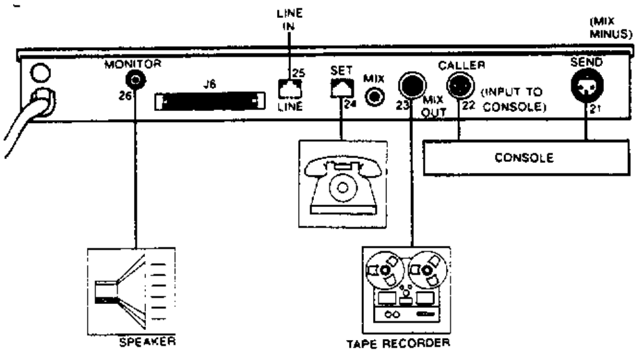
RED } TIP AND RING TO TELEPHONE SET
GREEN }

YELLOW } NOT USED
BLACK }

*THIS CAN ONLY BE USED IF HOOKSWITCH IS A SPST SWITCH. OTHERWISE, THE HOOKSWITCH CAN BE REPLACED WITH A MULTI POLE RELAY AND A-LEAD CAN BE USED TO OPERATE THE RELAY.

IF THE HOOKSWITCH IS NOT REMOTED, THE SYSTEM WILL OPERATE PROPERLY EXCEPT THE OPERATOR WILL BE REQUIRED TO REMOVE THE HANDSET WHEN TAKING CALLS.

10. Appendix C - SPH-4 Applications.



Broadcast use. Numbers refer to designation on line drawing.

#21 Mix minus audio from your console (or a MIC preamp output)—should not contain any of the caller's audio.

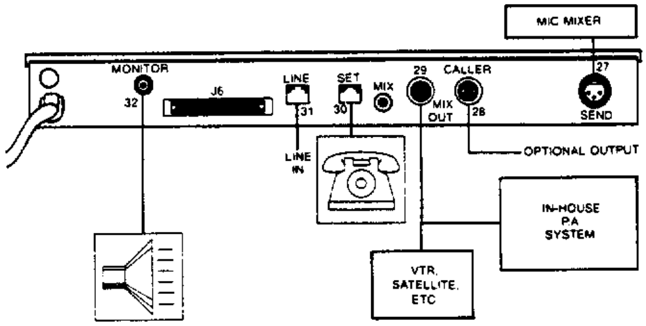
#22 This is the caller's audio for live use.

#23 When used with the MIC preamp output, this permits recording of both ends of the call (at the same level) for later use.

#24 Your telephone plugs in here. The set will not work when the unit is in use.

#25 Connection from your telephone system here.

#26 Plug your speaker in here and place in studio for monitoring callers.



Teleconferencing. #s Refer to line drawing.

#27 Local talent audio appears here.

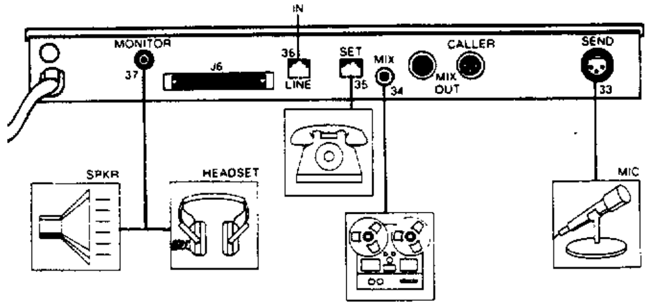
#28 Optional output.

#29 Connection to in-house public address system so your local audience can hear both the talent and the callers. This output may be fed to a VTR, satellite, etc. for long distance conferencing.

#30 Your telephone plugs in here. The set will not work when the unit is in use.

#31 Connection from your telephone system here.

#32 Plug in speaker to be located next to talent (so the talent can hear the callers).



Recording of telephone interviews. Refer to drawing.

#33 Connection for interviewer's MIC. A simple modification is required for a MIC level input.

#34 Both sides of the conversation are recorded at equal levels onto a tape recorder (cassette deck).

#35 Interviewer's telephone set plugs in here. It will not work when the unit is in use.

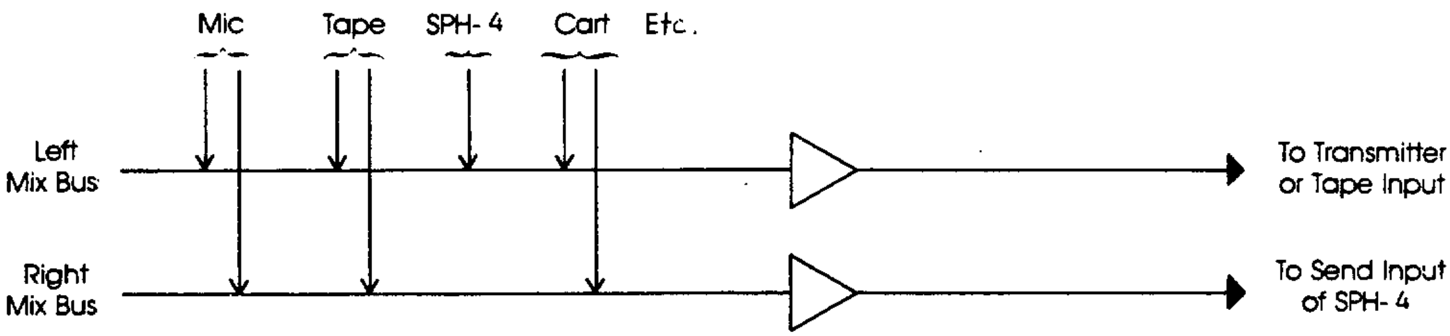
#36 Connection from your telephone system here.

#37 Speaker or headset plugs in here. If the speaker is used, a headset is not needed.

11. Appendix D - Set-up Options.

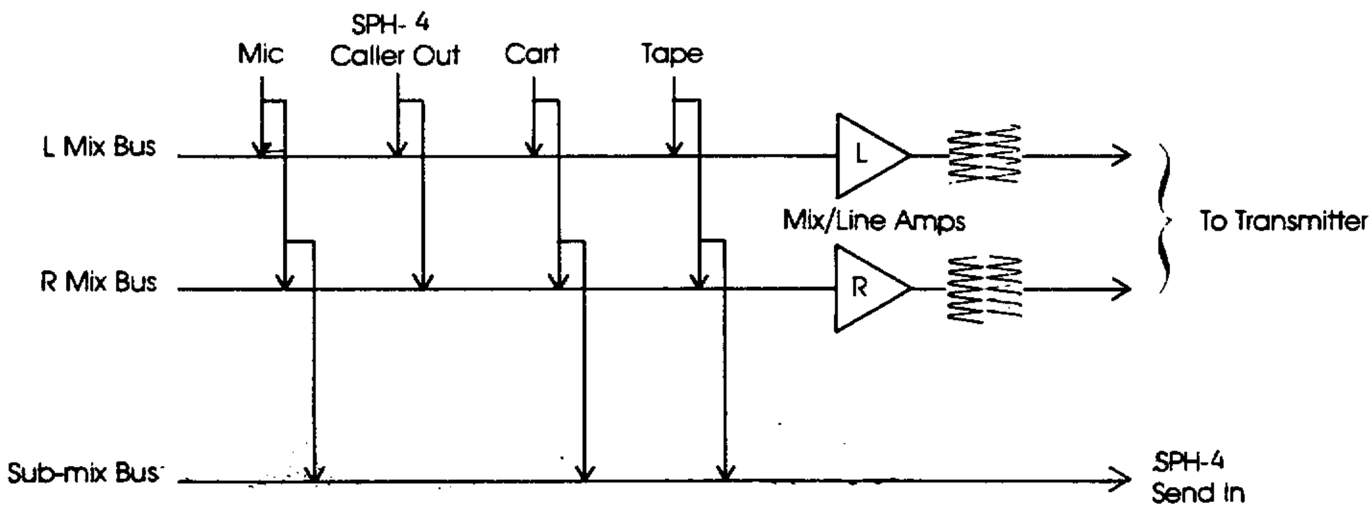
Option 1 - Use of Stereo Board. (See Diagram)

Feed all inputs except the SPH-4 to both the left and right channels of the console. Connect the caller output to the left channel only of the console and use the right channel of the board to feed the phone hybrid.

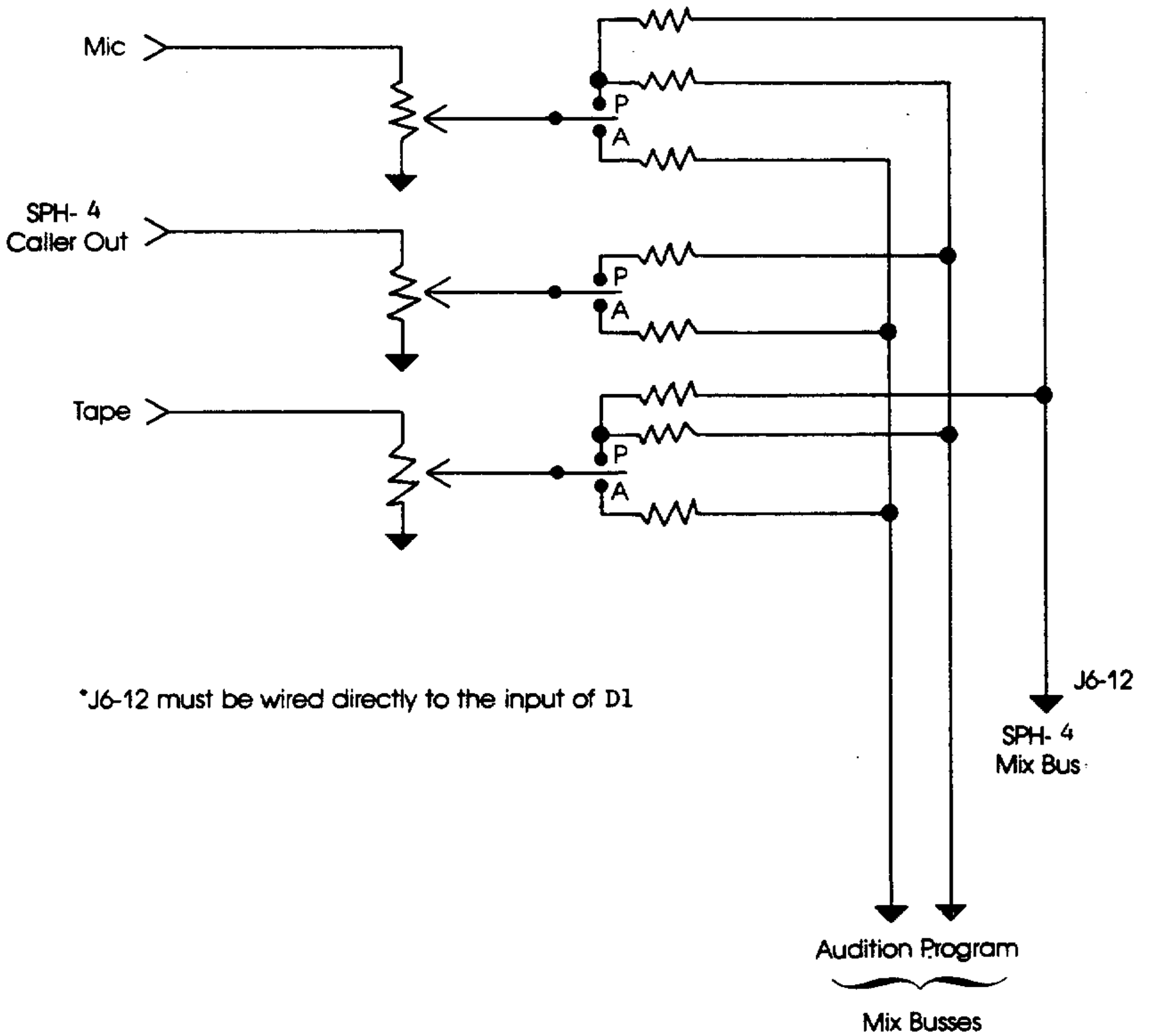


Option 2 - Submix Bus. (See Diagrams)

This option requires a modification both to the SPH-4 and to your console. Although it is the most difficult to set-up, it is the easiest to use. The idea is to create a separate submix bus of all audio that is input to the console, except the caller audio:



To implement this option, install mixing resistors to the output of your console's faders (past the program/audition switches). Create a shielded mix bus to be fed to J6-12 of the SPH-4. Now remove the PC mounted auxiliary 10K resistor and place a jumper so that J6-12 feeds directly into D1. This input into D1 provides a zero impedance mix point for your submix bus. The send level can be adjusted via the "FEED" pot on the front panel.

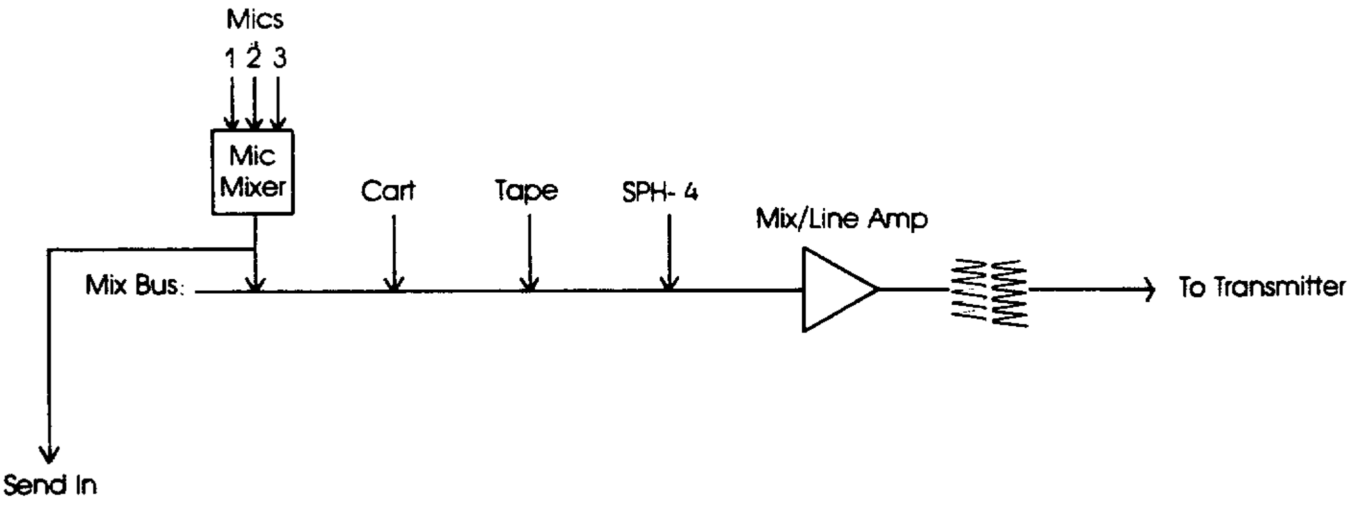


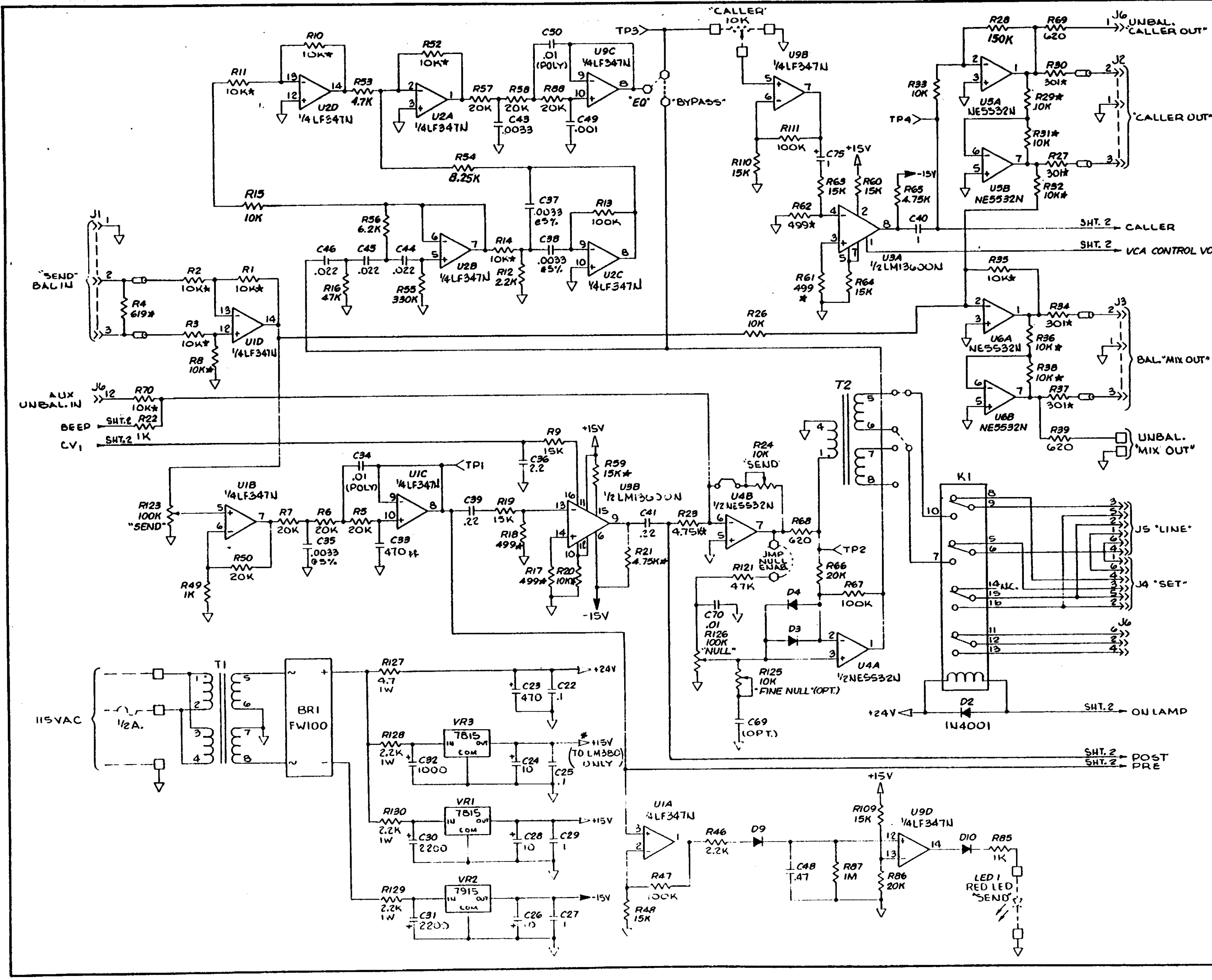
Option 3 - Down Stream Mixing:

This option mixes the caller into your console audio after it has the console. Connect the output of the console to the "SEND" input and connect the "MIX" out to your tape, transmitter, or whatever. This will mix the caller audio with the send audio. When the switch is in the "off" position, the send to mix amplifiers remain active. Incidentally, the send to mix amplifiers are very transparent offering a flat frequency response (.1 Db maximum change from 20hz to 30khz), low distortion (less than .01%) and excellent noise figures.

Option 4 - Mic Mixer Feed Option

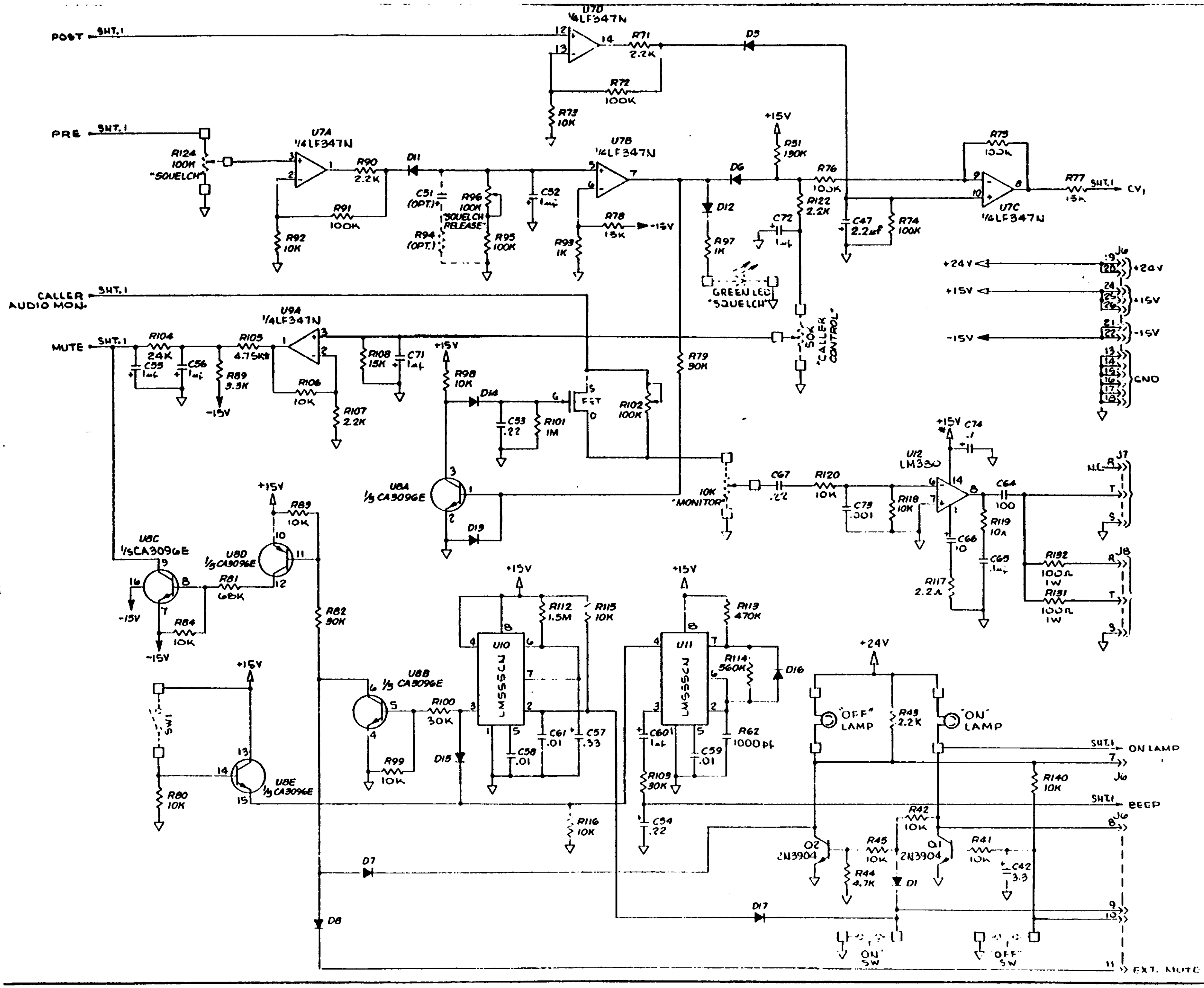
This is a very simple option. Just pre-mix all of your mix sources with an external mic mixer and route the output to the SPH-4.





NOTES:
1. UNLESS OTHERWISE SPECIFIED:
   A. FIXED RES. ARE 1/4 WATT.
   B. RES. VALUES ARE IN OHMS ±5% TOL.
   C. ★ INDICATES ±1% TOL.
   D. CAP. VALUES ARE IN MICROFARADS.
   E. VOLTAGES ARE D.C.
   F. DIODES ARE IN4148.
2. O- INDICATES FLEE CLIPS.
3. DOTTED LINES ARE PARTS NOT MTD. DIRECTLY ON BOARD.
4. U12 HAS GND TO PINS 2—5 & 8—13.
5. LF347N'S HAVE +15V TO PIN 4 & -15V TO PIN 11.
6. NE5532N'S HAVE +15V TO PIN 8 & -15V TO PIN 4.
7. DECOUPLING CAPS. NOT SHOWN ARE:
   A. +15V

GENTNER ENGINEERING COMPANY INC.			
TITLE: ELECTRICAL SCHEMATIC, SPH-4			
DRAWN: CTD/JC. 2-8-84	CHECKED:	APPROVED:	
DWG NO. 003-01-01	SIZE B	SHT. 1 OF 2	REV. A



GENTNER ENGINEERING COMPANY INC.

TITLE: ELECTRICAL SCHEMATIC, SPH-4			
DRAWN: CTD/JC 2-8-84	CHECKED:	APPROVED:	
DWG NO. 003-01-01	SIZE D	SHT. 2 OF 2	REV. A