



SPH3A

*Installation and
Operations Manual*

Contents

Introduction	1
Warranty Registration	1
Unpacking	1
Features and Benefits	2
Applications	2
Teleconferencing	2
Broadcast	2
Recording Telephone Interviews	3
Product Description	3
SPH3A	3
Front-Panel Controls	3
Back-Panel Connectors	4
Before You Install	5
Acoustic Room Treatment	5
Power Requirements	5
Telephone Line Requirements	5
Equipment Placement	5
Installation	5
Step 1 — Back-Panel Connections	5
Step 2 — Connect Caller Audio	6
Step 3 — Connect Send Audio	7
Step 4 — Determine Telephone A-Line	7
Step 5 — Connect Telephone Line	8
Calibration	11
Send Level Adjustment	11
Hybrid Nulling	12
Operation	13
Local Operation	13
Remote Operation	13
Helpful Hints	14
SPH3A Teleconferencing	14
Specifications	16
Warranty	17
Safety Information	17
Appendix A: Setup Options	18
Stereo Board Option	18
Submix Bus Option	18
Downstream Mixing Option	19
Mic Mixer Feed Option	19
Mic Input Option	19
Appendix B: Telephone Basics	20
Definition: Telephone Line.	20
Telephone-Line Processing	20
Tip and Ring: A More Detailed Look	20
Single-Line Telephone Systems	21
Multiline Telephone Systems	22
Appendix C: Technical Description	24
Appendix D: Schematics	26
Appendix E: Block Diagram	27

List of Figures

Figure 1.	Equipment diagram	1
Figure 2.	Broadcast application diagram	2
Figure 3.	Teleconferencing application diagram	2
Figure 4.	Interview-recording application diagram	3
Figure 5.	Basic setup block diagram	3
Figure 6.	Front-panel controls	3
Figure 6a.	Front-panel controls	4
Figure 7.	Back-panel connectors	4
Figure 8.	J6 cable connection	5
Figure 9.	Back-panel connectors	6
Figure 9a.	Back-panel connectors	8
Figure 10.	Fat-wire system diagram	9
Figure 11.	Slim-wire connection to a digital phone system	10
Figure 12.	Front-panel controls	11
Figure 12a.	Front-panel controls	12
Figure 12b.	Front-panel controls	14
Figure 13.	Two-SPH3A configuration	14
Figure 14.	Multiple-SPH3A configuration	15
Figure 15.	Stereo-board diagram	18
Figure 16.	Submix-bus diagram	18
Figure 17.	Submix-bus modification diagram	28
Figure 18.	Downstream-mixing diagram	19
Figure 19.	Mic-mixer feed diagram	19
Figure 20.	Typical telephone line	20
Figure 21.	Two wire-to-four wire telephone line conversion	20
Figure 22.	Telephone transmission system	21
Figure 23.	USOC RJ11C pinout	21
Figure 24.	Single-line telephone set	21
Figure 25.	Multiple-line call direction	22
Figure 26.	SPH3A phone system A-lead closure	23
Figure 27.	Slim-wire digital phone system	23
Figure 28.	SPH3A installation in a slim-wire system	23

List of Tables

Table 1.	Slim-Wire System Connections	11
Table 2.	J6 Auxiliary Input Pinout	15

Introduction ≡

Congratulations on purchasing the SPH3A Telephone System. The SPH3A is a complete telephone interconnection system. It provides two wire-to-four wire connection and hybrid functions for a variety of applications.

This manual explains how to install, set up and operate your SPH3A. It also provides instructions on how to improve room acoustics and resolve minor technical problems, should any arise.

If you need information on how to install, set up or operate your system, please contact Gentner Communications Corporation at the location noted below. We welcome and encourage your comments so we can continue to improve our products and serve your particular needs.

Gentner Communications Corporation

1825 Research Way
Salt Lake City, Utah 84119

TEL: Worldwide (801) 975-7200 In U.S.A. (800) 945-7730

FAX: Worldwide (801) 977-0087 In U.S.A. (800) 933-5107

FAX-On-Demand 24-Hour Information Service (800) 695-8110

Worldwide Web Page @ <http://www.gentner.com>

**Warranty
Registration** ≡

Please register your SPH3A by completing the self-addressed, postage prepaid warranty registration card and return it to Gentner Communications by mail. You may also FAX it to the above listed fax number or call Gentner Communications. When your product is properly registered, Gentner Communications will be able to serve you better should you require technical assistance or desire to receive upgrades, new product information, etc.

Unpacking ≡

Ensure that the following equipment (See Figure 1, below.) was received with your shipment:

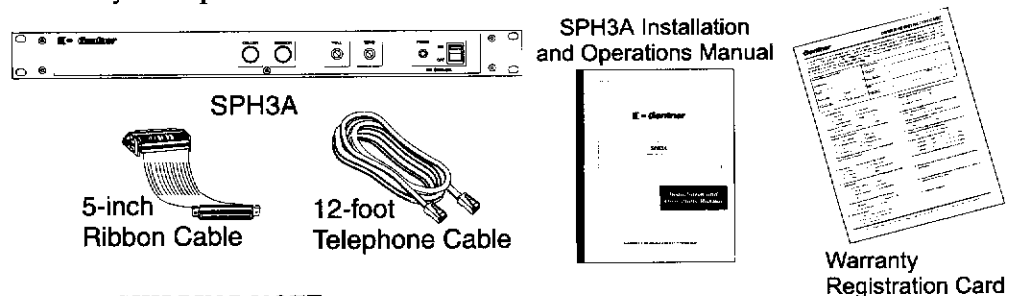


Figure 1. Equipment diagram

SHIPPING NOTE:

Gentner Communications is not responsible for product damage incurred during shipment. You must make claims directly with the carrier. Inspect your shipment carefully for obvious signs of damage. If the shipment appears damaged, retain the original boxes and packing material for inspection by the carrier. Contact your carrier immediately. This manual provides instructions for the use of the Gentner SPH3A Telephone System. This manual should be totally read and understood before the unit is installed.

Features and Benefits

- Continuous caller audio
- 2W power amp (no need for headsets on guests or talent)
- Circuitry protected against damage caused by short circuiting
- Balanced inputs and outputs
- Cross-feeding for multiple caller use
- Simple installation and operation (no special equipment required)
- Downstream mixing capability
- Compatible with any telephone system, including digital
- Connects directly to the telephone line
- No outside RF interference will affect performance
- High quality "speakerphone" functions

Applications

Broadcast

This application (See Figure 2, below.) requires mix-minus audio from your console (or a mic preamp output), which should not contain any of the caller's audio. The caller's audio is available for live use. When used with the mic preamp output, recording of both ends of the call (at the same level) for later use is also available. Your telephone will not work in this application when the unit is in use.

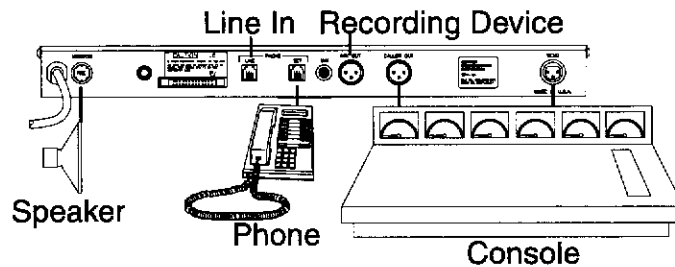


Figure 2. Broadcast application diagram

Teleconferencing

This application (See Figure 3, below.) allows connection to your in-house public-address system so your local audience can hear both the talent and the callers. Output may be fed to a VTR, satellite, etc. for long-distance teleconferencing, as well. Your telephone will not work when the unit is in

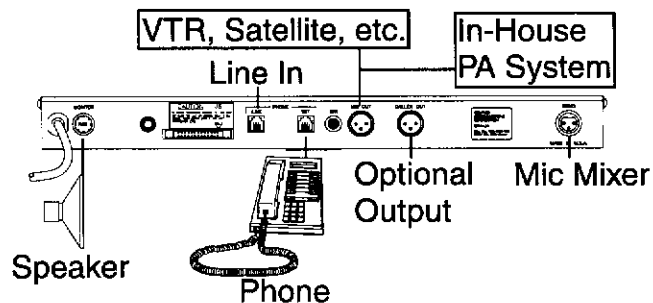


Figure 3. Teleconferencing application diagram

use.

Recording Telephone Interviews

In this application (See Figure 4, below.), a simple modification is required for a mic-level input (See Mic Input Option, Page 19.), so both sides of the conversation can be recorded at equal levels with a recording device. The interviewer's telephone is plugged in, although it will not work when the unit is in use. If a speaker is used, a headset is not needed.

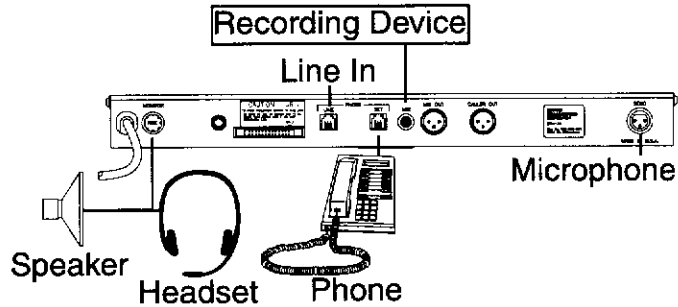


Figure 4. Interview-recording application diagram

Product Description

SPH3A

The SPH3A was designed for ease in operation. Operators have a single switch and two knobs for control. Of the five internal adjustments for setup, two are located on the front panel.

The SPH3A uses the telephone line on one leg of a Wheatstone bridge to achieve isolation of send and receive audio while interfacing with the rest of the existing telephone system (Figure 5, below).

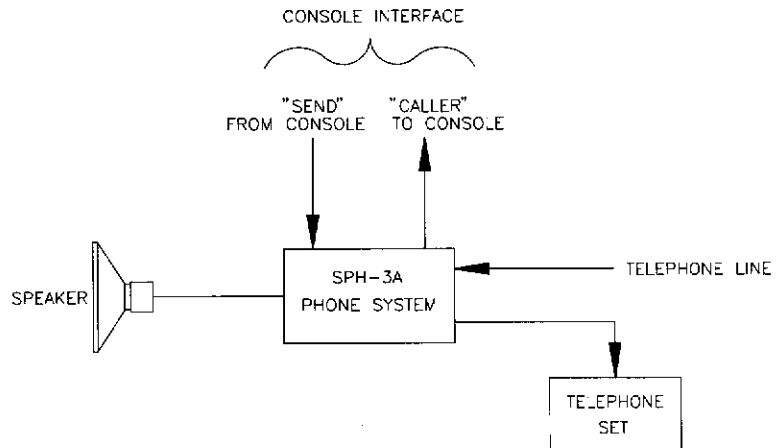


Figure 5. Basic setup block diagram

Front-Panel Controls

The SPH3A front-panel controls (See Figure 6, below.) perform the following functions:

1. *Caller.* Controls the volume of the incoming caller's voice.
2. *Monitor.* Controls the volume of the incoming caller's voice at the speaker

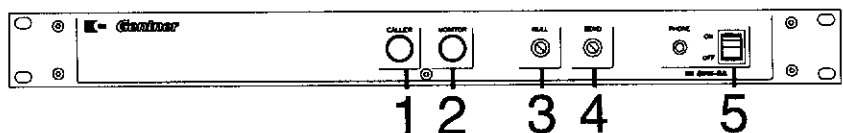


Figure 6. Front-panel controls

Product Description
Continued ≡

only (Figure 6a, below). Keep this level low enough that the caller's voice does not echo or sound "hollow" by acoustically coupling between the microphone and the speaker.

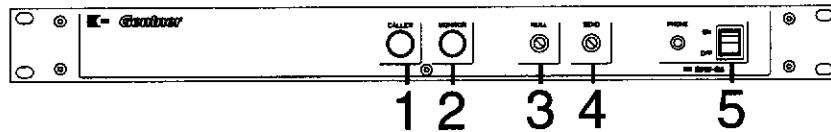


Figure 6a. Front-panel controls

3. *Null*. This trim pot is used to adjust the separation between send and caller audio. This adjustment may need to be reset occasionally to account for signal drift.

4. *Send*. This control adjusts the feed-audio volume heard by the caller. This is a one-time setup adjustment and should not be adjusted under normal operating conditions. It is typically adjusted to just less than the normal volume heard by the caller.

5. *Phone On/Off*. This is not a power switch. Moving the switch to the ON position will:

- Connect the SPH3A to the phone line
- Turn on the light on the phone (if applicable)
- Unmute the monitor speaker so you can hear the caller
- Unmute the caller audio going to your equipment

Back-Panel Connectors

1. *Monitor*. This is a 1/4" tip-ring-sleeve phone jack (Figure 7, below). This output will drive a 2W speaker and is used to listen to the caller without having to wear a headset.

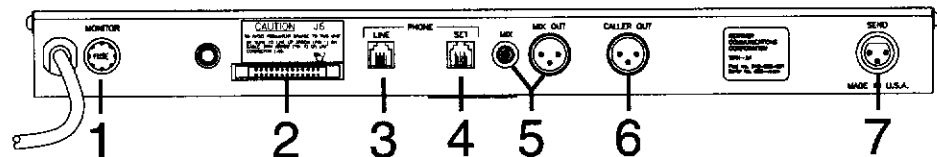


Figure 7. Back-panel connectors

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, and yellow and black are A-lead.

4. *Set*. This modular jack can be connected to a single-line phone. When the SPH3A is in the OFF position, the phone line will appear at the telephone. When the front toggle switch is ON, the telephone will not work.

5. *Mix/Mix Out*. These two outputs (MIX an RCA phono, MIX OUT an XLR) contain a mixture of send and caller audio. Even when the toggle switch is OFF, send audio will appear here.

6. *Caller Out*. 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.

Before You Install

The SPH3A is designed to work in almost any acoustic environment. However, to maximize your audio quality, prepare your site by taking the following factors into consideration:

Acoustic Room Treatment

Simple acoustic room treatment is recommended to improve the operation of your system. Rooms that have large areas of windows, white boards, hard floors, etc., are acoustically "live." These areas increase the amount of audio reverberation in the room which, in turn, reduces audio quality. You should minimize the amount of audio reverberation where possible.

You can improve room acoustics by installing acoustic panels, drapes and other wall fabrics. Another way to improve overall room acoustics is to keep room noise (i.e. computers and fans) to a minimum.

Power Requirements

The SPH3A's internal power transformer accommodates voltage requirements of 120Vac, 60Hz, 7W.

Telephone Line Requirements

Your SPH3A works on standard telephone lines and connects to the telephone system with a standard RJ11C modular jack. If you do not have an RJ11C jack where you want to install your SPH3A, call your telephone company for installation.

Equipment Placement

The SPH3A is designed for mounting within a standard 19" rack.

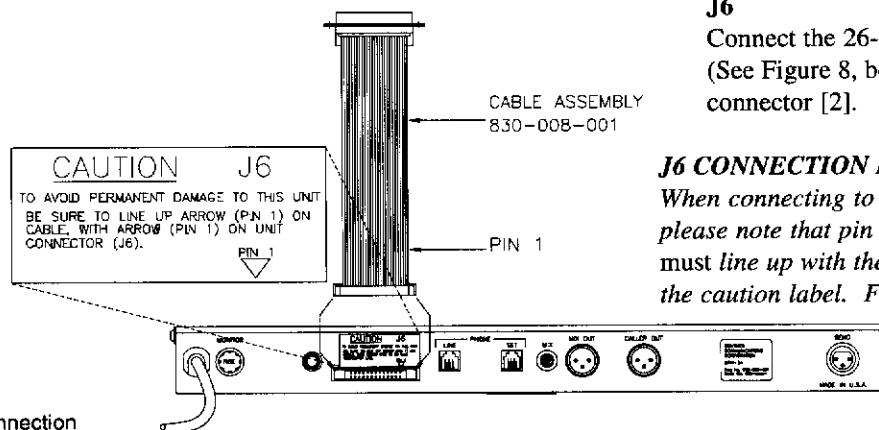
Installation

The SPH3A was designed for easy installation and setup. All necessary interface connections are made through rear panel XLR and modular phone connectors. This allows for easy installation, removal, and service. To install your SPH3A, follow these step-by-step instructions:

Step 1 — Back-Panel Connections

Monitor

Connect a speaker or a headset between the tip-ring and the sleeve [2] (Figure 7, previous page).



J6

Connect the 26-pin cable connector (See Figure 8, below.) to the J6 connector [2].

J6 CONNECTION NOTE:

When connecting to the J6 connector, please note that pin 1 on the cable must line up with the pin 1 arrow on the caution label. Failure to do so will

Figure 8. J6 cable connection

Installation
Continued ≡

result in permanent damage to your SPH3A unit, and you will be charged for resulting repairs.

Line.

Connect a standard, modular phone line to the LINE connector [3] (Figure 9, below). The red and green wires are tip and ring, yellow and black are A-lead.

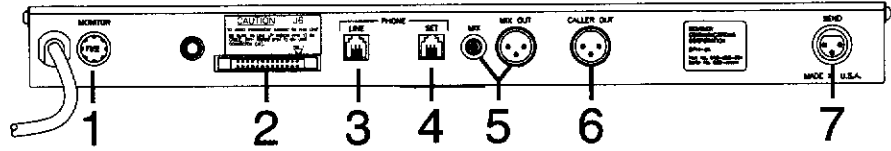


Figure 9. Back-panel connectors

Set

Connect the SET connector [4] to the modular jack on a single-line telephone. When the PHONE switch is OFF, the telephone line will appear at the telephone; when the PHONE switch is ON, the telephone will not function.

Step 2 — Connect Caller Audio

There are five output option for caller audio: balanced mix out, balanced caller out, unbalanced caller out, monitor out and equalized caller out. All receive audio (caller audio) is controlled by the front-panel control marked CALLER and is muted whenever the front-panel PHONE switch is in the OFF position.

Balanced Mix Out

This output is a mix of send and caller audio. Send and mix amplifiers remain active even when the phone switch is in the OFF position. Connect to pins 2 and 3 of the MIX XLR [5] (See Figure 9, above.) for a balanced output and to pins 1 and 2 or 1 and 3 for unbalanced output. The send-to-mix gain is unity. An unbalanced RCA phono connector is also provided on the rear panel.

The mix output should be used in applications that require downstream mixing of the caller audio (See Downstream Mixing Option, Page 20.) or for recording interviews directly onto tape.

Balanced Caller Out

This is a balanced output of caller audio only. Connect to pins 2 and 3 for a balanced output and to pins 1 and 2 or 1 and 3 on the MIX XLR [5] for unbalanced output.

Unbalanced Caller Out

This is another caller-only output that is unbalanced. Connect to pin 1 on the J6 connector [2].

Monitor Out

The MONITOR [1] output contains severely equalized caller-only audio and should be connected to an 6ohm speaker. This output will be muted whenever the PHONE switch is in the OFF position. The equalization in this output is designed to eliminate any possible chance of feedback through the monitor speaker.

SPEAKER NOTE:

It is important to place the MONITOR-attached speaker in an acoustically

strategic position so that maximum isolation between microphone and speaker is permitted. See Acoustic Room Treatment (Page 5) for further suggestions.

Since this equalized audio contains low energy audio (3kHz and below) it will clip at a relatively low listening level. This is normal and was designed into the system to ensure maximum null dependability and minimum operator error.

Equalized Caller Out

This contains the same audio as the MONITOR out but low level and unmuted. This output can be used to cross-feed another SPH3A for multiple-caller use. Connect to pin 19 on J6 connector [2].

We suggest that you make all audio connections before attempting installation of the phone line.

Step 3 — Connect Send Audio

This is the audio that the caller will hear. This audio cannot contain any of the caller's own audio.

SEND-RECEIVE ISOLATION NOTE:

Refer to Appendix C (Page 24) for suggestions on how to isolate send from receive audio.

Step 1

Connect the send audio to pins 2 and 3 on the SEND XLR [7] (Figure 9, previous page). The front-panel SEND trim pot and is adjusted with a screwdriver. Once this adjustment is made, it should not be changed.

UNBALANCED SEND AUDIO NOTE:

If this audio is unbalanced, connect between pins 1 and 2 or between pins 1 and 3 on the SEND XLR.

Step 2

Adjust the send audio for the minimum acceptable level. This will accomplish two things: improve the send-to-receive null and make the caller speak louder.

The feed amplifiers do a considerable amount of equalization before the audio hits the phone line. This equalization is required so the unit can pass FCC registration requirements. As a result, send audio may sound slightly muffled. This is normal and aids in the nulling process. If high quality feeding is required (for network newsfeeds etc.), the equalization can be bypassed (Mic Mixer Feed Option, Page 19).

UNBALANCED AUXILIARY INPUT NOTE:

An unbalanced auxiliary input is also provided at pin 12 of the J6 connector [2]. The level-control trim pot for this input is located on the printed circuit board.

Step 4 — Determine Telephone A-Line

A-lines operate by a closure provided on the black and yellow wires of the modular jack. To find the A-line we suggest the following:

A-LINE NOTE:

For single-line applications, you do not need to worry about connecting the A-

Installation
Continued ≡

line. For multiline phones with call directors, it is important to properly connect the A-line.

Step 1

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

Step 2

With the phone cover removed, short different combinations of terminals on the phone until the light on the line you punched lights up. Once you have found the connection, try punching different lines to ensure the A-line follows. If it does not, 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.

Step 3

Connect the terminals you found on the phone to the black and yellow wires on the modular jack.

Step 4

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

For Single-line Telephones. Make sure the handset is on the cradle. When you select the PHONE switch to the ON position, you will hear dial tone through the speaker or at the CALLER OUT port [6].

For Multiline Telephones. Make sure the handset is on the cradle. Punch any line on the call director and place the PHONE switch to the ON position. You will hear dial tone through the speaker or at the caller output port, and 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 SPH3A takes direct control of the phone. When using the SPH3A, make sure the handset is always in the cradle. If it is not, the unit will not null properly.

Step 5 — Connect Telephone Line

TELEPHONE LINE NOTE:

If the SPH3A is not connected directly to the phone line it will not null properly.

Analog Telephone Line

The analog phone line should be connected to the modular LINE connector [3] (Figure 9a, below). If you are ordering a connection from the telephone company, specify a USOC RJ11C jack (standard modular telephone jack).

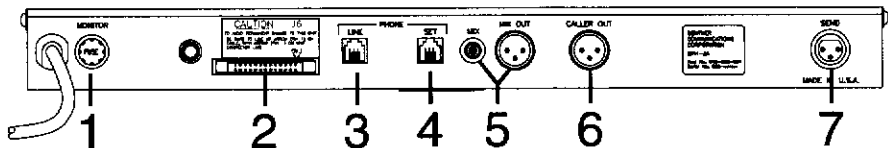


Figure 9a. Back-panel connectors

If you do not have a modular connection at the telephone line, and cannot easily locate the phone line in your telephone, we suggest the following:

Verify Audio Connections. Ensure that all audio connections are wired in. Listen either to the speaker output or to a caller output port by turning the CALLER trim pot to 12 o'clock and switching the PHONE switch to the ON

position.

Connect Leads. Connect clip leads to the SPH3A input phone line (the red and green wires on the modular jack). By trial and error, connect different combinations in the phone until you locate a phone line (the SPH3A will terminate the line and you will hear dial tone). For call directors, try punching different lines to ensure you are on the call director's output.

Digital Telephone Lines

Digital telephone systems are grouped into two categories: fat wire and slim wire.

Fat Wire. This system is named because of its 25-pair cable ("fat" wire) that connects to the back of the set. This system (See Figure 10, below.) simply switches the appropriate telephone line to the telephone set. The Key Service Unit (KSU) provides signaling (lights the lamps on all the phones), hold and ringing.

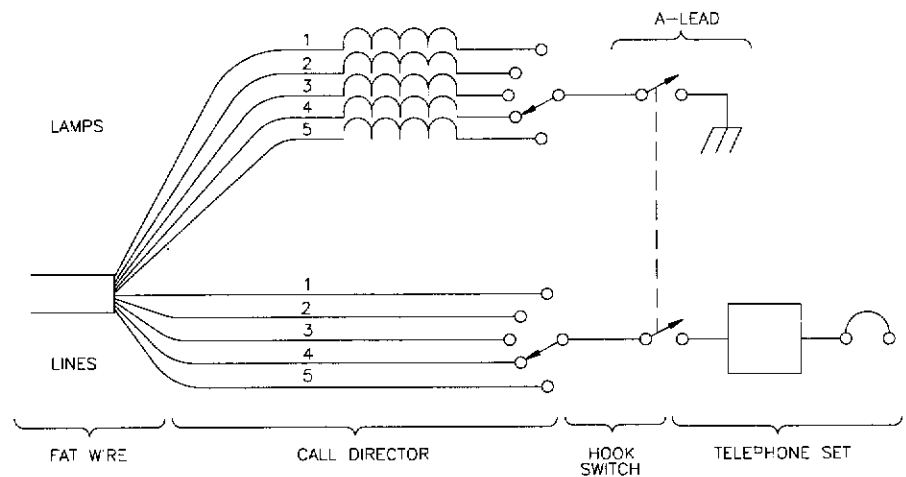


Figure 10. Fat-wire system diagram

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 secondary to the telephone; each telephone functions as its own system and is only *supported* by the KSU.

Slim Wire. In contrast, the slim-wire telephone system is a fully integrated system; the telephone and KSU are completely interdependent. 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 (See Figure 11, below.), depending on the type of system. 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 KSU routes the appropriate tip and ring to the telephone set. Thus, the slim-wire system does not need a lot of wires going to the phone system.

**Installation
Continued** ≡

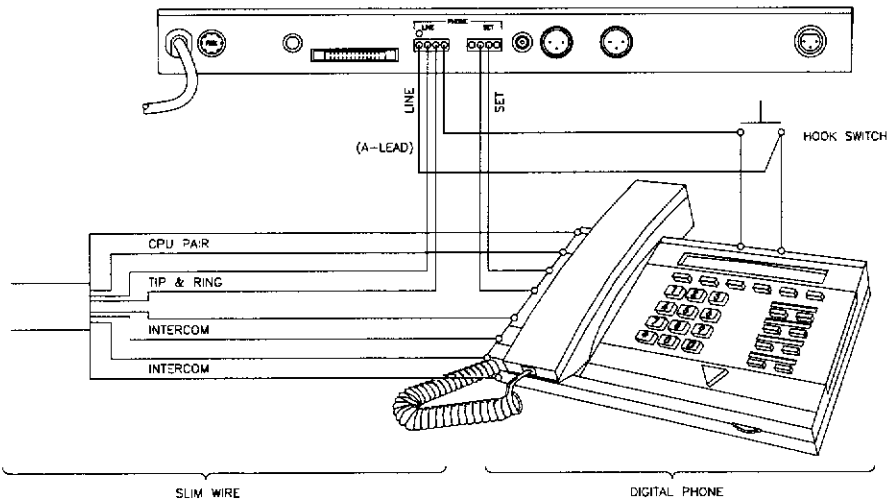


Figure 11. Slim-wire connection to a digital phone system

How to connect your SPH3A to a slim-wire telephone system (Figure 11, above):

Step 1. Locate the common tip and ring pair coming from the KSU. The best place to locate it is inside the telephone set itself. You may want to use your SPH3A to accomplish this by connecting a speaker to the MONITOR out, turning the PHONE switch ON, and using the LINE jack input to search for the line.

Step 2. Disconnect tip and ring going to the set.

Step 3. Connect the LINE tip and ring on the SPH3A (red and green wires) to the common tip and ring coming from the KSU.

Step 4. Connect the SET tip and ring on the SPH3A (red and green wires) to tip and ring going to the telephone set.

When the SPH3A PHONE switch is in the OFF position, your telephone will work properly because the tip and ring are simply routing through the SPH3A. However, when the PHONE switch is turned ON, the telephone line is routed to the SPH3A and the telephone set will no longer be able to take a call.

HANDSET NOTE:

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 you are ready to take a call. You must then decide whether you want to pick up the handset every time you want to take a call on the SPH3A system.

It is possible to use the A-lead closure on the LINE jack to simulate off-hook (so that you do not have to pick up the handset each time). However, the SPH3A will function normally 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 SPH3A without having to take the handset off-hook each time you take a call, do the following:

Step 1. Locate the hook switch inside the phone.

Step 2. If it is normally an open single-pole single-throw (SPST) switch, the A-lead closure can parallel the switch.

Step 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.

If you have any problems, please contact Gentner Communications at the number listed below.

Wire connections in slim wire system:

Table 1. Slim-Wire System Connections

<u>Wire Color</u>	<u>SET Jack Description</u>	<u>LINE Jack Description</u>
Red	Tip and ring to the telephone set	Tip and ring from the KSU
Green	Tip and ring to the telephone set	Tip and ring from the KSU
Yellow	Not used	Closure to the hook switch*
Black	Not used	Closure to the hook switch*

* This can only be used if the hook switch is an SPST switch. If not, the hook switch can be replaced by a multi-pole relay, and A-lead can be used to operate the relay.

DIGITAL INSTALLATION NOTE:

This can only be used if hook switch is a SPST switch. Otherwise, the hook switch can be replaced with a multiple relay and A-lead can be used to operate the relay. If the hook switch is not removed, the system will operate properly, except the operator will be required to remove the handset when taking calls.

Calibration ≡

Send Level Adjustment

This trim pot [4] (See Figure 12, below.) is located on the front panel and controls the send and auxiliary send levels. The unbalanced input send level adjustment is located internally. To adjust send level, do the following:

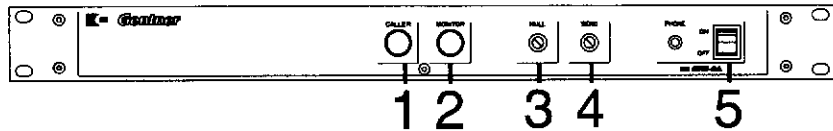


Figure 12. Front-panel controls

Step 1

Call a friend. Turn the SPH3A PHONE switch [5] to the ON position and hang up the phone. Talk on the console at a normal level.

Step 2

Listen to your friend either by bringing him in as an input to the console or on the monitor speaker.

PRECALIBRATION NOTE:

Since the system is not yet calibrated, your friend will either not hear you very well or you will be too loud. In the latter case, feedback may occur or it will sound very hollow.

Step 3

Adjust the SEND trim pot [4] for minimum acceptable level. Make it a bit lower than typical telephone audio. This will help in the nulling process.

Calibration
Continued ≡

Step 4

After you have nulled the hybrid, you may need to reset the send level.

Hybrid Nulling

This is the most critical step in the process of installing the SPH3A. This must be done properly to achieve satisfactory performance from the phone line.

Step 1

Keep your friend on the line while performing this process. Send a 1kHz tone down the line by applying it as an input to the console. Make sure you use a 1kHz tone. If a tone generator is not available, you can simply talk into your microphone and null your voice out through the speaker.

Step 2

Adjust the NULL trim pot [3] (See Figure 12a, below.) for minimum audio coming from the receive side of the hybrid. If you have a speaker connected, adjust the MONITOR control [2] for minimum audio. However, it is best to have a meter on the caller output of the hybrid.

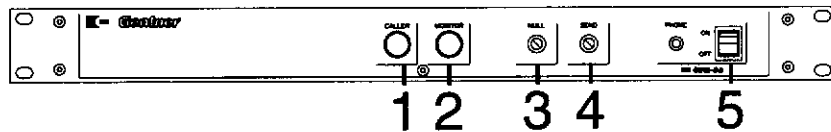


Figure 12a. Front-panel controls

Step 3

Remove the top of the SPH3A. Locate the null capacitor (it is soldered to PC board clips). Using clip leads, try different values for the capacitor. The unit is shipped with a .1uF. Every time you try a different capacitor, renull the NULL trim pot [3] on the front panel. This capacitor is used to match the capacity of line coming from the central office; therefore, it may be different for every installation but, once the hybrid has been optimized, you will find that it will remain nulled for most calls.

NULLING NOTE:

The hybrid will not null properly if the handset is off the cradle or if someone has picked up the phone in another room. The phone line cannot be double-terminated.

Once you have optimized the null, go back and reset the send level.

The SPH3A is now ready for use.

Operation 

Operation of the SPH3A is simple and straightforward. Operators have a single switch and two knobs for operation, and two options for use: local operation and remote operation.

Local Operation

If you use the unit at the console, we suggest the following:

Step 1

When you are ready to take a caller, turn the CALLER level control [1] (See Figure 12a, previous page.) all the way down.

Step 2

Bring the input into the console.

Step 3

Select the phone line you want and switch the PHONE switch [5] to the ON position.

Step 4

Turn the CALLER level control [1] up to put the caller on-air.

Step 5

Set the MONITOR [2] level as low as possible. It will track the CALLER level control.

Step 6

When you are through with the caller, turn the CALLER level control [1] down and turn the PHONE switch [5] OFF.

Remote Operation

If you locate the SPH3A away from your console, you can remotely control the PHONE switch by closing pins 3 and 13 on the J6 connector [2] (Figure 9, below). In this case, we suggest the following:

Step 1

Set the CALLER level control [1] at a convenient level and leave it there.

Step 2

If the monitor output is used, turn the MONITOR control [2] as low as possible.

OR

If the monitor output is *not* used, turn the MONITOR control [2] all the way down.

Step 3

When you are ready to take a caller, punch a line on the phone and hit the remote switch for the hybrid. The caller is now on-air.

Step 4

When you are done with the caller, turn the hybrid remote switch off.

Operation
Continued ≡

REMOTING NOTE:

Many stations have used the SPH3A without remoting the PHONE switch [5] by just leaving the unit ON and switching between lines. Remember that if the SPH3A is not connected to a phone line it will not null properly. The null capability of the unit is sufficient for the unit to not oscillate when it is not connected to a phone line, but it is not recommended.

Helpful Hints

- The PHONE switch [5] (See Figure 12b, below.) in the OFF position will mute any audio going to your equipment.

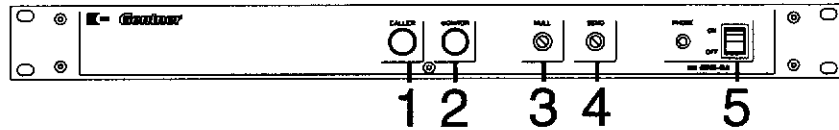


Figure 12b. Front-panel controls

- The CALLER level control [1] adjusts all caller audio, including the input to the monitor. Therefore, when you turn down the CALLER level control [1], you will also turn down the monitor level.
- Once the MONITOR control [2] has been set, it can usually be left alone.

SPH3A
Teleconferencing ≡

Two or more SPH3A units can be used in conjunction (See Figure 13, below.) to create a teleconferencing scenario. This is accomplished by cross-feeding the equalized CALLER OUT port to the J6 input port. When conferencing two units, connect pin 19 on unit 1's J6 connector to pin 12 unit 2's J6 connector, and connect pin 19 of unit 2's J6 connector to pin 12 of unit 1's J6 connector.

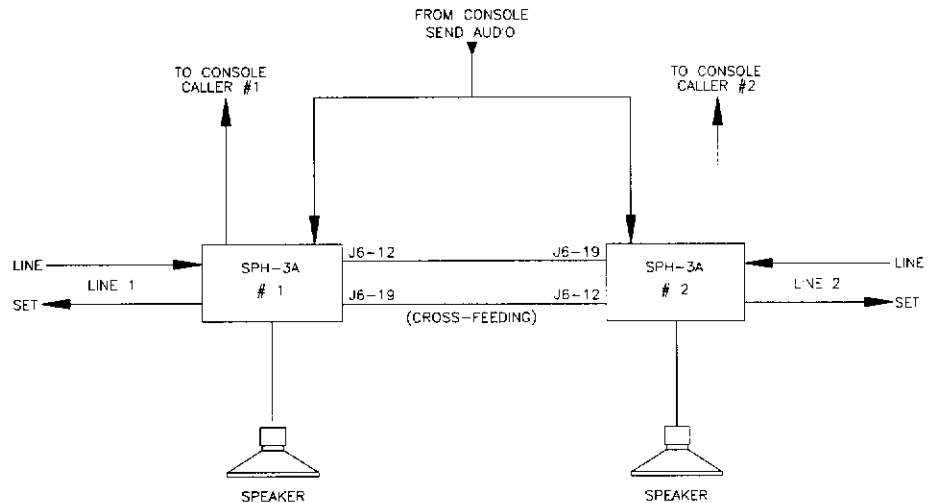


Figure 13. Two-SPH3A configuration

SPH3A TELECONFERENCING NOTE:

It is suggested that you remote the hybrids when using SPH3As in a teleconferencing scenario. This will ensure the CALLER level controls are left at the proper cross-feed level.

To adjust the cross-feeding levels, do the following:

Set the CALLER level controls [1] at 12 o'clock, then adjust the internal auxiliary level control for proper cross-feed levels. This adjustment should be as low as possible. It is best to get two callers on the two systems and obtain subjective opinions about cross-feeding levels. Adjustment of the CALLER level controls [1] from the 12 o'clock position will change the cross-feed level.

To conference more than two units, you will need to modify each unit (See Figure 14, below; Table 2, below, bottom.) to allow submixing of equalized caller outputs for every other unit. To accomplish this, remove the PC-mounted auxiliary feed level trim pot and place a jumper so pin 12 on the J6 connector feeds directly into A2. For each unit to be cross-fed, install a 10kOhm resistor in series to the mix point of A2. All level control (including the balanced send audio) will be controlled by the front-panel SEND trim pot [4].

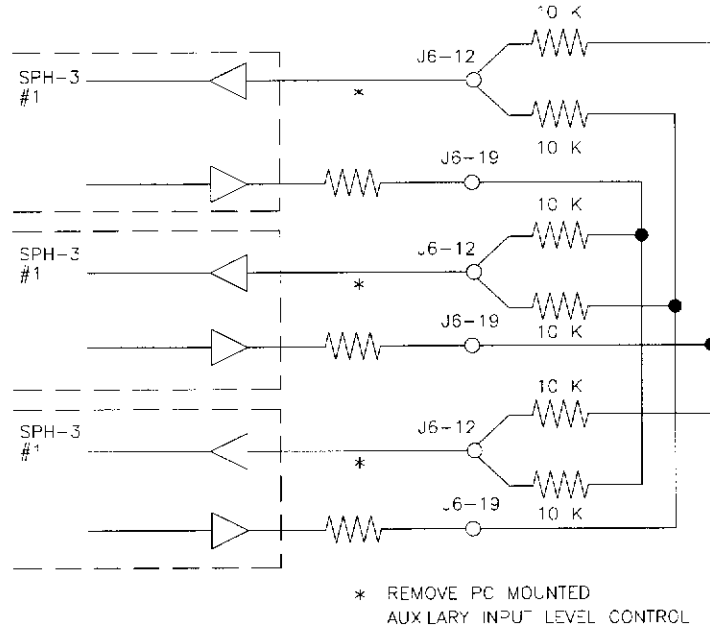


Figure 14. Multiple-SPH3A configuration

Table 2. J6 Auxiliary Input Pinout

J6 Square Connector Pins	DB25 Connector Pins
1	1
2	14
3	2
4	15
5	3
6	16
7	4
8	17
9	5
10	18
11	6
12	19
13	7
14	20
15	8
16	21
17	9
18	22
19	10
20	23
21	11
22	24
23	12
24	25
25	13

Specifications ≡

SPH3A

Dimensions

17"W x 1.75"H x 8"D

Weight

6 lbs. (dry) 8 lbs. (shipping)

Connectors

LINE:	RJ11C
SET:	RJ11C
SEND:	3-pin female XLR; -20dBm to +20dBm 600ohms, balanced
CALLER OUT:	3-pin male XLR; OdBm (+22dBm max) 600ohms, balanced
MIX OUT:	3-pin male XLR; OdBm (+22dBm max) 600ohms, balanced
MIX:	RCA phono; OdBm (+22dBm max) 600ohms, unbalanced
MONITOR:	1/4" phono (stereo); 8ohms, 2W, unbalanced
J6:	26-pin header; 10kOhm, -6dBm typical, unbalanced

Power Requirements

120Vac, 60Hz, 7W

Audio Performance

Frequency Response

330Hz to 3.3kHz within 1dB

Signal-to-Noise Ratio

>60dB

Distortion

<.1% THD

Null

-20dB (relatively line insensitive) @ 1kHz, adjustable

Operating Temperature

32-122° F/0-50° C

All specifications are subject to change without notice.

Warranty ≡

Gentner Communications Corporation (Manufacturer) warrants that this product is free of defects in both materials and workmanship. Should any part of this equipment be defective, the Manufacturer agrees, at its option, to:

- A. Repair or replace any defective part free of charge (except transportation charges) for a period of one year from the date of the original purchase, provided the owner returns the equipment to the Manufacturer at the address set forth below. No charge will be made for parts of labor during this period;
- B. Furnish replacement for any defective parts in the equipment for a period of one year from the date of original purchase. Replacement parts shall be furnished without charge, except labor and transportation.

This Warranty excludes assembled products not manufactured by the Manufacturer whether or not they are incorporated in a Manufacturer product or sold under a Manufacturer part or model number.

THIS WARRANTY IS VOID IF:

- A. The equipment has been damaged by negligence, accident, act of God, or mishandling, or has not been operated in accordance with the procedures described in the operating and technical instructions; or,
- B. The equipment has been altered or repaired by other than the Manufacturer or an authorized service representative of the Manufacturer; or,
- C. Adaptations or accessories other than those manufactured or provided by the Manufacturer have been made or attached to the equipment which, in the determination of the Manufacturer, shall have affected the performance, safety or reliability of the equipment; or,
- D. The equipments original serial number has been modified or removed.

NO OTHER WARRANTY, EXPRESS OR IMPLIED, INCLUDING WARRANTY OF MERCHANTABILITY OR FITNESS FOR ANY PARTICULAR USE, APPLIES TO THE EQUIPMENT, nor is any person or company authorized to assume any warranty for the Manufacturer or any other liability in connection with the sale of the Manufacturer's products.

Manufacturer does not assume any responsibility for consequential damages, expenses, or loss of revenue or property, inconvenience, or interruption in operation experienced by the customer due to a malfunction in the purchased equipment. No warranty service performed on any product shall extend the applicable warranty period.

In case of unsatisfactory operation, the purchaser shall promptly notify the Manufacturer at the address set forth below in writing, giving full particulars as to the defects or unsatisfactory operation. Upon receipt of such notice, the Manufacturer will give instructions respecting the shipment of the equipment, or such other matters as it elects to honor this warranty as above provided. This warranty does not cover damage to the equipment during shipping and the Manufacturer assumes no responsibility for such damage. All shipping costs shall be paid by the customer.

This warranty extends only to the original purchaser and is not assignable or transferable.

Gentner Communications Corporation, 1825 Research Way, Salt Lake City, Utah 84119

Safety
Information ≡

CAUTION: Users should not attempt to make such connections themselves, but should contact the appropriate electrical inspection authority, or electrician, as appropriate.

Appendix A: Setup Options

Stereo Board Option

Feed all inputs except the SPH3A to both the left and right channels of the console (Figure 15, below). Connect the CALLER OUT to the left channel *only* of the console, and use the board's right channel to feed the phone hybrid.

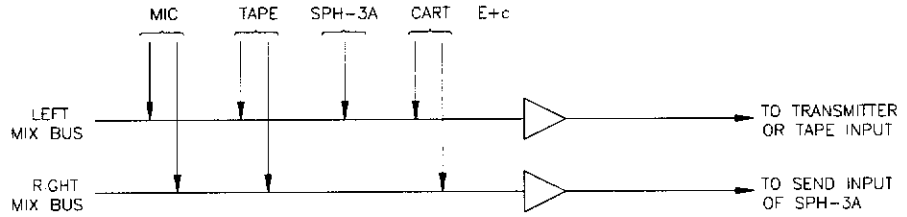


Figure 15. Stereo-board diagram

Submix Bus Option

This option requires a modification to both the SPH3A (See Figure 16, below.) 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.

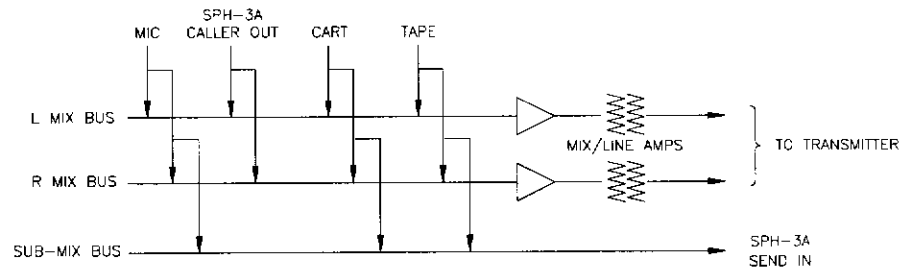


Figure 16. Submix-bus diagram

To implement this option (See Figure 17, below.), 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 pin 12 on the SPH3A's J6 connector. Now remove the PC-mounted auxiliary input trim pot, and place a jumper so pin 12 of the J6 connector feeds directly into A2. This input into A2 provides a zero impedance mix point for your submix bus. The send level can be adjusted via the front-panel SEND trim pot.

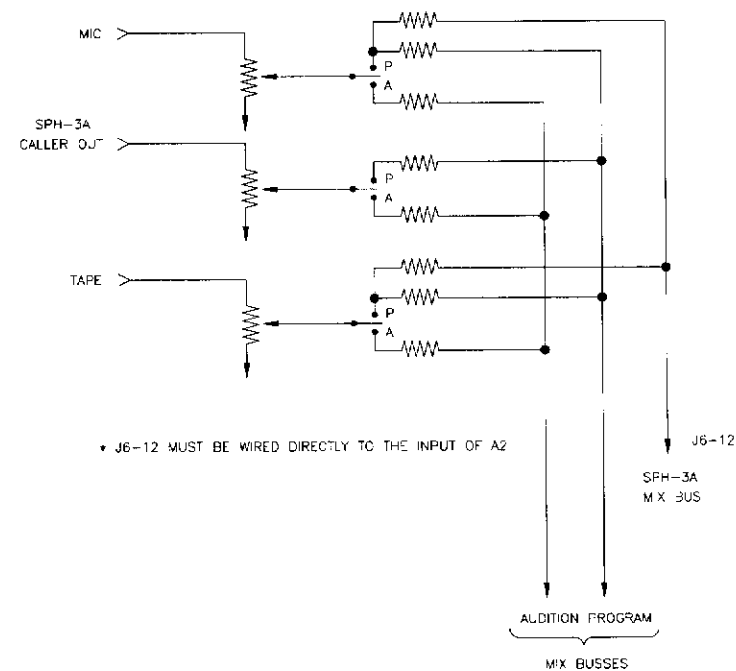


Figure 17. Submix-bus modification diagram

Downstream Mixing Option

This option mixes the caller into your console audio (See Figure 18, below.) after it has left the console. Connect the output of the console to the SEND input and connect the MIX out (RCA or XLR MIX OUT) to your tape, transmitter, etc. This will mix the caller audio with the send audio. When the PHONE 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 (.1dB maximum change from 20Hz to 30kHz), low distortion (less than .01 percent) and excellent noise figures.

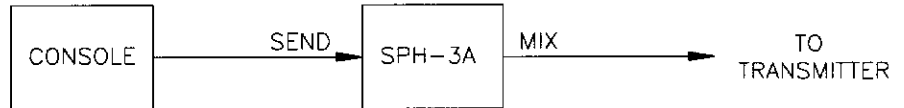


Figure 18. Downstream-mixing diagram

Mic Mixer Feed Option

This is a very simple option. Just premix all of your mix sources (See Figure 19, below.) with an external mic mixer and route the output to the SPH3A.

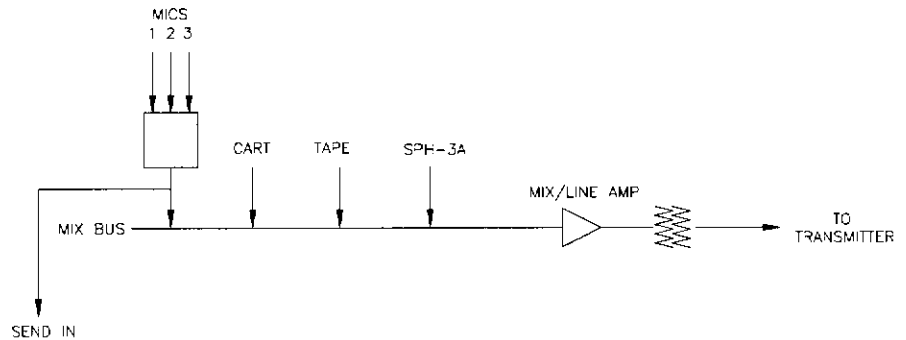


Figure 19. Mic-mixer feed diagram

Mic Input Option

This option allows you to make a mic-level input to record telephone interviews. To make it, replace R3 (See Appendix D, Page 26) with a 560kOhm resistor, then replace R5 with a 560kOhm resistor, as well. After, put a 680pf capacitor across R20, and a 680pf capacitor across R29. The two capacitors are for RFI suppression.

Appendix B: Telephone Basics 

Definition: Telephone Line.

A simple way to define a telephone line is to describe it as a two-way, two-wire communication transmission line (Figure 20, below). It gives the user the ability to send and receive information simultaneously (a great advantage over one-way systems).

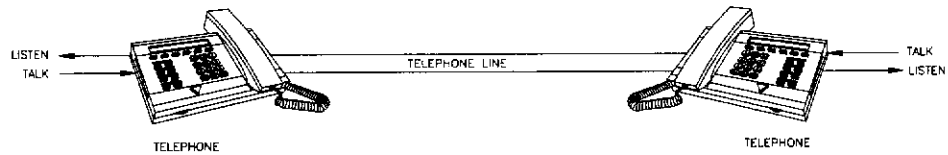


Figure 20. Typical telephone line

Telephone-Line Processing

The two wires making up the telephone line are called “tip” and “ring.” These names are carried over from the telephone’s early days, 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 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 (CO) to each user’s telephone equipment (homes, businesses, etc.). When the user’s tip and ring reaches the CO, it is usually two wire-to-four wire converted (Figure 21, below). This means that the send and receive audio is separate so the CO can process the two sources of audio independently. The information is then sent to other COs by a variety of methods including cable, microwave and satellite.

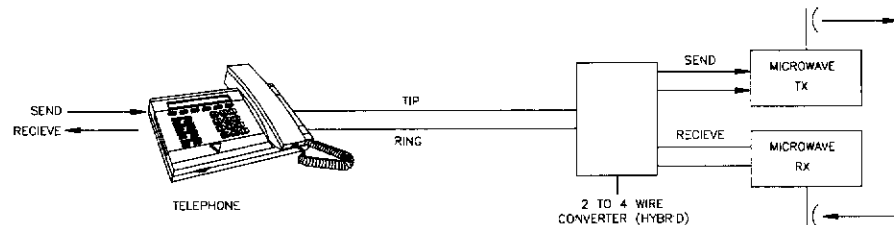


Figure 21. Two wire-to-four wire telephone line conversion

Figure 21 (above) shows how send audio is taken from the two wire-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 (See Figure 22, next page, top.) is often thought of as a single path, when in reality it is a dual system.

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 CO, a more detailed look at tip and ring will help explain the telephone’s operation.

Battery

At the CO, the telephone company (telco) places a DC voltage on tip and ring. This DC voltage is usually provided by a trickle-charged battery (the

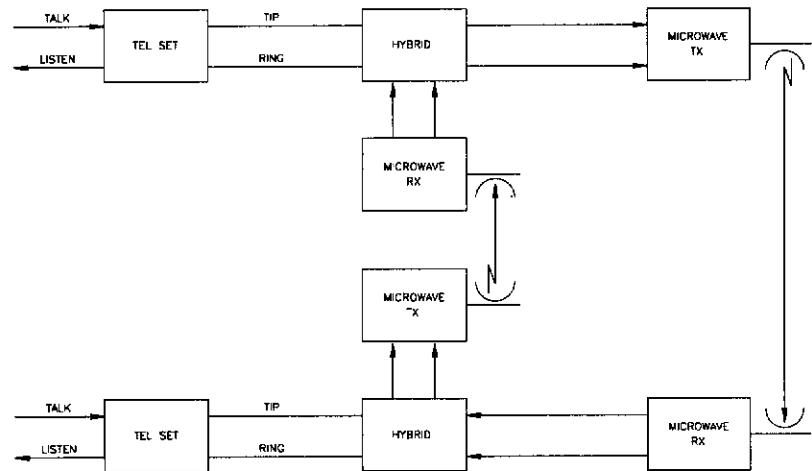


Figure 22. Telephone transmission system

battery keeps the system working during power outages). The voltage varies from 20Vdc to about 48Vdc with the tip and ring on hook (or no load). This voltage provides DC for the telephone instrument to operate.

Ringing

When someone calls, a ring voltage of 105Vac is provided to drive the ringer inside your telephone. When you pick up the handset, the hook switch disconnects the ringer and connects the telephone to the telephone line. This process is known as “terminating the line.” Anytime a DC path of 600ohms or less is provided to the tip and ring, the line will be terminated and the call will be answered.

Disconnect

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

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 RJ11C; Figure 23, below). This jack has four wires going into it; the inner two conductors are tip and ring and the outer conductors are not used.

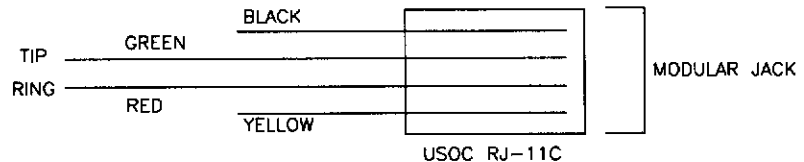


Figure 23. USOC RJ11C pinout

Inside the telephone, tip and ring go to the hook switch (Figure 24, below). When you pick up the telephone, tip and ring are routed to a hybrid that

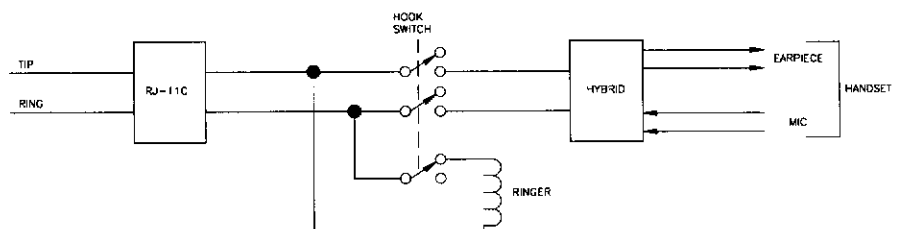


Figure 24. Single-line telephone set

Appendix B:
Continued 

provides separation for the talk and listen paths going to the handset. In addition, the ringer is disconnected.

Multiline Telephone Systems

In multiline usage, the various lines enter the customer's KSU, providing the necessary routing and control functions required by the customer. There are two types of KSU's: fat wire and slim wire.

Fat Wire

The fat-wire telephone 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 (See Figure 25, below.) is routed to the telephone instrument and the button light is turned on.

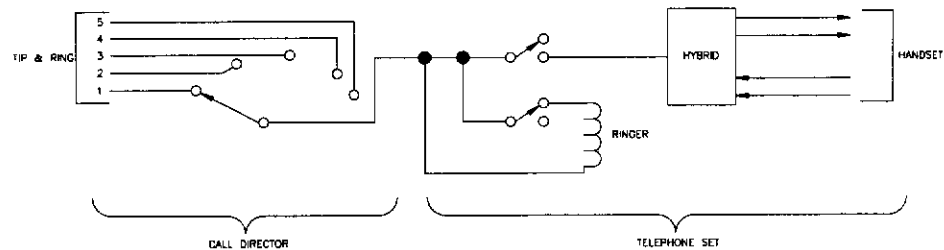


Figure 25. Multiple-line call direction

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.

Thus, the function of the KSU is to

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 SPH3A easily adapts to the fat-wire phone system by connecting the tip and ring output of the call director (See Figure 26, next page, top.) to the tip and ring input of the SPH3A. In addition, the SPH3A provides an A-line closure to turn on the lamp (refer to Installation, Page 5, for more details).

Slim Wire

A slim-wire phone system (usually referred to as digital) accomplishes the same task with four pairs of wire or less. This is done with the help of a microprocessor at the EKSU (Electronic Key Service Unit). When 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.

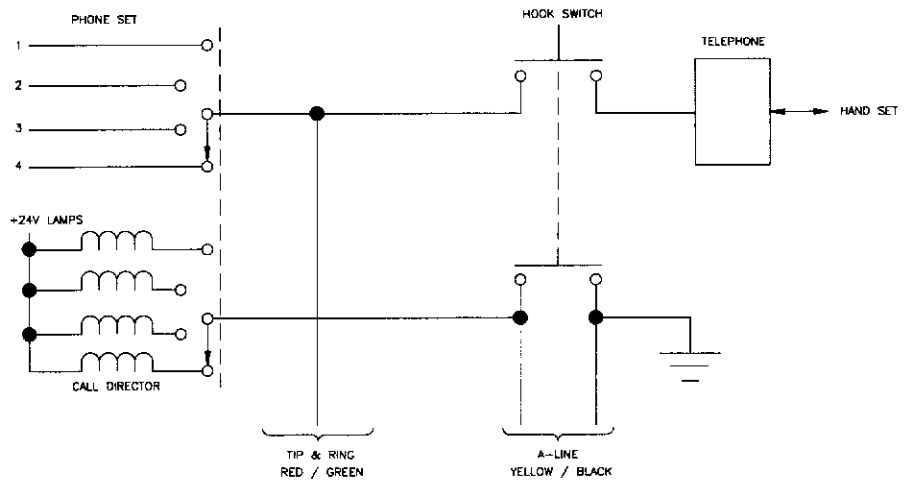


Figure 26. SPH3A phone system A-lead closure

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

The SPH3A can be installed to this type of phone system (See Figure 27, below.) by routing the tip and ring through the unit.

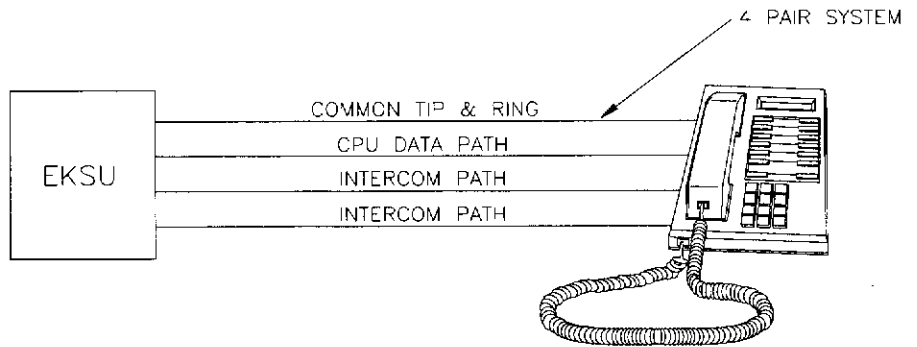


Figure 27. Slim-wire digital phone system

When the SPH3A's front-panel PHONE switch is OFF, the phone will work normally; when the switch is ON, the SPH3A will process the phone line.

Since the slim-wire system has no A-line functions (See Figure 28, next page, top.) it will be necessary to take the handset off-hook so the microprocessor will know when and what lines to process. It is, however, possible to simulate off-hook by placing a relay at the hook switch and using the A-lead closure to turn the relay on. This will eliminate the need to remove the handset when you want to use the SPH3A.

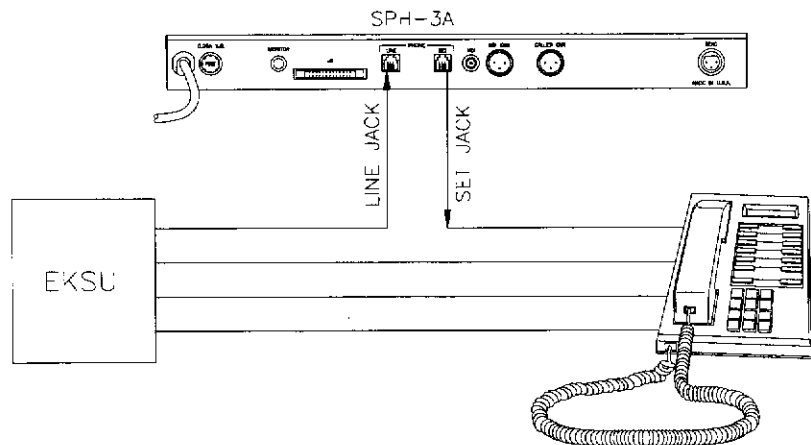


Figure 28. SPH3A installation in a slim-wire system

**Appendix C: Technical
Description** 

TECHNICAL DESCRIPTION NOTE:

Please refer to Appendix D (Page 26) for SPH3A schematics.

Balanced input audio appears at the SEND XLR. A2 provides balancing and isolation before appearing at the mix amplifier C1. Auxiliary unbalanced input audio appears at pin 12 of the J6 connector and is level controlled through a PC mount 1kOhm trim pot. This audio is mixed with the balanced send audio at C1. C1 drives the front-panel SEND trim pot, which should be adjusted for proper send level. C2 buffers the send audio before being routed to the send filtering circuits. Flee clips are provided at the output of C2 for optional bypassing the send filters.

The send filters D1, D2 and E1 provide bandpass filtering in the 330Hz to 3kHz frequency range. This filtering is provided to meet part 68 of FCC regulations. Additionally, the filtering aids in the nulling process. For high-quality send audio, the send filter may be bypassed by clipping the jumpers at the input and the output of the send filters and running a bypass jumper.

After leaving the filtering circuits, the audio is buffered through E2 and appears at the input of F2. This input has two back-to-back 7.5V Zener diodes that limit the send level at the phone line to -9dBm. The voltage divider, consisting of a 4.12kOhm 1-percent resistor and a 8.25kOhm 1-percent resistor, will set the gain structure for the maximum -9dBm send level. This maximum send level is required in part 68, FCC Rules & Regulations.

The Wheatstone bridge isolates send from receive audio. One leg of the bridge goes to the phone line. The opposite leg of the bridge is used to match the phone-line impedance. This NULL trim pot and null capacitor are adjusted for maximum isolation or null between send and receive audio.

The transformers isolate the phone line from the rest of the circuits and the four 20uF capacitors block the DC voltage on the line.

Receive audio is amplified by G1 and then bandpass filtered by G2, H2 and H1. This filtering shapes the caller audio, eliminating low-frequency hum and high-frequency multiplex carrier noise. This also aids the nulling process. This audio is then fed to the front panel CALLER trim pot.

Audio on the CALLER trim pot wiper is muted through K1 when the PHONE switch is in the OFF position.

The caller bus feeds three amplifiers: B1, J1 and J2, and I2.

B1

B1 mixes caller audio with send audio. B1 and B2 actively balance the MIX output that appears at the MIX XLR and the MIX RCA phono connector. This output level is nominally 0dBm at the XLR connector and -6dBm at the RCA connector for a 0dBm send level.

J1 and J2

J1 and J2 balance the caller audio to 0dBm. The output appears at the CALLER XLR. Unbalanced caller can be found at pin 1 of the J6 connector at -6dBm into 600ohms.

I2

I2 further equalizes the caller audio to avoid the possibility of acoustic feedback through the speaker. The output of I2 is also fed to pin 19 of the J6 connector. This output can be used to feed the auxiliary input of another SPH3A for cross-feeding purposes. Its level is -6dBm into 600ohms.

The output of I2 feeds the front-panel MONITOR control. This control then feeds K, a 2W power amp that appears at the rear-panel LINE jack.

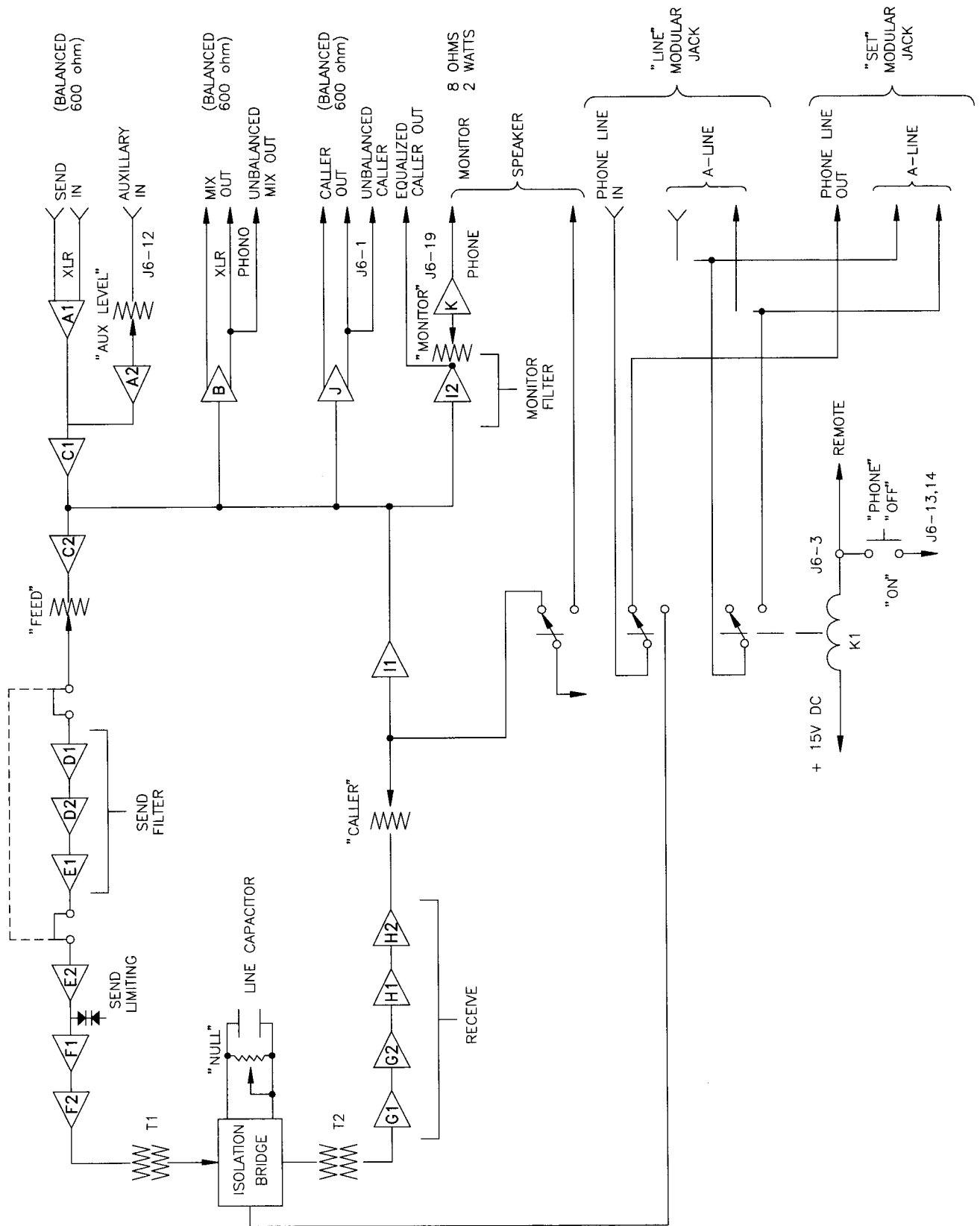
The ground for the monitor speaker is provided at this phone jack only when the front-panel PHONE switch is in the ON position. This provides further caller muting when the PHONE switch is OFF.

K1 provides unit remotability by connecting the phone line to the bridge, closing the A-line, and grounding the speaker. Remote activation of the relay occurs by applying a ground at pin 3 of the J6 connector.

A regulated power supply is provided to ensure good dependability and clean audio.

All circuits can be shorted, including the power supply, without harm.

Appendix E: Block Diagram



6. Pinouts.

<u>Pins</u>	<u>Function</u>
<u>J6:</u>	
1	Caller Out - Unbalanced
3	Remote Control (Apply Ground)
12	Auxiliary Input - Unbalanced
13,14,15,16, 17,18	Ground
19	Equalized Caller Out - Unbalanced
21	Mix Send In
23	-15VDC Regulated Power Supply
24,25,26	+15VDC Regulated Power Supply

XLR Connections:

Pin 1	Ground
Pin 2	Low
Pin 3	High

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

Modular Phone Jacks:

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