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SPL 34/44 SERIES
OPERATORS MANUAL

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19350 SW 89th Avenue

TUALATIN, OREGON 97062

A HARTZELL CORPORATION COMPANY

SPL 34/44 Series Mixing Consoles

OWNER'S MANUAL

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1. OVERVIEW

1-1. Introduction

Congratulations on your purchase of the Sunn SPL 34/44 series mixer. The 34/44 series represents a significant step forward in mixing technology. Whether used in live sound reinforcement (concerts, theaters, auditoriums), live recording or studio recording, the features and performance capabilities have been specifically designed to accommodate the needs of the most demanding professional sound engineers.

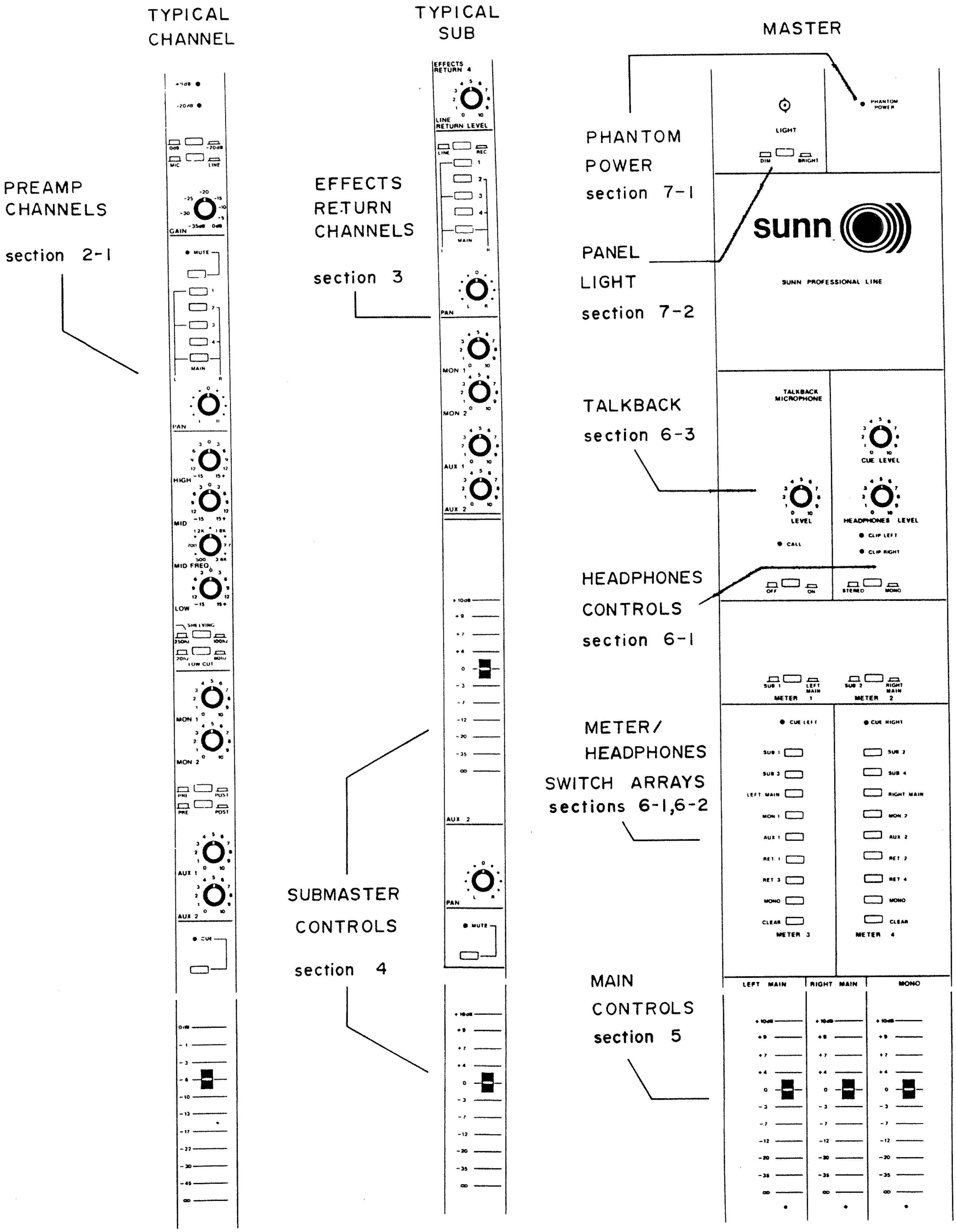
The 34/44 series has been designed with many features that insure reliability even under the duress of heavy road use. The front panel is constructed with reinforcing ribs to minimize panel flex. Each channel circuit board is braced at three points with heavy steel brackets and all back panel 3-pin connectors are secured using steel braces. In addition, flex-jumpers are used for all board inter-connects to eliminate contact breakage. Also, low profile switches have been chosen to reduce circuit board bending stresses. The 34 series mixers use high slew, high common mode rejection electronic balancing on all balanced inputs, and the 44 series balanced inputs and outputs use high quality, low distortion floated transformers. In addition, the 100mm faders used in the 44 series are constructed using double-steel guides for smooth travel and high strength. It is this type of design philosophy that gives the SPL 34/44 series the road-worthiness and quality performance that Sunn feels is required in a truly professional mixer.

The patching and bussing systems of the mixer allow it to be used for the following applications: a 4-submix, or 8-submix live sound console; an independent 6-out monitor mixer; and a 2, 4, 8, 16 or 24 (3424 & 4424) track recording console. The low distortion and high output performance, combined with the bussing and patching features make the 34/44 series flexible enough to interface with virtually any outboard equipment available.

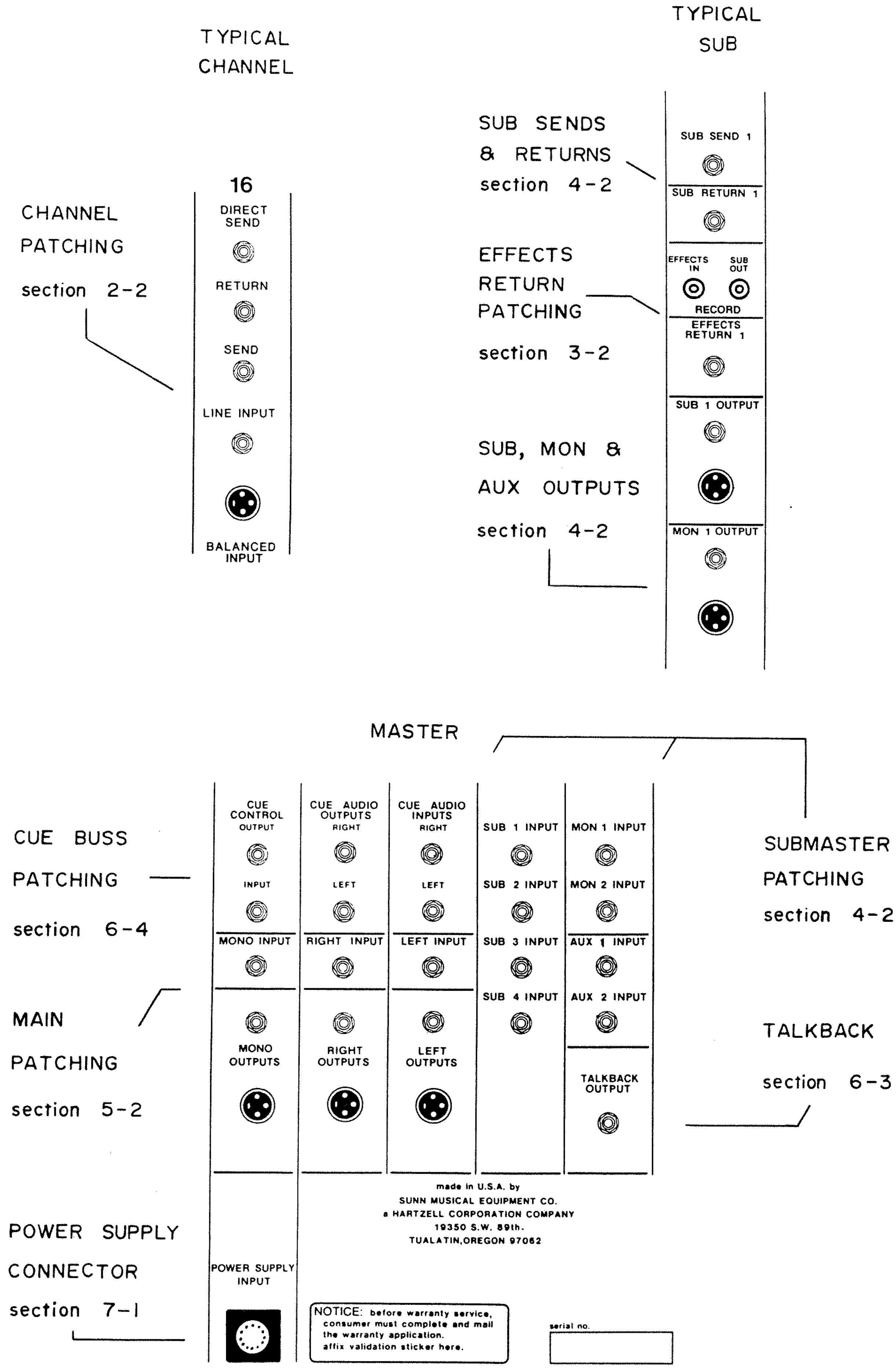
The SPL 34/44 remote power supply has been designed for fail-safe operation. Full power output is maintained even when AC line voltages and surrounding air temperatures vary radically. Turn-on and turn-off surges have been eliminated by causing all supply lines to turn on and off slowly and symmetrically in a stabilizing "power up, power down" sequence.

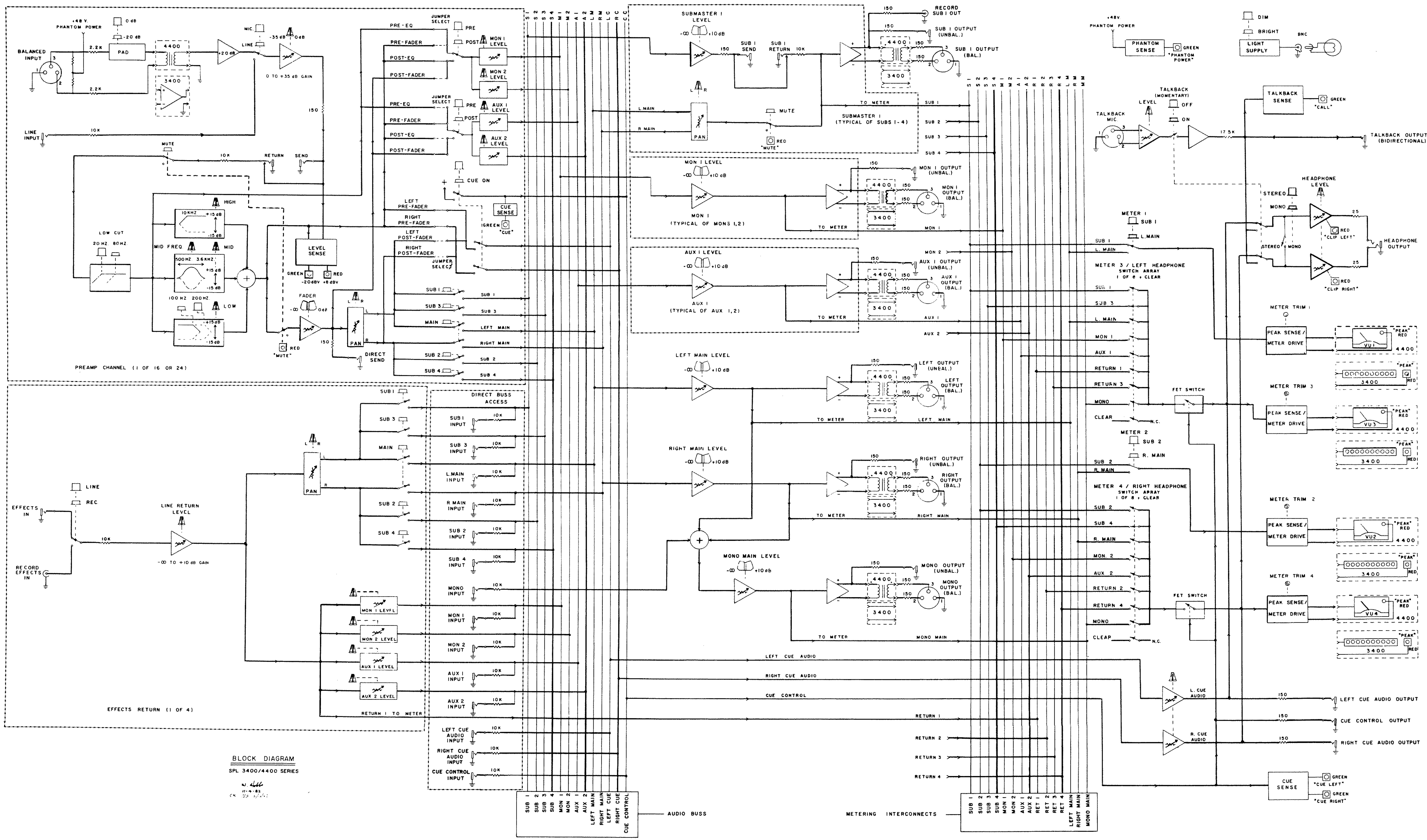
In using the 34/44 series mixer, you will realize that Sunn has designed the system with a thorough understanding of the real-life situations that are encountered by the professional sound engineer. We are proud of our 34/44 series mixer and we are certain you will be proud of it too.

SUNN SPL 34/44 SERIES Front Panel Controls



SUNN SPL 34/44 SERIES — Back Panel Layout





BLOCK DIAGRAM

SPL 3400/4400 SERIES

1-3. SYSTEM DESCRIPTION

The SPL 34/44 series of mixing consoles consists of the 24 channel SPL 3424 and SPL 4424 and the 16 channel SPL 3416 and SPL 4416. The 34/44 series is designed for live PA sound mixing, live dedicated monitor mixing and multitrack studio recording. Both the SPL 3400 and the SPL 4400 mixers have the same basic signal processing capabilities, but there are differences in the manner in which the balancing of inputs and outputs is accomplished, and in the type of faders and meters employed.

The SPL 3400 mixers utilize electronic balancing on all balanced inputs and outputs, providing better than 80dB of common mode rejection. The metering system uses high intensity bar-graph style LED's calibrated in dB and covering a range of −18dB to +9dB in 3dB steps.

The SPL 4400 mixers employ transformer-balanced floated inputs and outputs, providing up to 80dB of common mode rejection. The 100mm faders use high-quality vertically-oriented conductive plastic resistive elements for greater reliability and lower noise with two-point steel guides for smooth operation and long life. The meters of the SPL 4400 mixers are true VU "moving-pointer" meters covering a range of −20 VU to +3 VU.

Other than these differences and the slight variations in the specifications that arise from these differences, the features of the SPL 3400 and SPL 4400 mixers are identical.

Each channel consists of the following:

A. Input level indicators −20dB, +8dB
B. −20dB attenuator switch
C. Mic/Line switch
D. Input Gain control
E. Mute switch with red LED
F. Assignment switches 1, 2, 3, 4, Main
G. Pan control
H. 3-band EQ
I. 11-detent mid frequency select
J. Low EQ shelf select switch
K. Low cut select switch
L. 2 monitor controls with pre/post select
M. 2 auxiliary controls with pre/post select
N. Cue switch with LED
O. 100mm fader
P. Balanced input
Q. Unbalanced input
R. Send and Return jacks
S. Direct Send jack

The signal from each channel can be routed to four submaster busses, two monitor busses, two auxiliary busses and left, right and mono main outputs. There are four line level effects return channels which may be assigned to the eleven major outputs.

Signal monitoring is accomplished via four high-brightness bar-graph LED displays (on the SPL 3400 mixers) and four large VU meters (on the SPL 4400 mixers), with signal selection accomplished by means of two meter/headphones switch arrays.

Each submaster, monitor, auxiliary, left, right and mono main output channel provides both balanced and unbalanced outputs. All balanced outputs are capable of driving +24dBm into a 600 ohm load.

A talkback system is included for establishing communications through the mixer's bussing system, or for interfacing with a commercial intercom.

2. PREAMP CHANNELS

2-1. CHANNEL CONTROLS

2-1-1. LED LEVEL INDICATORS. These two LED's are used with the −20dB attenuator pad and the GAIN control to achieve the proper operating level in the preamp channels. The green LED will light at a signal level of −20dBV or greater. The red LED will light at +8dBV. After the red LED is activated, there is 10dB of headroom before the +18dBV clip level is reached in the channel.

2-1-2. −20dB ATTENUATOR. Engaging the −20dB ATTENUATOR pad reduces the input signal level in the BALANCED INPUT by 20dB. Use this pad when signal levels cannot be kept below +8dBV (see section 2-1-1) by means of the gain control alone. This occurs when input signal levels are too high, such as those from some keyboards, and microphones used to mic drums and bass amplifiers. NOTE: The −20dB ATTENUATOR does not affect the LINE INPUT jack.

2-1-3. MIC/LINE SWITCH. This switch is used to select between BALANCED and LINE INPUTS. Both inputs cannot be used simultaneously.

2-1-4. INPUT GAIN CONTROL. The GAIN control adjusts the channel gain over a range of 35dB. This control, when used with the two LED indicators, allows the channel to accept any signal source and achieve the maximum signal-to-noise ratio. The GAIN control, which works on the BALANCED and LINE INPUTS, should be adjusted so that when the channel is operating, the green LED is on and the red LED flashes during signal peaks. This adjustment will vary with the type of input signal. Allow more headroom for transient-related signals such as drums than for more constant-level signals like guitars.

2-1-5. MUTE SWITCH AND LED. The MUTE switch, when engaged, will light the red MUTE LED and interrupt the signal in the channel at two points: just after the channel SEND jack, and before the channel fader. Muting a channel prevents the signal from being sent to any of the SUBMASTER, MAIN, MON and AUX busses.

2-1-6. CHANNEL ASSIGNMENT SWITCHES AND PAN CONTROL. The channel assignment switches and the PAN control function together to send the signals from the channels to any of the four submaster busses and to the left and right main busses. When the PAN control is set in its center position (0), engaging any of the channel assign switches will route the signal from the channel to that particular buss. When the PAN control is rotated all the way to the left, the signal from the channel can only be assigned to submaster busses 1 and 3 and to the left main buss.

When the PAN control is rotated to the far right, the signal from the channel can only be assigned to submaster busses 2 and 4 and to the right main buss. See section 9-1 for examples of assignments.

2-1-7. CHANNEL EQUALIZATION. The channel equalization consists of four controls and a low shelving select switch.

The HIGH EQ is a high pass shelving filter with a low end roll-off at 10kHz. It has a control range of 15 dB of cut or boost.

The MID EQ is a band pass filter with 15dB of cut or boost.

The MID FREQ control allows adjustment of the center frequency for the MID EQ between 500 and 3.5 kHz in eleven detented steps.

The LOW EQ is a low pass shelving filter with selectable high end roll-off. It has a control range of 15dB cut or boost.

The LOW SHELVING switch allows the top end roll-off of the low pass filter to be set to either 250Hz or 100Hz.

2-1-8. LOW CUT SWITCH. This switch selects the low frequency cutoff in each channel at either 20Hz or 80Hz. The low cut filter is a −12dB per octave Butterworth filter.

2-1-9. MON 1 & 2 CONTROLS AND PRE/POST SWITCH. The two MON controls adjust the signal levels from the channels to the two MON busses, 1 and 2. These busses provide separate mixing for stage monitoring. The MON PRE/POST switch has been factory wired PRE-EQ when set in the PRE position and POST-EQ when set in the POST position. The position of the MON PRE/POST switch will affect the signal sent to both monitor busses.

The monitor PRE/POST switch can be rewired to be PRE-fader in the PRE position and POST-fader in the POST position by moving jumper wires on the channel circuit boards. See section 12 for further details.

2-1-10. AUX 1 & 2 CONTROLS AND PRE/POST SWITCH. The two AUX controls adjust the signal levels from the channels sent to the two AUX busses 1 & 2. The busses provide separate mixes which can be used for sends to external accessory units or monitors. The AUX PRE/POST switch has been factory wired PRE- fader when placed in the PRE position and POST-fader when placed in the POST position. The AUX PRE/POST switch will affect the signal sent to both AUX busses.

The AUX PRE/POST switch can be rewired to be PRE-EQ when placed in the PRE position and POST-EQ when placed in the POST position by moving jumper wires on the channel boards. See section 12 for further details.

2-1-11. CUE SWITCH & LED. These switches allow any of the channels to be monitored via headphones plugged into the headphone jack. To monitor a single channel or a group of channels, engage the CUE switch on the desired channel(s). The green LED will light above the switch when a channel CUE switch has been engaged. Two green LEDs will also light above the METERS 3 and 4 switch arrays to indicate that the cue buss has priority over the switch arrays.

The CUE switch is factory wired PRE-fader and can be changed to be POST-fader. See section 12 for further details.

NOTE: The cue switch, factory wired, will send a mono signal to the headphones. If rewired POST-fader, a stereo signal will be sent to the headphones. The stereo balance is set by the PAN control in the channel assignment section (see section 12-3).

2-1-12. CHANNEL LEVEL FADER. The CHANNEL FADER adjusts the level of the signal sent to the submaster and main busses. These controls are accurately marked in dB for easy reference in comparing channel settings. A recommended starting position for these controls is between −6 and −10dB. This will provide a good signal-to-noise ratio and maintain enough range for boosting the output level of a channel when required.

2-2. CHANNEL PATCHING

2-2-1. BALANCED INPUTS. These three pin audio connectors have been designed for balanced input signals from low impedance sources. The BALANCED INPUT is activated when the MIC/LINE channel switch is placed in the MIC position.

2-2-1. BALANCED INPUTS, cont.

The maximum input level that should be applied to the BALANCED input is dependent upon the position of the −20dB PAD switch, and, in the case of the 44 series, on the frequency of the input signal. With the PAD switch out, the BALANCED INPUT stage will clip signals greater than 0.9V RMS. With the PAD switch in, this clipping level is 9V RMS. Due to the possibility of transformer saturation at low frequencies on the balanced inputs of the 44 series, in order to achieve a THD of less than 0.1% at frequencies below 40 Hz, the maximum input level should be 0.5V RMS (PAD switch out) and 5V RMS (PAD switch in).

When the PHANTOM POWER switch is ON, +48 VDC will be supplied to pins 2 and 3 of the BALANCED INPUT connector. See section 11-13 for further details in mic compatability with phantom power and section 12-4 for defeating the phantom power in a channel.

2-2-2. LINE INPUTS (Unbalanced). The LINE INPUTS accept signals from high impedance microphones and line level sends. These line level signals come from sources such as keyboard mixers, synthesizers and tape recorders. The LINE INPUT is activated when the MIC/LINE switch is placed in the LINE position.

2-2-3. CHANNEL SEND AND RETURN (Unbalanced). The channel SEND and RETURN jacks provide a convenient point for inserting an accessory unit for processing that channel's signal. The SEND jack is connected after the GAIN control and before the EQ section. The SEND jack can be used without affecting the signal in the channel. Inserting a plug in the RETURN jack will interrupt the signal just after the channel SEND. Only a signal placed into the RETURN jack will be sent to the channel EQ and all the channel busses.

2-2-4. DIRECT OUT (Unbalanced). The DIRECT OUT jack delivers a line level output signal, post-channel-fader. These outputs are useful as sends to multi-track recorders or other units that require a signal that has been fully processed by the channel. The DIRECT OUT can be used without affecting the signal in the channel. The DIRECT OUT cannot be used to send a signal to an effect whose output is connected to the channel's RETURN jack, as this may cause the channel to oscillate. If the DIRECT OUT is used to send a signal to an effect, the output of the effect should be brought back to an EFFECTS RETURN channel, or to the LINE INPUT of any available preamp channel.

3. EFFECTS RETURN CHANNELS

3-1. EFFECTS RETURN CHANNEL CONTROLS

3-1-1. EFFECTS RETURN LEVEL CONTROL. This control adjusts the signal level sent to the EFFECTS RETURN channel from the EFFECTS IN jack or from the RECORD EFFECTS IN jack. The post-level-control signal level can be monitored through the headphones and meters by selecting the correct pushbuttons on the METERS 3 & 4 switch arrays.

3-1-2. LINE/RECORD SWITCH. This switch determines which input jack will deliver the signal to the EFFECTS channels. With the switch placed in the LINE position, the EFFECTS channel will receive its signal from the EFFECTS IN jack, whereas switching to the RECORD position selects the RECORD EFFECTS IN jack.

3-1-3. ASSIGNMENTS AND PAN CONTROLS. The channel assignment system for the EFFECTS RETURN channels is identical to the channel assignment system as described in section 2-1-6.

3-1-4. MON 1 & 2 CONTROLS. These two controls adjust the signal level sent from the EFFECTS RETURN channel to the two MON busses.

3-1-5. AUX 1 & 2 CONTROLS. The two AUX controls are used to adjust the signal level sent to the AUX busses from the EFFECTS RETURN channel.

3-2. EFFECTS RETURN PATCHING

3-2-1. RECORD EFFECTS INPUTS (Phono Jacks). With the LINE/RECORD switch in the RECORD position, the EFFECTS RETURN channel receives its signal from the RECORD EFFECTS IN jack. Recorded music can be played by connecting the "line out" from a 2- or 4-track tape recorder to the RECORD inputs. These inputs are also useful for mixing down a 4-track recording to stereo or mono.

3-2-2. EFFECTS RETURN INPUTS (Unbalanced). With the EFFECTS RETURN channel LINE/RECORD switch in the LINE position the EFFECTS RETURN INPUTS send signals to the EFFECTS RETURN channels. These inputs can function as return inputs from external effects units, inputs from other mixers, or as line inputs from musical instrument amplifiers.

4. SUBMASTER SECTION

4-1. SUBMASTER CONTROLS

4-1-1. MON 1 & 2 SEND LEVEL FADERS. The MON SEND LEVEL FADERS, 1 & 2, adjust the output signal level sent from the MON busses to the MON OUTPUTS on the back panel.

4-1-2. AUX 1 & 2 SEND LEVEL FADERS. The AUX SEND LEVEL FADERS adjust the signal levels from the AUX busses to the AUX OUTPUT jacks.

NOTE: All balanced inputs and outputs are 3-pin audio connectors. On the 34 series they are electronically-balanced, while on the 44 series they are transformer-balanced and floated. All unbalanced inputs and outputs are ¼″ phone connectors, unless otherwise noted.

4-1-3. SUBMASTER PAN CONTROLS. These controls pan the submaster signals between the left and right MAIN busses.

4-1-4. SUBMASTER MUTE SWITCHES AND LEDS. Engaging the SUBMASTER MUTE switch prevents the submaster signal from being sent to the MAIN busses, without affecting the signals sent to the SUB OUTPUT jacks on the back panel. The SUBMASTER MUTE LED lights when the MUTE switch is engaged.

4-1-5. SUBMASTER SEND LEVEL FADERS. These faders adjust the signal level sent from the submaster busses to the SUB OUTPUT jacks, SUB SEND jacks, RECORD SUB OUT jacks, and the LEFT and RIGHT MAIN busses.

4-2. SUBMASTER PATCHING

4-2-1. MON & AUX DIRECT BUSS INPUTS (Unbalanced). These jacks provide pre-fader line level input to the MON and AUX busses. They are useful when patching two mixers together. (See section 9-3).

4-2-2. SUB 1, 2, 3, & 4 DIRECT INPUTS (Unbalanced). These jacks accept line level input signals and deliver them to the submaster busses prior to the submaster faders. These jacks are also used when patching two boards together (See section 9-3).

4-2-3. SUB SENDS AND RETURNS (Unbalanced) The SUB SEND and RETURN jacks provide a convenient point for inserting an accessory unit for processing the submaster signal.

The SUB SEND jacks are connected after the sub fader, prior to the SUB OUT, PAN and MUTE. Inserting a plug in the SUB RETURN jacks will interrupt the signal just after the SUB SEND. Only a signal placed into the SUB RETURN jack will continue in the SUB channel. The return signal will be sent to the SUB RECORD OUT jack, SUB OUT jacks (balanced and unbalanced) and MAIN busses.

4-2-4. SUB OUTPUTS (Balanced & Unbalanced). These jacks are the line level outputs of the submaster channels, and can be used for 4-out live mixing, 4-track recording, or for patching two boards together (section 9-3). The SUB OUTS will not be affected by the submaster MUTE switches. The balanced and unbalanced outputs can be used simultaneously.

4-2-5. SUB RECORD OUTPUTS (Phono Jacks). An unbalanced line level output from the submaster channels is available from these jacks. This output is compatible with most types of tape recorders.

4-2-6. MON & AUX OUTPUTS. (Balanced and Unbalanced) These jacks deliver a line level output from the two MON busses and the two AUX busses. Stage monitor systems and external signal processing units can be driven by these outputs. These jacks are also used when patching two boards together (see section 9-3). If desired, both the balanced and unbalanced output jacks can be used simultaneously.

5. MAIN SECTION

5-1. MAIN CONTROLS

5-1-1. LEFT & RIGHT MAIN FADERS. The LEFT & RIGHT MAIN faders adjust the signal level sent to the LEFT and RIGHT OUTPUTS. Their settings also affect the amount of signal sent to the mono main.

5-1-2. MONO MAIN FADER. The signal level at the MONO OUTPUT jack is controlled by the MONO MAIN fader. The mono signal is the sum of the right and left main busses plus any input signal present at the MONO INPUT.

5-2. MAIN PATCHING

5-2-1. LEFT & RIGHT OUTPUTS (Balanced & Unbalanced). These output jacks provide a line level output from the left and right main busses. In a stereo system, the LEFT and RIGHT OUTPUTS are used to drive the PA system. They are also used when patching two boards together (section 9-3). The balanced and unbalanced output jacks can be used simultaneously if desired. This allows a stereo recording to be made from the unbalanced outputs while the balanced outputs are being used to drive the PA.

5-2-2. LEFT & RIGHT INPUTS (Unbalanced). These input jacks allow a signal to be patched into the left and right main busses prior to the LEFT & RIGHT MAIN faders. They are also used when patching boards together (section 9-3).

5-2-3. MONO OUTPUTS (Balanced & Unbalanced). The MONO OUTPUTS deliver a line level output from the mono main buss. The balanced and unbalanced jacks may be used simultaneously. In a mono PA system, the MONO OUTPUTS are used to drive the PA.

5-2-4. MONO INPUT (Unbalanced). The MONO INPUT allows a signal to be added to the mono main buss, prior to the MONO main fader.

6. HEADPHONES/METERS/TALKBACK/CUEING

6-1. HEADPHONES CONTROL

6-1-1. CUE LEVEL. The CUE LEVEL control adjusts the level of the cue buss signal sent to the headphones and LEFT and RIGHT CUE AUDIO OUTPUTS.

6-1-2. HEADPHONES LEVEL. This control adjusts the overall signal level sent to the headphones. The input to the headphones is selected by engaging the desired switches in the headphones/meter select arrays.

NOTE: All balanced inputs and outputs are 3-pin audio connectors. On the 34 series they are electronically-balanced, while on the 44 series they are transformer-balanced and floated. All unbalanced inputs and outputs are ¼″ phone connectors, unless otherwise noted.

6-1-3. HEADPHONES LEFT & RIGHT CLIP LEDS. The red LEFT & RIGHT CLIP LEDS indicate that the headphones amp is producing a distorted signal. This makes it possible to distinguish between distortion present in the original signal, and distortion produced by the headphones amplifier. If the clip LEDS light, turn down the HEADPHONES LEVEL control.

6-1-4. STEREO/MONO SWITCH. The headphones may be operated in either mono or stereo. With the switch in the MONO position, the same signal appears in both sides of the headphones. In the STEREO position, the signal selected on the METER 3 switch array is assigned to the left headphones speaker, and the signal selected on the METER 4 switch array is assigned to the right headphones speaker. As wired from the factory, a cue signal is in mono. The cue signal can be re-wired in stereo (see section 12-3).

6-1-5. HEADPHONES OUTPUT. The headphones output jack is located at the right front edge of the console, below the arm rest. This stereo output can be used with as many pairs of stereo headphones as needed. However, as more headphones are added, the maximum level of each will be reduced, depending on the impedance of the headphones.

6-1-6. HEADPHONES/METER ASSIGNMENTS. The assignment of signals to the headphones and METERS 3 and 4 is described in section 6-2-2.

6-2. METERS. One of the most noticeable differences between the 34 series and the 44 series mixers is in the type of meters (displays) they use. The 34 series uses a bar-graph style LED (light emitting diode) display calibrated in decibels, while the 44 series uses a "moving -pointer" VU meter. The manner in which signals are assigned to the meters (displays), however, is identical in both series. In the following sections references will be made to METER 1, METER 2, and so on, but all comments apply equally to the LED displays as well. Where differences do exist, they will be specifically noted.

6-2-1. METERS 1 & 2 ASSIGNMENTS. These two switches are used to select which output signals will be displayed on METERS 1 & 2. METER 1 switch allows either the SUB 1 or the LEFT MAIN signal to be displayed on METER 1. METER 2 switch selects either the SUB 2 or the RIGHT MAIN signal to be displayed on METER 2.

6-2-2. METERS 3 & 4 AND HEADPHONES ASSIGNMENTS. The METER 3 and METER 4 switch arrays allow complete monitoring of all critical output busses and effects return busses. Under normal conditions, engaging one or more of the switches of these arrays causes the signal(s) from the corresponding buss(es) to be routed to METERS 3 and 4 and the headphones amp. This allows the signal(s) to be monitored both visually and audibly. Depressing any switch in an array disengages the other switches in that array, removing unwanted signals from the headphones/meter system. Depressing the CLEAR button removes all signals from that switch array from the headphones/meter system.

METERS 3 and 4 are also dedicated to the cue buss, which takes priority over the switch array assignments—whenever a preamp channel CUE SWITCH is engaged, these meters and the headphones amp monitor the cue signal.

The talkback system has the highest priority in the headphones amplifier. Engaging the talkback ON/OFF switch overrides all other signals (cue and switch arrays) and connects the talkback signal to the headphones amp. The meters are not affected by the talkback ON/OFF switch.

6-2-3. METER ADJUSTMENT. Each VU METER in the SPL 44 series has a "zero adjustment" screw located just under the meters. This screw is used to align the VU METER needle to the meter scale. With no signal present, adjust the screw so that the meter needle points exactly at the left-most mark (0%) of the meter scale. Since there is no needle on the LED display of the 34 series, there is no equivalent adjustment for those mixers.

The signal level required to produce a meter reading of 0dB on the 34 series and 0 VU on the 44 series may also be changed (see section 12-5).

6-3. TALKBACK SYSTEM

6-3-1. TALKBACK MICROPHONE INPUT (Balanced). This actively-balanced low impedance input is located on the front panel of the mixer. Plug the talkback microphone into this connector. Phantom power is supplied to the mic when the PHANTOM POWER supply switch is engaged. See section 12-4 for details on defeating the phantom power supply for this input and section 11-13 for phantom power and mic compatability.

6-3-2. TALKBACK LEVEL CONTROL. This control adjusts the talkback signal level sent to the TALKBACK OUTPUT jack.

6-3-3. CALL INDICATOR LED. With the talkback ON/OFF switch engaged, speaking into the talkback mic will light this green LED, as will any audio signal entering the TALKBACK OUTPUT jack from an external source. NOTE: Some commercial intercom systems use a DC level-shift on the audio line as a "call" signal. Such a level shift will also cause the LED to come on.

6-3-4. TALKBACK ON/OFF SWITCH. Depressing this momentary switch accomplishes two tasks: it connects the talkback microphone amp to the TALKBACK OUTPUT jack, and it switches the input of the headphones stereo amp from the CUE/HEADPHONES buss to the talkback system. Two-way communication can then take place.

6-3-5. TALKBACK OUTPUT (Unbalanced). The TALKBACK OUTPUT is used to patch the talkback system to any

NOTE: All balanced inputs and outputs are 3-pin audio connectors. On the 34 series they are electronically-balanced, while on the 44 series they are transformer-balanced and floated. All unbalanced inputs and outputs are ¼″ phone connectors, unless otherwise noted.

6-3-5. TALKBACK OUTPUT (Unbalanced), cont.

input on the mixer, to another mixing board, or to an external intercom system. This jack has an input/output impedance of 17.5K ohms which makes it compatible with most intercom systems. The TALKBACK OUTPUT is a bi-directional connection.

6-4. CUE BUSS PATCHING

6-4-1. LEFT & RIGHT CUE AUDIO OUTPUTS (Unbalanced). These two jacks deliver line level output from the cue/headphones buss. The output signals at these jacks are the same signals sent to the headphones stereo amp in the mixer, and thus may be used to drive additional headphones amplifiers. These jacks are also used when patching two SPL mixers together (section 9-3).

6-4-2. LEFT AND RIGHT AUDIO INPUTS (Unbalanced). These jacks provide input to the cue/headphones buss and are used when patching two SPL mixers together (section 9-3).

6-4-3. CUE CONTROL OUTPUT (Unbalanced). This jack carries the cue priority logic signal for the CUE buss override of the headphones/meter circuits and is used when patching two Sunn SPL mixers together (section 9-3).

6-4-4. CUE CONTROL INPUT (Unbalanced). This input is designed to accept the CUE CONTROL OUTPUT from another Sunn SPL Mixer when patching the two mixers together (section 9-3).

7. REMOTE POWER SUPPLY AND PANEL LIGHT

7-1. POWER SUPPLY

7-1-1. CONNECTOR CORD AND CONNECTOR. This cable and the female 9 pin connector are used to patch the supply voltages from the remote power supply to the mixing console. The female connector is keyed and will only connect one way with the male connector on the back of the mixing console. Push the connectors together and twist the locking ring clockwise to secure the connection.

7-1-2. ON/OFF SWITCH. This is the power switch for the mixing console. When the switch is turned on, a light located in the switch will come on. Make sure the cable is connected to the mixing console before turning the power on.

7-1-3. USER ACCESSIBLE FUSE. The only user accessible fuse is located on the back panel of the power supply. To replace this fuse, insert a small screwdriver into the slot located at the top of the fuse holder and push down gently. The fuse holder cap should release and pop partially out of the fuse holder. Remove the fuse holder cap and the fuse should come out with the cap. Replace the fuse with a fuse of the same type and rating specified on the back panel [DOMESTIC (U.S., Canada), 2 amp/125 volt slow blow; most EXPORT, 1 amp/250 volt slow blow]. Make sure the power switch is off when replacing the fuse.

7-1-4. PHANTOM SUPPLY ON/OFF SWITCH AND LED. The PHANTOM POWER supply switch is located on the front panel of the power supply. Engaging this switch will send +48VDC to all microphone input jacks on the mixing console. A green LED located on the mixing console will light when the phantom power is on. It is normal for the phantom power and its indicating LED to turn on very slowly. NOTE: To prevent system pops and possible damage to preamp channel inputs, it is best to avoid plugging and unplugging microphones when the phantom power is on.

7-1-5. MOUNTING POWER SUPPLY. The power supply for the SPL mixing console has been designed for 19″ rack mounting in a 3.5″ rack space. When rack mounting this unit, allow adequate space for ventilation. It is not advisable to mount the supply next to a power amplifier in a rack.

7-1-6. EXPORT MODIFICATIONS. See qualified service personnel to modify domestic units (120V 60Hz) to meet export requirements (240V 50/60Hz).

7-1-7. POWER UP/POWER DOWN SEQUENCE. Before powering up the SPL mixing console, make sure all connections are made in the PA system (power amp, EQ's, electronic crossovers, microphones, external effect units, etc.). The power supply for the SPL mixing console has been designed for a slow turn-on time, thus preventing turn-on pops. It is advisable to turn on the power amps in the PA system last, or at least keep the power amp level controls at their off position when powering up the other parts of the PA system.

7-1-8. AC GROUND MODIFICATION. The chassis of the remote power supply is AC ground-referenced via the ground lead of the AC cord. However, the ground of the mixer itself is floated, which means that there is no connection between the mixer ground system and the AC ground. Connecting the two grounds is quite straightforward, requiring the use of a Phillips screwdriver and the Sunn 34/44 series ground modification kit #32-1001-0.

To install this kit, remove the top of the remote power supply. Orient the power supply so that the phantom power switch is toward you and you are looking down into the chassis. You will see a long aluminum heat sink with several power transistors mounted on it. It is secured to the chassis by means of four 8-32 screws. Remove the screw attaching the left end of the heat sink to the chassis (the one closest to the thick multi-wire power cable that connects the supply to the mixer) and replace this screw with the screw supplied in the kit. The kit also contains a short wire with a spade lug on one end and an Amp connector (barrel-type connector) on the other end. Place the spade lug on the screw and secure it with the nut provided. On the left end of the power supply circuit board you

NOTE: All balanced inputs and outputs are 3-pin audio connectors. On the 34 series they are electronically-balanced, while on the 44 series they are transformer-balanced and floated. All unbalanced inputs and outputs are ¼″ phone connectors, unless otherwise noted.

7-1-8. AC GROUND MODIFICATION, cont.
should see a cluster of wires connected to small vertical posts (Amp posts). Connect the Amp connector on the free end of the wire you just installed to the one unused Amp post on the circuit board. Replace the cover of the power supply, and the modification is complete.

7-2. PANEL LIGHT

7-2-1. GOOSENECK EXTENSION. The flexible gooseneck extension allows the light to be adjusted to any position within an 18″ radius. The panel light is located at one end of the extension and a BNC female connector is located at the opposite end. The panel light gooseneck is connected to the front panel light connector.

7-2-2. LIGHT CONNECTOR. The male BNC light connector is located in the upper right corner of the mixing console. The male connector is keyed so that the female connector on the gooseneck extension will only fit one way. Line up the two connectors, then push the female connector down over the male connector and turn clockwise to lock the gooseneck extension to the mixing console. Turn the female connector counter-clockwise to release the gooseneck extension.

7-2-3. BRIGHT/DIM SWITCH. This switch is used to change the intensity of the panel light.

7-2-4. LIGHT STORAGE CLIPS. When not in use, the gooseneck light extension mounts to the two plastic clips located on the back panel of the mixing console.

8. SPL 34/44 SERIES MIXER SPECIFICATIONS

Unless otherwise noted, specifications apply to both 34 & 44 series.

FREQUENCY RESPONSE +0dB, −3dB

	SPL 3400	SPL 4400
Bal Inputs to Main Outputs:	20Hz to 30kHz	20Hz to 40kHz
Line Inputs to Main Outputs:	20Hz to 40kHz	20Hz to 50kHz
Effect Return Inputs to Main Outputs:	20Hz to 40kHz	20Hz to 50kHz

DISTORTION

CONDITIONS: Gain @ −20dB, EQ flat, fader @ max, 7V RMS output into 600 ohms, Main Output

Total Harmonic Distortion (THD)

	SPL 3400	SPL 4400
Bal Input:	0.03% max, 20Hz to 20kHz	0.05% max, 50Hz to 20kHz 0.1% max, 30Hz
Line Input:	0.03% max, 20Hz to 20kHz	0.03% max, 20Hz to 20kHz

Intermoldulation Distortion (IMD)

Bal Input: 0.04% max. SMPTE

HUM AND NOISE

Signal to Noise Ratio (S/N) A-weighted, referenced to 10V output into 600 ohms.

Residual (all faders at min):	120dB
Main (Main fader at 0dB, other faders at min):	95dB
Main (Main fader at 0dB, one pre-amp at −20dB gain, EQ flat, fader max):	92dB
Equivalent Input Noise (150 ohm source):	−127dB

MAXIMUM GAINS (EQ FLAT)

Bal Input to Direct Out:	55dB
Bal Input to Submaster Out:	65dB
Bal Input to Main Out:	75dB
Line Input to Direct Out:	20dB
Effects Return Input to Main Out:	30dB

INPUT IMPEDANCE

	SPL 3400	SPL 4400
Bal Inputs:	1.3K ohms	2.2K ohms
All Other ¼″ Inputs:	10K ohms	10K ohms

OUTPUT IMPEDANCE

All Bal Outputs: 300 ohms
All other ¼″ Outputs: 150 ohms

SLEW RATE

Greater than 6V/usec

MAXIMUM OUTPUT LEVELS

Pre-amp Send and Direct Out: 6V RMS into 600 ohms
8V RMS into 10K ohms

8. SPL 34/44 SERIES MIXER SPECIFICATIONS, cont.

All other unbalanced outputs: 8V RMS into 600 ohms
10V RMS into 10K ohms
All balanced outputs: 12.3V RMS into 600 ohms [+24dBm]

CROSSTALK [AT 1kHz]

Between adjacent Submasters or Monitors:	−65dB
Between adjacent Mains:	−75dB
Between adjacent Pre-amps:	−65dB

COMMON MODE REJECTION RATIO [CMRR]

	SPL 3400	SPL 4400
Bal Inputs:	80dB min, 20Hz to 20kHz	80dB min at 1kHz 50dB min at 20kHz

EQUALIZATION

Range: 15dB cut and boost, low, mid and high
Low: 100Hz or 250 Hz shelving
Mid: 11 detent, selectable frequencies, 500 Hz to 3.6kHz bandpass filter
High: 10kHz shelving

HI-PASS FILTER

Selectable 20Hz or 80Hz 12dB/octave, Butterworth

TALKBACK SYSTEM

Input-Output Impedance: 17.5K ohms
Output Level: 2V RMS max into 200 ohms
Gain: Talkback Mic Input to Talkback Output: 60dB max

HEADPHONES OUTPUT

Output Impedance: 25 ohms
Power Output into 8 ohms: 0.7W/Channel
Clip Indicators Threshold: 6dB below actual clipping

PRE-AMP LEVEL INDICATORS

Red: +8dBV
Green: −20dBV
Sensing: Pre and Post EQ

METER RECALIBRATION LIMITS

3400 factory set at 0dB = 1V RMS, 600 ohms
4400 factory set at 0 VU = 1V RMS, 600 ohms
Meter can be adjusted to read 0 VU [0dB] at any output level from .25V RMS to greater than 10V RMS.

DIMENSIONS:	SPL 3416, 4416	SPL 3424, 4424
	[H x W x D]	[H x W x D]
	11.6 x 37.3 x 36.3″	11.6 x 48.5 x 36.3″
	29.5 x 94.7 x 92.2 cm.	29.5 x 123.2 x 92.2 cm.

WEIGHT	SPL 3416	SPL 4416	SPL 3424	SPL 4424
	53 lbs.	71 lbs.	70 lbs.	90 lbs.
	24.09 kg.	32.26 kg.	31.8 kg.	40.9 kg.

SPL 3400, 4400 POWER SUPPLY SPECIFICATIONS

AC POWER REQUIREMENTS AND REGULATION:

1. 120VAC [nominal] 50/60Hz 2.5A max.
   Regulation: 90 to 132 VAC
2. 240VAC [nominal] 50/60Hz 1.25A max.
   Regulation: 180 to 264VAC
3. 100VAC [nominal] 50/60Hz 3.0A max.
   Regulation: 75 to 110VAC

TEMPERATURE STABILITY:

Stable under full power to 120°F [50°C] ambient.

PHANTOM SUPPLY DELAYED TURN ON:

Slows turn on time to eliminate system pops:
+9VDC @ 7 sec.
+24VDC @ 16 sec.
+45VDC @ 49 sec.

POWER SUPPLY SPECIFICATIONS, cont.

DIMENSIONS:	3.5 x 19 x 5.1″
[H x W x D]	7.9 x 48.3 x 13 cm.
WEIGHT:	16 lbs.
	7.27 kg.

9. APPLICATIONS

9-1. LIVE SOUND MIXING. The following examples assume you will be mixing sound for a band. The principles illustrated, however, apply equally to sound reinforcement for theater, church, etc.

9-1-1. CHANNEL ASSIGNMENTS. The channel assignment system is explained in section 2-1-6. To describe a typical setup, we'll assume the following "normal" situation and describe one way to use the assignments. The actual assigns will vary with the type of band, and the soundman's preferences.

# OF SOURCES	SOURCE	CHANNELS
(4)	Vocal Mics	1, 2, 3, 4
(1)	Guitar Mic	5
(1)	Sax Mic	6
(3)	Keyboard Sends	7, 8, 9
(1)	Bass Send	10
(4)	Drum Mics	11, 12, 13, 14
(1)	Announce Mic	15
(1)	Effects Return	16
(2)	Left and Right Tape	17, 18

9-1-2. MONO MIX ASSIGMENTS. The following describes a way to set up our hypothetical band for a mono mix.

SOURCE	CHANNELS	CH. ASSIGN*	CH. PAN
Vocals	1, 2, 3, 4	(1) 2 3 4 M	CTR
Guitar	5	1 (2) 3 4 M	CTR
Sax	6	1 (2) 3 4 M	CTR
Keyboards	7, 8, 9	1 2 (3) 4 M	CTR
Bass	10	1 2 3 (4) M	CTR
Drums	11, 12, 13, 14	1 2 3 (4) M	CTR
Announce	15	1 2 3 4 (M)	CTR
Effects Return	16	1 2 3 4 (M)	CTR
Tape Deck	17, 18	1 2 3 4 (M)	CTR

*The figures in parentheses indicate which assignment buttons are depressed.

SUBMASTERS:	SUB 1 (vocals)	Pan L
	SUB 2 (guitar, sax)	Pan L
	SUB 3 (keyboards)	Pan R
	SUB 4 (bass & drums)	Pan R
MAINS:	LEFT: Vocals, guitar, sax	
	RIGHT: Keyboards, bass, drums	
	MONO: Full mix plus effects, announce and tape	

The MONO OUTPUT will typically be connected to a single equalizer, then sent to an electronic crossover, which will drive the main system amplifiers and speakers.

9-1-3. STEREO MIX ASSIGNMENTS. The following is a description of a stereo setup for our hypothetical band.

SOURCE	CHANNELS	CH. ASSIGN*	CH. PAN
Vocals	1, 2, 3, 4	(1) (2) 3 4 M	L ◄ - ► R**
Guitar	5	(1) (2) 3 4 M	L ◄ - ► R**
Sax	6	(1) (2) 3 4 M	L ◄ - ► R**
Keyboards	7, 8, 9	1 2 (3) (4) M	L ◄ - ► R**
Bass	10	1 2 (3) (4) M	L ◄ - ► R**
Drums	11, 12, 13, 14	1 2 (3) (4) M	L ◄ - ► R**
Announce	15	1 2 3 4 (M)	CTR
Effects Return	16	1 2 3 4 (M)	CTR
Tape Left	17	1 2 3 4 (M)	L
Tape Right	18	1 2 3 4 (M)	R

*The figures in parentheses indicate which assignment buttons are depressed.

**Depends on left/right placement desired.

9-1-3. STEREO MIX ASSIGNMENTS, cont.

SUBMASTERS:	SUB 1 Left vocals, guitar & sax	Pan Left
	SUB 2 Right vocals, guitar & sax	Pan Right
	SUB 3 Left keyboards, bass & drums	Pan Left
	SUB 4 Right keyboards, bass & drums	Pan Right
MAINS	LEFT: All left mixes	
	RIGHT: All right mixes	
	MONO: Not used	

There are many different ways to use the assignment system, depending on the desired effect. In this example we used an open channel for our effect return, which allowed us to use the channel equalization with the effect. An EFFECTS RETURN channel could also be used if no equalization is desired. Another option on the mono mix is to keep the submaster pans centered, and use the left & right master outputs so that left & right speaker systems can have separate level controls and equalizers.

9-1-4. 8 SUBMIX ASSIGNMENTS. To create 8 separate submixes on the 34/44 series, assign the channels to the SUBS 1, 2, 3, 4, MON 1 & 2 and AUX 1 & 2. To mix MON & AUX outputs to the LEFT and RIGHT MAINS, patch the 4 AUX & MON outputs to the EFFECTS RETURN inputs 1-4 using ¼″ shielded cables. The MAIN switches on the EFFECTS RETURN channels should be engaged, and the PAN control adjusted for the desired left & right assignment. This allows the 34/44 series to be used as a 24 (or 16) x 8 x 2 x 1 mixer.

Be certain that the EFFECTS RETURN AUX & MON level controls are set to 0 (OFF). Otherwise, the output of the EFFECTS RETURN channel will be routed back into its input, and oscillation may result.

9-1-5. RECORDING A LIVE MIX. There are several ways to accomplish the recording of a live mix.

1. Use a split snake to send all inputs to a separate mixer for recording.
2. Use the channel SENDS to either record the "dry" signal, or patch the sends to another mixer for processing before recording. The channel SENDS will be affected by any adjustment made to the input GAIN, MIC/LINE switch or −20dB switch.
3. Use the channel DIRECT SENDS for recording a signal that has been fully processed by the channel. These can be recorded directly, or remixed.
4. Use the SUB OUTPUTS for a direct 4-track recording mix. The MON 1 & 2 and AUX 1 & 2 OUTPUTS can also be used for a 4-track signal.
5. A stereo mix can be recorded from the main LEFT and RIGHT OUTPUTS.
6. Any combination of channel sends and submixes can be combined to create the desired recording format.

9-2. MONITOR MIXING

Four independent mixes are easily obtained for monitor use when the board is used as the primary sound reinforcement mixer, and even more mixes are available if the mixer is dedicated strictly to monitor use.

A. The MON 1, MON 2, AUX 1 and AUX 2 channels provide four independently mixed busses and outputs. The output level of each mix is controlled by the associated fader in the area of the board above the submaster section, and the outputs are available at the rear panel MON 1 & 2, AUX 1 & 2 OUTPUT jacks. Typically, the MON PRE/POST switches would be used in the POST position (post-EQ, pre-fader) and the AUX PRE/POST switches would be used in the PRE position (pre-fader, post-EQ). Sections 12-1 and 12-2 describe the jumper select options available for re-assigning the PRE/POST switches.

B. A fifth fully independent mix is available by assigning the channel output to one of the submasters. The mix of each channel is determined by the channel fader setting and the assigned submaster fader controls the output level. The output is available at the SUB OUTPUT jacks of the chosen submaster channel.

C. A sixth mix is possible by using the channel PAN control to split the channel output between two submasters. In this case, the channel PAN will control the relative signal levels sent to the two submasters, and the channel fader will adjust the level of that channel in both submaster mixes.

D. Occasionally, the output of only one channel is desired in a monitor (some lead singers request this). In this case the channel DIRECT SEND can be used as an output, or that channel's signal can be the only input applied to a buss.

E. More outputs are possible by using all four submasters, the PANS to the LEFT and RIGHT MAINS, and the MONO MAIN. In these cases, each mix will be similar to one of the other mixes, but will have a separate output and output level adjustment. This can be useful since each output can be used with a separate equalizer for improved feedback suppression.

9-3. PATCHING TWO BOARDS TOGETHER

9-3-1. DESCRIPTION. Two or more boards can easily be patched together. One is defined as the "slave" board, the other as the "master" board. The buss outputs of the slave board are patched to their respective inputs on the master board. Patching in this manner functions the same as adding extra channels to the master board. All major output functions are then controlled by the master board. The cue audio and cue control busses are also patched together. This allows the master board to monitor the slave cue system including cue light indication, headphones

9-3-1. DESCRIPTION, cont.
monitoring and meter display.

9-3-2. CONNECTIONS. The slave and master boards should be connected as follows:

SLAVE		MASTER
SUB 1 OUTPUT	◀ . . . . ▶	SUB 1 INPUT
SUB 2 OUTPUT	◀ . . . . ▶	SUB 2 INPUT
SUB 3 OUTPUT	◀ . . . . ▶	SUB 3 INPUT
SUB 4 OUTPUT	◀ . . . . ▶	SUB 4 INPUT
AUX 1 OUTPUT	◀ . . . . ▶	AUX 1 INPUT
AUX 2 OUTPUT	◀ . . . . ▶	AUX 2 INPUT
MON 1 OUTPUT	◀ . . . . ▶	MON 1 INPUT
MON 2 OUTPUT	◀ . . . . ▶	MON 2 INPUT
R MAIN OUTPUT	◀ . . . . ▶	R MAIN INPUT
L MAIN OUTPUT	◀ . . . . ▶	L MAIN INPUT
CUE AUDIO L OUTPUT	◀ . . . . ▶	CUE AUDIO L INPUT
CUE AUDIO R OUTPUT	◀ . . . . ▶	CUE AUDIO R INPUT
CUE CONTROL OUTPUT	◀ . . . . ▶	CUE CONTROL INPUT

All connections should be made using unbalanced shielded cable and ¼″ phone plugs.

9-3-3. LEVEL ADJUSTMENTS. On the slave board, all of the following output level faders should be set at these positions:

SUB 1, 2, 3, 4	0dB
AUX 1,2	0dB
MON 1, 2	0dB
LEFT & RIGHT MAIN	0dB
MONO MAIN	(OFF)

This will allow the channel faders of both boards to correspond to the same position vs. level.

The CUE LEVEL control of the slave board should be set by ear or meter to give similar levels for cue signals in either board. This will typically be approximately ¾ maximum right rotation.

9-3-4. CUE SYSTEM OPERATION USING TWO BOARDS. When two boards are patched together as described (9-3-2), the cue systems will function as follows:

1. Depressing a CUE switch on the slave board will override both slave and master headphones/meter assignment switch arrays, and apply the cue signal to the master headphones.
2. The master board cue system cannot override the slave board arrays. For this reason, the slave board arrays are typically set to CLEAR, and only the master board headphones/metering system is used.
3. The CUE LEVEL control of the slave board adjusts the level of the cue audio signal sent to the master board. The CUE LEVEL control on the master board controls the level of the cue audio signal from the slave board and the master board cue buss.

9-3-5. OTHER NOTES

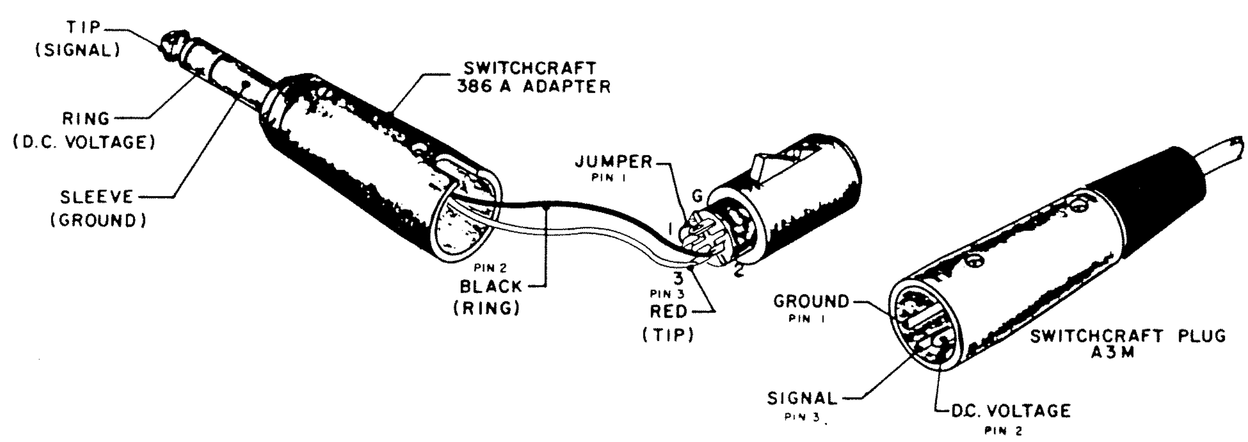
1. All unused channels should have the MUTE buttons depressed. It is also best to release all assign buttons and drop the channel faders to min. (DOWN).
2. The MONO MAIN fader of the slave board should be kept in the OFF position (DOWN). Even though the slave MONO OUTPUT is not used, its output stage can still be driven into clipping. This creates high frequency harmonics which can radiate into other outputs.

9-4. TALKBACK OPERATION

9-4-1. TALKBACK PATCHING TO BUSSES. The TALKBACK system in the SPL mixing board can be assigned to any buss by patching the TALKBACK OUTPUT jack to an unused preamp channel or to one of the EFFECTS RETURN channels. The talkback signal can then be assigned to the desired buss(es). The talkback signal can also be added to any buss by patching the TALKBACK OUTPUT to a direct buss input. For example, to connect the talkback signal to the MON 1 buss, patch from the TALKBACK OUTPUT jack to the MON 1 INPUT jack.

9-4-2. TALKBACK BETWEEN TWO BOARDS. The talkback systems of two SPL mixing boards can be patched together. In many live sound systems, two separate mixers are used—one for the main PA sound system and one for the stage monitor system. By patching a shielded cable between the TALKBACK OUTPUT jacks on the two mixers, both mixing board operators can communicate through the talkback system.

9-4-3. INTERFACING WITH EXTERNAL INTERCOMS. The TALKBACK system can be easily interfaced with many external intercom systems. The input/output impedance at the SPL mixer TALKBACK OUTPUT jack has been designed to be compatible with most intercoms. Normally, a female 3-pin to stereo ¼″ phone adapter will be required. The drawing for the wiring of the adapter is shown on next page.



In typical intercom system, the 3-pin connector is wired with pin 1 - ground, pin 2 - DC voltage, pin 3 - signal. Verify that the intercom uses this convention before wiring the adapter.

To wire the adapter, solder the following connections:

1. Connect the signal pin (3) to the tip of the ¼" plug.
2. Connect the DC voltage pin (2) to the ring of the ¼" plug.
3. Connect the ground pin (1) to the ground terminal (sleeve connection) by soldering a small jumper wire between the two terminals.

To connect the SPL mixer's talkback system to the intercom, patch the TALKBACK OUTPUT jack to the intercom input/output jack, using the described adapter if necessary. The board operator can talk over the intercom system by depressing the talkback ON/OFF switch and speaking into his microphone. With the ON/OFF switch depressed, the board operator can also listen to communication on the intercom system through his headphones. The call signal of most intercoms will normally light the CALL LED on the console.

9-5. HEADPHONES/METER OPERATION

The SPL 34/44 series mixers allow headphones monitoring of any channel or buss in the mixing console.

To monitor a preamp channel, engage the CUE switch located in that channel. For example, if CHANNEL 8 is to be monitored, engage the CHANNEL 8 switch. The cue buss will automatically override all other signals fed from the switch arrays to the headphones and display METERS 3 & 4. The cue LED's are reminders that a channel cue is switched on and that the signal sent to the headphones and METERS 3 & 4 is a cue signal. NOTE: More than one preamp channel may be cued at the same time.

To monitor a single buss through the headphones, engage the switch corresponding to that buss in one of the METERS 3 & 4 switch arrays. Next, engage the CLEAR switch located in the other switch array. Then place the STEREO/MONO switch in the mono position. For example, if the AUX 2 buss is to be monitored, engage the AUX 2 switch located in the METER 4 switch array. Then engage the CLEAR switch located in the METER 3 assignment switch array. Place the STEREO/MONO switch in the MONO position. The signal from the AUX 2 buss will now be present in both channels of the headphones.

To monitor two busses with the headphones, engage the switches corresponding to those busses in the METER 3 and METER 4 switch arrays. Signals on these busses will now be sent to the headphones. For example, if the SUB 1 and SUB 2 busses are to be monitored, engage the SUB 1 and SUB 2 switches located in the METER 3 and METER 4 switch arrays. Set the STEREO/MONO switch as desired. If the switch is set in the stereo position, SUB 1 will be in the left channel and SUB 2 will be in the right channel.

If the switch is in the MONO position, both channels of the headphones will carry the sum of the signals from the SUB 1 and SUB 2 busses. NOTE: It is possible to monitor more than two busses by simultaneously depressing more than one button of a switch array.

10. RECORDING

The SPL 34/44 series mixers can produce excellent results when used for studio recording. There are many different recording formats, most of which can easily be accommodated by the 34/44 series.

10-1. RECORDING IN STEREO. Creating a stereo recording is relatively easy. Simply adjust the mixer as you would for a stereo live mix. Patch the LEFT and RIGHT MAIN OUTPUTS to the left and right tape deck inputs. The tape deck outputs can be brought back into any available input for playback (spare channels, effects returns, etc.). Be careful to avoid patching errors (see section 11-1).

10-2. RECORDING IN 4-TRACK. The SPL 34/44 series mixers are ideal for 4-track recording, and can be set up in various ways, depending on requirements. One typical way is as follows:

A. Use the MON 1 & 2 and AUX 1 & 2 to develop the 4 mixes. Select the MON POST-EQ and AUX PRE-fader.

B. The MON 1 & 2 and AUX 1 & 2 OUTPUTS are connected to the recorder's inputs 1, 2, 3, 4. The AUX & MON faders control the output level to the deck.

C. The channel assigns and faders can be adjusted to provide studio monitoring through SUBS 1, 2, 3, 4. Mix SUBS down to LEFT and RIGHT MAIN for stereo monitoring.

D. There are several valid options for assigning tape deck playback returns.

1. Connect the deck outputs to EFFECTS RETURNS 1, 2, 3, 4. Adjust all AUX & MON controls on EFFECTS RETURN channels to 0 (OFF) to prevent oscillation when recording. The return channels can be assigned to SUBS 1, 2, 3, 4 if desired, or assigned directly to left and right mains for stereo playback.
2. Return to four unused preamp channels if playback equalization is needed. Again, set AUX & MON controls to 0 (OFF) on playback channels.

With either playback setup, the MIC/LINE switches, MUTE switches or LINE/RECORD switches can be used to activate or mute playback.

10-3. RECORDING IN 8-TRACK. The SPL 34/44 series can be used for 8-track recording and mixing. There are some options available, but the following setup is fairly straightforward.

A. The 8 outputs to the recorder are SUBS 1, 2, 3, 4, AUX 1, 2 and MON 1, 2. The channel assign switches and faders adjust SUB 1, 2, 3, 4 signals, and AUX & MON controls determine the other 4 outputs. The 8 faders in the submaster section (this includes the four AUX & MON faders) control signal levels sent to the deck.

B. The SUBMASTER MUTE buttons can be engaged to prevent the record signal from being sent to the MAINS.

C. The tape deck outputs are returned to EFFECTS RETURNS 1, 2, 3, 4, and four available channel inputs. These channels are assigned to MAIN.

D. Studio monitoring and playback signals are mixed to stereo (or mono) via the LEFT & RIGHT MAINS and channel PAN controls.

10-4. RECORDING IN MORE THAN 8-TRACK. For 16- or 24-track recording, typically little sub-mixing is done, and the channel signals are sent directly to the recording tracks. On each channel, the channel DIRECT SEND can be patched to the deck input, and the deck output patched to the channel LINE INPUT. Use the MIC/LINE switch to choose record or playback. The rest of the mixer's capabilities are used for studio monitoring. When mixing down, the normal format for mixing to 8, 4, 2 or 1 output(s) is followed.

10-5. LEVEL ADJUSTMENTS WHEN RECORDING. Adjusting for proper signal level is critical to recording. If the levels are too low, the signal gets lost in the noise. Levels which are too high cause distortion and tape saturation. There are many systems available for noise reduction, compression and expansion that help the problem, but proper levels are also critical to their correct operation.

The meters on the 34/44 series mixers are factory-set so that 0dB or 0 VU = 1V RMS into 600 ohms. Depending on whether the recording equipment is pro, semi-pro or consumer oriented, other meter calibrations may be required. The meters may be adjusted such that 0 VU (0dB) indicates any output level from 0.25V to 10V RMS. These should be set such that they give the most useful readings for the equipment being used (see section 12-5).

11. OPERATING HINTS & NOTES

11-1. PROBLEMS CAUSED BY PATCHING ERRORS. Due to the large number of inputs and outputs available in the SPL 34/44 mixer, it is relatively easy to create problems by simple patching errors.

The most common error is to connect the output of a buss back into an input of the same buss. As an example, let's assume that SUB 1 is used for the drum mix, and a Sunn SPL 4120 equalizer is to be used to adjust overall frequency response. The SUB 1 OUTPUT would be sent to the equalizer, the SUB 1 MUTE depressed to prevent the unequalized signal from being sent to the MAINS. The equalizer output would be brought back to an EFFECTS RETURN channel. If the EFFECTS RETURN channel SUB 1 assign button should mistakenly be depressed, the equalizer would probably oscillate, since its output would then be patched back into its input.

11-2. DELAY EFFECTS REGENERATION. There is only one instance in which patching the output of a buss back into the buss input can be desirable. This is when a post-fader output is patched to the input of delay, and the delay output is mixed back into the original buss in order to create multiple delays. For this to work, the delay must be adjusted to have no direct (undelayed) signal on the output. The delay output would be patched into either an EFFECTS RETURN INPUT, or an open channel INPUT. Often, an open channel is desirable for the delay return, since most delays tend to lose high frequencies with each delay, and the channel equalization can be used to restore this loss. Also, the channel level fader may then be used as a delay "regeneration" control. The higher the fader setting, the more delay regenerations are created. Caution must be used when connecting and adjusting the components of the delay "loop," as oscillation and regeneration "run-away" can easily occur.

11-3. INITIAL SETUP RECOMMENDATIONS. When first setting up the mixer, initial settings are critical for arriving at a satisfactory mix quickly. Often, to the distress of many soundmen, the show must begin without a preliminary sound check at all.

INITIAL SETTINGS:

GAIN. Try GAIN settings at approximately -20dB (straight up). On some drum mics, guitar amp mics, and other loud sources, start with the -20dB pad engaged. Be ready to adjust the GAINS quickly, since without proper gain adjustment, the rest of the mix cannot correct for the error. Also, start with relatively low GAIN settings, since any boosting of the EQ section will tend to boost the channel toward clipping.

MONITORS, AUXILIARIES. A good typical MON and AUX channel setting is 5, or again, straight up. Assuming that the GAIN control settings are corrected first, the MON and AUX busses should end up at similar levels for similar settings. The overall outputs of these busses can be increased slowly as corrections are made.

CHANNEL EQUALIZATION. For starters, most vocal, keyboard and other full-range signals are best left flat, with the 80Hz subsonic filter engaged, and the low shelving at 250Hz. Bass guitar, keyboard bass and kick drum signals may work best with the 20Hz subsonic filter engaged, provided the speaker system is capable of output in this range. In this case, the 100Hz low shelving selection is probably best. Often, the MID FREQ control is set low, 500Hz or so, and the MID level reduced or cut, to control the "muddiness" of these low frequency signals.

Of course, all of the equalization settings are mixed to achieve the most pleasing sound, but often, these initial settings will aid in accomplishing a good mix more quickly.

FADER SETTINGS. A good nominal channel fader selection is between −6dB and −10dB. These settings can be quickly corrected as the music gets started. Master faders are adjusted for a desirable sound level; however, most soundmen adjust them to about 0dB.

11-4. GROUNDING. To minimize system hum it is best to have only one system ground. In unbalanced systems, or in balanced systems having the shields of signal-carrying wires tied to ground at both ends, all electrically connected components will be ground-referenced to each other via the signal ground. If two or more of these components are AC ground-referenced through 3-prong AC plugs, a ground loop will exist between the AC ground system and the signal ground. Such a ground loop is usually the cause of system hum.

Many sound engineers remove this hum by "floating" all components but one, using 3-prong to 2-prong adapters on the AC cords. This system of grounding will remove the hum, but care must be taken to be absolutely sure that all components have one solid AC ground. Before using this AC floating technique, check local electrical code and building code where you are working, since AC floating may be prohibited. Be especially careful when using polarity switches on amplifiers as these can often be the cause of system failure and electrical shock to performers if not used properly.

NOTE: As it arrives from the factory, the signal ground and chassis of the SPL 34/44 mixer are floated with respect to the AC line. Therefore, if the mixer is chosen as the AC ground reference point, it will have to be modified. See section 7-1-8 for details.

11-5. OUTPUT LEVELS, DRIVE CAPABILITIES. All identical input and output connectors have been designed to have the same input and output characteristics. All balanced outputs will drive +24dBm (12.3V into 600 ohms), and all unbalanced outputs will drive 10V RMS into 10K ohms.

If any one balanced output must drive more than one 600 ohm balanced input, an impedance matching network should be used. Driving impedances below 600 ohms will result in reduced output level, variations in frequency response and increased distortion.

11-6. CHANNEL EQUALIZATION NOTES. The channel equalization system is described in section 2-1-7. The use of the equalization will be entirely up to the soundman. It is best, however, to try to use the EQ as sparingly as possible. High levels of boost will promote overloading, and detract from the naturalness of the sound.

11-7. LOW CUT SWITCH USAGE. The use of the low cut filter can often be the most important factor in live system reliability, power headroom, and sound quality. Many of the speaker systems used for live sound are not designed for efficient output below 80Hz. For a speaker to reproduce 20Hz, it requires approximately 4 times the excursion and power as it does to reproduce 80Hz at the same level. Horn-loaded systems, and bass reflex systems are especially susceptible to over-excursion at frequencies below their design cutoff, and the speakers in these enclosures can easily be destroyed by significant power input below their cutoff point. In general, use the 80Hz low cut filter unless your specific application requires output to 20Hz, or you are using a product that offers subsonic filtering such as a Sunn SPL 4320 digitally-controlled crossover.

11-8. LEVEL ADJUSTMENT NOTES. The balance of all of the level control settings in the system is critical to the quality of the mix. Running a level control too low decreases signal-to-noise ratio, whereas running a level control too high can cause overload distortion of either the output stage associated with the control, or of the input stage of the next section. A careful balance of the level controls is the key to a quality mix.

11-9. PATCHING TO EXTERNAL EFFECTS. Effects patching will generally fall into one of three categories: 1) effects in line with the input signal before the mixer, 2) effects loops within the mixer system, and 3) effects in line with the mixer output.

11-9. PATCHING TO EXTERNAL EFFECTS, cont.

Effects patching before and after the mixer is fairly straight-forward. The input signal is patched into the effect and the effect output goes to the next component in the chain.

Effects loop patching within the mixer system can be accomplished in many different ways. The signal can be taken from virtually any output, patched through the effect, and returned to a number of different inputs to be remixed. Typical effects include: 1) submixing all vocals to one submaster, patching the submaster send to a compressor, and returning the compressor output to the submaster return; 2) patching an AUX output to a delay and returning the output of the delay to an EFFECTS RETURN channel. Section 11-2 explains how to mix a delayed signal back onto the originating buss to achieve regeneration effects. Be aware that problems may arise from incorrect patching (see section 11-1).

11-10. BALANCED AND UNBALANCED INTERFACING

1. **TRANSFORMER BALANCING.** The Sunn transformer-balanced inputs and outputs used in the 44 series mixers are "floated," which means that they are not referenced to ground. This is advantageous when there is a possibility of stray DC or AC signals riding on any lines. A floated transformer system can reject these signals as long as they are common to both lines, regardless of the signal's relationship to true ground. Floating simplifies system grounding and provides greater immunity to hum. Also, since transformers are passive (non-powered) components, they do not introduce any noise to the system. Transformers also reject radio frequency interference well, and do not suffer from signal loss when driving unbalanced loads.
2. **ELECTRONIC BALANCING.** There are several good features to an electronically-balanced system such as that incorporated into the 34 series mixers. First, the rejection of common signals is excellent. In addition, well-designed electronically-balanced systems have good slew rate, low distortion and are relatively insensitive to problems caused by mis-matched impedances. However, since an electronically-balanced system is referenced to ground, there is an upper limit to the permissible voltage level that may be allowed on the signal-carrying lines.
3. **INTERFACING BALANCED AND UNBALANCED SYSTEMS.** When interfacing between balanced and unbalanced systems, proper wiring of the 3-pin audio connector to a 2-terminal connector is important. There are three types of balanced inputs and outputs: transformer-balanced floated (non-center-tapped and not referenced to ground), transformer-balanced non-floated (transformer center-tap grounded), and electronically-balanced. There are six types of interfacing to consider:

   A. Transformer-balanced floated outputs to unbalanced inputs.

   B. Unbalanced outputs to transformer-balanced floating inputs.

   C. Transformer-balanced non-floated outputs to unbalanced inputs.

   D. Unbalanced outputs to transformer-balanced non-floated inputs.

   E. Electronically-balanced outputs to unbalanced inputs.

   F. Unbalanced outputs to electronically-balanced inputs.

   In cases A, B and F, wire PINS 1 and 2 to ground, PIN 3 to the tip (+). In cases C, D and E, wire PIN 1 to ground, PIN 3 to tip, and leave PIN 2 open (there will be a signal loss of 6dB in this case). Transformer-balanced systems will interface directly with electronically-balanced systems and require no special wiring.
4. **PHASING.** There are two conventions for wiring 3-pin audio connectors. The accepted standard for the audio industry is; pin 1 ground, pin 2 minus (inverting), pin 3 plus (non-inverting). The majority of the pro-audio equipment available is wired in this fashion. However, some standards list: pin 1 ground, pin 2 plus (non-inverting), pin 3 minus (inverting). When interfacing equipment, the wiring of the connectors should be checked to insure proper phase relationships.

11-11. POWER SUPPLY RELIABILITY. The SPL 34/44 power supply is a tracking multiple-output DC source. Under turn-on and turn-off conditions, the SPL 34/44 will maintain tracking and control of all supply lines to minimize system pops that could potentially damage speakers. The supply will maintain full power operation even when AC line voltages vary −25% to +10%. The SPL 34/44 supply will also maintain full output when surrounding air is as high as 120°F. It is important however, that adequate ventilation is provided, especially when mounted in a rack.

The power supply has been designed such that any failure mode of the mixer or power supply will not be catastrophic. Should a given stage fail and draw excess current from the supply, either the external fuse or internal current limiting and fusing will protect the system. If the failure mode should short any two lines of the power supply together, it may cause a temporary malfunction, but will not cause other components to fail. If the system fault is corrected, the supply will return to normal operation. Though it may never be needed, some sound engineers like to carry an extra power supply.

11-12. PHASE INTEGRITY. This system has been designed such that there are no phase reversals from any input to any output. This is not always maintained in other manufacturers' equipment. Phase integrity is critical in a device that might be inserted in a loop: (compressors, delays, reverbs, equalizers, etc.) A phase reversal in this case will create a "dead spot" in a level control where the direct and mixed signal levels are equal. Also, if a component causes phase reversals between two speaker systems, a bass cancellation will occur.

11-13. PHANTOM POWER USE. When using phantom power, consideration must be given to the following:

1. Condenser mics compatible with +48V supplies.
2. Microphones not compatible with +48V supplies.
3. Mics not affected by presence of +48V supplies.
4. Mics requiring unique power supplies.
5. Other possible +48V supply interactions.

1. Condenser mics compatible with +48V supplies. The following microphones require phantom power for operation, and function well from a +48V supply:

AKG: C series, CE Series, C452E Series
BEYER: Electret Condenser Series
EV: PL77, PL77A, 1777, 1778, CS15 Series, PL76 (requires 506 adapter)
SONY: ECM Series, C74, C76, C38B, C37P, C500, C47P
NEUMANN: KM83, KM84, KM85, KM86, U87, U89
AUDIO TECHNICA: ATM31R, ATM91R
SHURE: SM81

2. Microphones not compatible with +48V supplies. The following mics will not function correctly when connected to a +48V supply and may be damaged by its presence.

NEUMANN: FET 70 Series, KTM
SENNHEISER: MMKH 110 (−1), MKT-T Series, MKH 435U, MKH 415, 815
Unbalanced microphones

3. Mics not affected by presence of +48V supplies. Any balanced, low impedance microphone is generally safe to use with a +48V supply.

4. Mics requiring unique power supplies. The following mics require special supplies for operation, and are not affected by the phantom supply.

ALTEC: M20, 21, 50, 51
AKG: C60, 12, 24
NEUMANN: KM,56 & 64, U47, & 67, M49 & 269
SONY: C37 & 37A

5. Other possible +48V supply interactions. Often other signal sources besides microphones will be plugged into the balanced inputs of the console. Most mixers, keyboards, guitar amps or other electronics **will be damaged** by connection to a +48V supply. See section 12-4 for defeating phantom supply.

ONE FINAL NOTE: To prevent system pops and possible damage to preamp channel inputs, it is best to avoid plugging and unplugging inputs when the phantom power is on.

12. OPTIONS, MODIFICATIONS & ADJUSTMENTS

12-1. MON PRE/POST JUMPER SELECT. The preamp channel MON PRE/POST switches are factory-wired pre-EQ in the PRE position and post-EQ in the POST position. By moving two jumper wires on the preamp channel boards these switches can be set to be pre-fader in the PRE position and post-fader in the POST position. To change the location of the jumper wires, unsolder and remove them from the factory-set position, and resolder them into the desired positions. The factory-set locations are indicated by solid lines around the jumper wires labeled MON PRE-EQ and MON POST-EQ. The optional positions are indicated by dotted lines labeled MON PRE-FADER and MON POST -FADER.

When changing the position of the jumper wires from the factory-set position, one jumper must be placed in one of the two PRE positions and the other jumper must be in one of the two POST positions.

NOTE: The POST-EQ position and the PRE-FADER position are electrically the same.

12-2. AUX PRE/POST JUMPER SELECT. The preamp channel AUX PRE/POST switches are factory-wired to select between PRE-FADER and POST-FADER. By moving the jumper wires on the preamp channel boards these switches can be changed to be PRE-EQ and POST-EQ. To change the jumpers, unsolder them from the factory-set position and resolder them into the selected positions. The factory-set positions on the boards are indicated by a solid line, labeled AUX PRE-FADER and AUX POST-FADER. The optional positions are indicated by dotted lines labeled AUX PRE-EQ and AUX POST-EQ.

As stated in 12-1, one jumper wire must be located in one of the two PRE positions and the other jumper must be located in one of the two POST positions. The PRE-FADER and the POST-EQ positions are electrically the same.

12-3. CUE PRE/POST FADER JUMPER SELECT. The CUE switches have been factory-wired to send a pre-fader mono signal to the headphones. A jumper select option is available to send a post-fader stereo signal to the headphones. The factory-set position on the channel boards is indicated by solid lines around the two jumper wires labeled CUE/PRE-FADER. The optional locations are indicated by dotted lines around the two jumper positions labeled CUE/POST-FADER. When changing the location of the CUE jumpers, **both** jumpers must be removed from the factory-set positions and reinstalled in the optional positions.

12-4. DEFEATING PHANTOM SUPPLY. To defeat the phantom power supply at a preamp channel balanced INPUT, unsolder one end of resistors R101 and R103 (6.49K 1% resistors) on the channel input board. Lift the ends of the resistors out of the circuit board and position them where they cannot touch any other component on the board. To defeat the phantom power supply at the TALKBACK MICROPHONE input, unsolder one end of resistors R644 and R645. (6.49K 1% resistors.) Lift the ends of the resistors out of the circuit board and position them where they cannot make contact with any other components.

12-5. METER CALIBRATION. The four LED displays on the 34 series have been calibrated at the factory so that 0dB = 1V RMS into a 600 ohm load, and the VU meters of the 44 series have been calibrated to indicate 0 VU at 1V RMS into a 600 ohm load. They can be recalibrated to read 0dB or 0VU at any output level from .26V RMS to 10V RMS by means of trim pots located on the two right-most circuit boards in the mixer. The far right board (the one with the METERS 2 & 4 switch arrays) has the trim pots for METERS 2 & 4, and the board containing the METERS 1 & 3 switch arrays has the trim pots for METERS 1 & 3.

On the first series of the SPL 4400 mixers, the bottom panel must be removed to access the trim pots. For the sake of orientation, we will call the end of the circuit boards towards the meters the top, and the end with the headphone/meter select switch arrays the bottom. The meter trim pots are located approximately ¼ of the way from the top of the boards. The trim pots nearest the top adjust METERS 1 & 2, and immediately below these are the METERS 3 & 4 trimpots.

On later 44 series mixers and on all 34 series mixers, top panel holes marked METER TRIM 1, 2, 3 & 4 provide access to these trimpots, and the bottom panel does not have to be removed.

To recalibrate the LED displays or VU meters, first select the four submasters on the meter select switch arrays. Calibrate each meter individually by injecting a 1kHz sine wave into the SUB INPUT corresponding to the meter being adjusted, and place a 600 ohm load across the corresponding unbalanced SUB OUTPUT. A 600 ohm load is easily created by wiring two 1200 ohm ¼w. resistors in parallel across the two terminals of a standard ¼" phone plug, and inserting this plug into the SUB OUTPUT jack. A 560, 630 or 680 ohm ½w. resistor may be used in place of the two 1200 ohm resistors, with slightly less accurate results.

With an AC voltmeter connected across the 600 ohm load, adjust the fader on the submaster channel until the AC voltmeter displays the level desired for 0dB (or 0 VU). Now adjust the proper trimpot until the display (meter) being calibrated reads 0dB (0 VU). Repeat this procedure for the other three displays (meters).

13. WARRANTY AND MAINTENANCE

13-1. CARE AND UPKEEP. The Sunn SPL 34/44 series mixers have been designed to give years of excellent operation. With a few precautions you can help to insure the mixer's reliable operation in portable applications. Sunn highly recommends that these mixers be transported in a Sunn hard shell ATA travel case. Periodically check the mixer for any loose screws and tighten if necessary, being sure not to overtighten.

If the SPL mixer is installed in a permanent installation, place a soft cover over the mixer when it is not in use. This will prevent dirt and other contaminants from getting into the unit causing noisy controls and switches. One caution cannot be over-emphasized—**do not set food or drinks on the mixer.** Spillage from these items can cause extensive damage to the mixer and gum up the controls. When using the mixer outdoors make sure it does not come in contact with moisture (rain, beer, sprinklers, etc.).

To clean the mixer, place a small amount of furniture polish on a soft cloth and wipe clean the mixer's surface.

The power supply cable connector is made of hardened plastic, and may break if items are set on it or if someone should step on it. Do not set any heavy objects on the front panel of the mixing console as this may cause warping of the panel and break some of its components.

13-2. WARRANTY SERVICE. If your Sunn SPL 34/44 series mixer is in need of service, contact your dealer for the location of the nearest Sunn Authorized Service Center.

13-3. SUNN MUSICAL EQUIPMENT COMPANY'S LIMITED WARRANTY

SUNN Musical Equipment Company warrants new products to be free from defective materials and workmanship for one year from date of purchase to the original owner when purchased from an AUTHORIZED SUNN DEALER according to the following conditions:

The purchaser is responsible for completing and mailing to SUNN, within 15 days of purchase, the warranty application enclosed with each product. Upon receipt of the warranty application, SUNN will issue a warranty validation sticker that must be affixed to the product. Where a warranty validation area has not been provided on a few SUNN products, the validation sticker is to be affixed to your original proof of purchase and presented at the time of warranty service. PROOF OF PURCHASE ON UNREGISTERED EQUIPMENT IS NOT SUFFICIENT FOR RECEIVING IN-WARRANTY SERVICE. In the event you do not receive your validation sticker within 60 days of mailing, you are to notify SUNN Musical Equipment Company in writing immediately. The purchaser has the sole responsibility of completing and mailing the warranty application.

Light bulbs and meters carry a 90 day warranty from date of purchase.

SUNN Products that have been subject to accident, alterations, abuse, rental or defacing of the serial number are not covered by this warranty. Loudspeakers and drivers misuse due to overpowering or improper installation resulting in torn, burned or charred components will not be covered by this warranty.

The normal wear and tear of appearance items such as handles, corners, casters, and knobs are not covered under this warranty.

If your SUNN product requires service during the warranty period, SUNN will repair or replace, at its option, defective materials provided you have identified yourself as the owner of the validated product to any SUNN Authorized Service Center or contact SUNN for service assistance. Transportation charges to and from an Authorized Service Center or factory for SUNN products and components to effect repairs shall be the responsibility of the owner. In the event a product is to be returned to SUNN for repairs, a written return authorization from Sunn must be obtained prior to shipping.

SUNN is not liable for any incidental or consequential damages resulting from any defect or failure of this instrument other than the repair of the SUNN product subject to the terms of this warranty. This warranty gives you specific legal rights and you may also have other rights which vary from state to state. This warranty is expressly in lieu of all other agreements and warranties, expressed or implied, except as may otherwise be required by law.

Thank you for choosing SUNN!

14. GLOSSARY

AC (Alternating Current)

An electric current with periodically changing polarity (for example: 60 times a second for 60Hz power mains, etc). Also, AC generally refers to the power main supply. 60Hz, 120VAC in the U.S.

AMPLIFIER, POWER

An amplifier which is designed for driving loudspeakers and consequently has higher power output capability than preamplifier.

AMPLITUDE

Another term for "level" or "volume" of an electrical or acoustical signal. A measurement of the height of a waveform.

ATTACK TIME (Musical)

The time it takes for a note to reach its full volume. Percussive instruments have short attack times (reach maximum volume quickly) and wind instruments have long attack times (reach maximum volume more gradually).

ATTENUATE

To reduce the level of an electrical signal, usually with a volume or loudness control. Also, to reduce sound levels acoustically through the use of absorption materials.

AUDIO FREQUENCY

Any frequency within the range of normally audible sounds, roughly from 20Hz to 20,000 Hz (Hz = cycles per second).

BALANCED

An audio circuit with 3 wires; two wires carry the signal, non-inverted (+) and inverted (−), and the third is a shield which is connected to chassis or system ground.

BANDWIDTH

Refers to the "space" in the frequency response of a device through which audio signals can pass (between lower and upper frequency limits, those points where the signal level has rolled off 3 dB).

BASS

The low audio frequency range, normally considered to be below 500 Hz.

BASS REFLEX

A type of speaker enclosure in which the speaker's rear wave emerges through a vent or port to reinforce the bass (low frequencies).

BIAMPLIFICATION

In a conventional sound system the full range audio signal passes through one amplifier and feeds a high-level crossover within the speaker that divides the audio to feed the low and high frequency drivers. In contrast, a biamplified system utilizes a low level crossover or electronic crossover to divide the full range audio signal into two parts for feed into two separate power amplifiers, one for low and one for high frequencies. The output of the high frequency amplifier feeds the high frequency driver and the output of the low frequency amplifier feeds the low frequency driver.

BUSS (Also Bus)

A signal path to which a number of inputs may be connected for feed to one or more outputs. In a mixing console a buss is usually a long piece of wire to which any input channel may be connected by means of a switch or control. The end of the wire goes into a combining amplifier or summing amplifier which then feeds one of the console outputs.

CENTER FREQUENCY

The frequency at which the greatest amount of boost or attenuation (cut) occurs in a peak/dip type equalizer or a notch filter.

CLIPPING

Occurs when the capabilities of an amplifier are exceeded. The result is very audible distortion, also visible on an oscilloscope.

CORNER FREQUENCY

The upper frequency limit, usually defined as the point where the level falls to 3dB below the midband reference level.

CROSSOVER NETWORK

A unit which divides the audio spectrum into two or more frequency bands.

LOW LEVEL CROSSOVER NETWORK. A crossover network designed to operate at line levels and which is placed before the power amplifier(s); used in bi- or tri-amplified speaker systems. May be an electronic crossover.

HIGH LEVEL CROSSOVER NETWORK. A crossover network designed to operate at high levels and which is placed between the power amplifier and the loudspeakers (the type of crossover normally built into a speaker system).

CROSSOVER NETWORK, ELECTRONIC

A line level (low level) crossover network which has built-in-preamplifiers to avoid signal losses and/or to feed long audio cables. (See "crossover network, low level".)

CROSSTALK

Undesired signal appearing in one channel as a result of leakage from another channel, usually specified in dB.

CUE BUSS

In a mixing console, the buss or channel which is used to feed a program for signal monitoring. Also known as the foldback buss.

CUE CIRCUIT (Foldback Circuit).

CHANNEL CUE. A circuit which enables one to monitor an input position with the fader down (no program feed); for program identification or troubleshooting.

COMMUNICATIONS CUE. An intercom system made up of amplifiers, headphones, microphones and signaling lights or tones; for coordination of sound, lighting, stage and other personnel. Not for the actual audio program.

PERFORMER HEADPHONES CUE. In recording studio work, a mono or stereo mix which is fed to performers so they can hear other performers and themselves, or can monitor a previously recorded program (tracks) for over-dubbing.

CUTOFF FREQUENCY

An upper or a lower limit of useable frequency response beyond which signals cannot pass; usually created by a filter network.

dB

Abbreviation for decibel, expresses two quantities as a ratio. The dB is based on a logarithmic power curve, which is convenient for comparing electrical and acoustical signals. For comparing two **powers** (P1 & P2) the dB formula is:

$$dB = 10 \log \frac{P1}{P2}$$

For comparing voltages, pressures, currents etc. (X1 & X2)

$$dB = 20 \log \frac{X1}{X2}$$

The reason for the difference between the 10 multiplier for powers vs. 20 for other quantities is that:

$$Power = \frac{(Voltage)^2}{R} = (current)^2 R$$

The 20 multiplier for non-power quantities accommodates the squared factor, so that both formulas express equivalent terms.

A) dBV is a **voltage** ratio. It is normally used to compare any voltage (E) to 1.0V RMS, which is 0dBV. The formula is:

$$dBV = 20 \log \frac{E}{1.0V}$$

B) dBm is a **power** ratio. The 0dBm reference is 1 milliwatt (0.001W) across a 600 ohm load, which is 0.775V RMS. When a voltage (E) is known, to find dBm, first calculate

$$P1 = \frac{(E)^2}{600}$$

Then calculate: $dBm = 10 \log \frac{P1}{.001W}$

C) dB SPL is an acoustic pressure ratio (not power). 0dB SPL is the approximate threshold of human hearing (0.002 dynes/cm^2). An increase of 10dB SPL is generally agreed to be twice as loud to the human ear, which requires a power increase of 10X. Doubling the power increases dB SPL by 3dB.

DC (Direct Current)

An electrical current of constant and uniform polarity, unlike AC (alternating current), which regularly changes its direction and polarity.

DIRECT SEND

Refers to an audio path where the signal is routed from one point directly to another without passing through additional amplifiers or signal processors. A direct send from a mixing console generally refers to a feed from each mixing position (channel), usually after the preamp and equalizer, that is applied directly to another point without going through the console's submaster and/or master mixing busses as normal.

DISTORTION

Distortion is an unwanted change that occurs to an audio signal, causing frequencies to appear at the output that were not present at the input of the audio device. There are several types of distortion.

DISTORTION, IM (Intermodulation)

Distortion signals that result from the interaction of at least two input signals. The distortion components are generally not whole number multiples, not harmonically related to the input signals, and are usually more objectionable than harmonic distortion components.

DISTORTION, THD (Total Harmonic Distortion)

The numerical sum of all harmonic distortion components in a signal. Usually measured in percent, but may be measured in dB below the fundamental frequency.

DYNAMIC RANGE
The difference, in decibels, between the loudest and quietest portions of a musical performance (or between the maximum signal level and the noise floor of electronic equipment).

EFFECTS BUSS, AUXILIARY BUSS
In a mixing console, the buss or channel which is used to feed a program to special signal processing equipment or effects devices. Also known as the effects send buss. Typical effects devices include delay lines, phasers, and flangers, as well as echo/reverb units.

EFFECTS RETURN
A mixing console input that receives the signal from an effects device. Many mixing consoles have a level control to adjust the amount of effects signal returned to the program mix; this control is called the effects return control.

EFFECTS SEND, AUXILIARY SEND
A mixing console output that sends a signal to the input of an effects device. Most mixing consoles have an effects send level control for each input position (channel) so the musical mix sent to the device can be adjusted; many consoles also have a master effects send control for the effects send buss.

EIGHT-TRACK TAPE RECORDING FORMAT
Either of two professional tape recording formats in which eight independent channels can be recorded in the same direction.

8-TRACK HALF-TRACK. Eight channels on 1-inch wide recording tape.

8-TRACK QUARTER-TRACK. Eight channels on ½-inch wide recording tape.

EQUALIZER (EQ)
An electronic device that will amplify (boost) and/or attenuate (cut) certain portions of the audio frequency spectrum.

FIXED FREQUENCY. An equalizer which operates at one or more specific frequencies which can be boosted or attenuated.

GRAPHIC EQUALIZER. An equalizer which operates simultaneously at a number of preset frequencies, any of which may be boosted or cut independently of the others. Often done at standard 1, ½, ⅓, or ⅙ octave center frequencies. Graphic equalizers are usually peaking-type equalizers.

PEAKING EQUALIZERS. A boost or cut characteristic which has a bell-shaped response. (The shape of the "bell" determines the "Q" of the equalizer.) Maximum boost or cut occurs at the center frequency, and there is less effect at frequencies farther away, above or below the center frequency.

FADER
A potentiometer that controls the signal level for a console input position or output channel. Generally considered a linear (straight line) control.

FEEDBACK, ACOUSTIC
Unwanted interaction between the output and input of an audio system (i.e., between loudspeaker and microphone or turntable cartridge) which can lead to "howling".

FILTER
A device which attenuates certain portions of the audio frequency spectrum.

BAND PASS FILTER. A filter which has high frequuency and low frequency cutoff points, and which only passes signals between these two points.

HIGH PASS FILTER. A filter which will only pass frequencies above a certain cutoff frequency point. (Also known as low-cut-filter).

LOW PASS FILTER. A filter which will only pass frequencies below a certain cutoff frequency point (Also known as high-cut-filter).

FOLDBACK
Another term, usually on European mixers, used interchangeably with "cue" in recording work or "monitor send" in sound reinforcement work.

FREQUENCY
The rapidity of change in current or voltage in an electrical signal or of air pressure in an acoustical (sound) signal. Frequency is measured in cycles per second; 1 cycle per second (cps) is 1 Hertz (Hz). The higher a note on the musical scale, the higher its frequency.

GAIN
The amount an amplifier increases the power of a signal, usually specified in dB.

GROUND
Any point in an electrical or electronic device used as the zero voltage reference.

AUDIO GROUND (audio common). The zero voltage reference or audio signal "low" or "-" path. This is usually connected to the chassis or electrical ground at some point.

SYSTEM GROUND. One point to which the grounds from various pieces of equipment in an audio system are connected. The system ground is generally the best point to connect to earth ground.

GROUND LOOP

A condition when two or more paths to ground exist and a voltage is induced unequally in these paths causing hum, buzz, or noise.

GROUPING

A mixing process where the signal level from two or more inputs can be adjusted simultaneously using a single control. A typical application would be the grouping of several microphones inputs for related instruments (i.e., a drum set), where individual mics are balanced in relation to the others using the input faders, and then an overall level adjustment is done using a group fader. Grouping of this nature simplifies mixing.

HARMONICS

Harmonics are integer multiples of a fundamental frequency. The first harmonic is the fundamental. The second harmonic is twice the fundamental frequency. The third harmonic is three times the fundamental frequency, and so forth.

HEADROOM

"Headroom" refers to the difference between the nominal operating level and the maximum level at any point in an audio system or device, usually expressed in dB.

HERTZ, KILOHERTZ

Abbreviated "Hz", the unit of measurement for frequency; 1 Hz is equal to one cycle per second (cps). 1 Kilohertz (kHz) is equal to one thousand cycles per second.

HUM

A low frequency tone, usually a multiple of the 50Hz or 60Hz power mains frequency (i.e., 120Hz, 180Hz, etc).

IMPEDANCE

The total opposition to the flow of alternating current in an electrical circuit. Impedance is measured in ohms.

IMPEDANCE, INPUT

The impedance "seen" by the signal source connected to the input of a device. Same as "load impedance".

IMPEDANCE MATCHING

The term "impedance matching" describes a technique or a device which makes the output impedance of one circuit approximately equal to the input impedance of the next.

IMPEDANCE, OUTPUT

The impedance of a signal source, such as the output of an amplifier (not the impedance of the input to which the device is connected). Same as "source impedance".

INPUT LEVEL

Refers to the level in dB, dBm or volts that is acceptable for an input signal to a particular connector in any given piece of electronic equipment. Such input levels are rated either as maximum (i.e., the level above which overdrive distortion occurs) or as nominal (i.e., the average level which would be fed to the input under normal operating conditions).

JACK

Female receptacle for a plug (male type connector).

kHz

Abbreviation for kilohertz, or one thousand cycles per second. Formerly called kilocycles (kc).

LED

Light Emitting Diode. A solid state device that emits infrared or visible light when a current flows through it. Usually red, green or amber, LEDs are available in other colors.

LEVEL

A term loosely used to describe the amplitude of a signal or a sound. More precisely, it is the value of that signal or sound relative to a given reference, expressed in dBm, dB SPL, etc.

LEVEL INDICATOR

A device which provides a visual display of the signal amplitude. This can be a meter, a series of light emitting diodes (LEDs), etc. Level indicators may respond to the peak, average or RMS value of the signal. A VU meter is a level indicator with a carefully specified average response. (See "VU meter").

LINE INPUT

An input designed to operate at line levels (see below) as opposed to microphone or speaker levels.

LINE LEVEL

Line level refers to the nominal (average) operating level of an audio system, and generally corresponds to a "0 VU" meter reading. There are several widely-used standard line levels:

A) +8dBm (1.95 volts RMS), for broadcast.
B) +4dBm (1.25 volts RMS), for most professional recording and sound reinforcement.

C) -10dBV (310 millivolts RMS), an alternate for professional recording.

LINE OUTPUT

Any output which delivers a signal at the nominal line level (see above), not microphone or speaker level.

LOAD IMPEDANCE

See "impedance, input".

LOW CUT FILTER

See "filter, high pass".

LOW-PASS FILTER

See "filter, low pass".

MATCHING IMPEDANCE

When the actual output impedance of a source device is approximately equal to the actual input impedance of the device it feeds, the condition is said to be an impedance match. Calculating or adjusting circuit values, or using special transformers to achieve this condition is said to be a process of impedance matching.

Note that many output circuits are not capable of driving an input of the same impedance. In this case, even though the impedances match, the condition would be considered an impedance mismatch. Such circuits must be "bridged" rather than "matched"—connected to an impedance of anywhere from 7 to 50 times higher in value, or more. A properly bridged circuit, while the impedances do not actually match in value, does not represent an impedance mismatch.

MIC LEVEL

The signal level in dBV, dBm or voltage that would approximate the nominal level of a microphone. While the nominal levels of microphones vary tremendously as a function of the sound level present and the type of mic involved, mic level is generally considered to be somewhere around 0.001 to 0.003 volts (about -60dBV to -50dBV). It is not unusual, however, for a mic to generate a $+4$dBV in the presence of a very loud sound.

MIX

The procedure whereby two or more signals from live and/or recorded sources are combined to achieve a desired balance. Mixing consoles provide separate level controls for each source, as well as overall controls for the mixed (combined) signal. Consoles may also provide equalization and auxiliary signal processing such as echo and reverberation.

MONITOR MIXER

A mixer or mixing console that is used primarily for balancing the sound fed to monitor speakers. Usually used for stage monitor speakers during live performances, where the on-stage sound balance requirements are substantially different from the house sound requirements.

MONOPHONIC (Mono)

Single channel sound.

MUTE SWITCH

A switch that stops signal flow: commonly found on input and output positions of mixing consoles. Similar to a channel ON/OFF or kill switch. The mute switch accomplishes the same function as bringing the fader to zero, but avoids disturbing the mix setting.

NOISE

Any unwanted signal, such as hum, hiss, rumble, crosstalk, etc.

NOISE, EQUIVALENT INPUT (EIN)

A measure of how quiet a microphone preamplifier is. To measure EIN, one actually measures the noise at the output of a device and subtracts the amount of amplifier gain in dB. Additional mathematics are required to account for different impedances. Usually the input source impedance is specified as 150 ohms for a mic input, and the EIN is specified in dBm. The theoretical lowest value for EIN in any preamplifier (the noise floor) depends on how the noise is expressed. Expressed as noise power, the lowest EIN is -124.8dBm (0dBm re. 1mW). Expressed as noise voltage (in dBV), the lowest EIN will vary with the source impedance - [-135.6dBV @ 50 ohms or -136.8 @ 150 ohms (0dBV re. 0.775V)]. If expressed in dBV, subtract 2.2dB from the "dBV" value.

NOISE FLOOR

The level at which noise exists in an electronic device, audio system or tape recorder, commonly measured in the absence of signal.

NOISE, WEIGHTED

The measured noise level in electronic equipment or in an acoustic environment, using a measuring device which includes any of several standard filters that restrict response to the audio spectrum or a portion thereof.

OHM

The unit of measure of electrical resistance or impedance.

OHM'S LAW

A group of mathematical formulas which describe the relationship between resistance (R) or impedance (Z) in ohms, electromotive force (E, voltage), current (I, amperes), and power (P, watts).

For calculating voltage: $E = IR$

For calculating power: $P = EI$ or $P = E^2/R$ or $P = E^2/Z$

For calculating current: $I = E/R$

For calculating resistance or impedance: $R = E/I$ or $Z = E/I$

OSCILLATION

The re-amplification of a signal by a gain stage, when the output of a gain stage is re-introduced into the input of that stage, and reamplified. Acoustic feedback is a form of oscillation.

OSCILLATOR (Audio Frequency Oscillator)

A device for producing audio-frequency tones or signals. Usually sine waves (pure tones), but can be square waves, triangle waves, sawtooth waves, etc.

OUT OF PHASE

When two related signals (or a single signal which has been split) are offset in time acoustically or electronically, they may be said to be out-of-phase (larger offsets that enable two signals to be audibly differentiated from each other are considered to be time delays). Depending on the actual phase difference, the out-of-phase signals may cancel or reinforce one another when combined.

When two circuits are reversed in polarity with respect to one another, sometimes they are mistakenly considered to be 180 degrees out-of-phase. Even though there is no real offset in signal arrival time here, the result is similar to a true 180 degree phase difference—the signals will cancel one another if combined.

OUTBOARD

Refers to electronic equipment or signal processors that are not incorporated in the mixing console. External equipment. Outboard equipment includes special equalizers, delay lines, reverbs, compressors, etc.

OUTPUT IMPEDANCE

See "impedance, output".

OUTPUT STAGE

The last section of any amplifier system, whether line amplifier or power amplifier.

OVERDRIVE LEVEL

The level at which a signal source exceeds the capability of the device which it is feeding. A level at which clipping may occur.

OVERLOAD

What occurs when a device is "asked" to supply more power than it is capable of delivering; usually caused by connection of too low an impedance to the device's output. May cause distortion and/or damage.

PA

Stands for "public address" system—generally applies to any sound reinforcement system, music or otherwise.

PAD

A passive resistor network which reduces the power level of a signal. In some cases a pad is utilized to match unequal input and output impedances for proper interface.

PAN

Abbreviation for panorama. A procedure whereby any mono signal can be placed in a stereo or quadraphonic perspective.

PAN POT

The control that places a signal in the stereo or quadraphonic perspective (see "pan").

PASSBAND

That range of frequencies which can pass through a filter, usually a bandpass filter. The cut-off or edge(s) of the passband are considered to be defined by the frequency(ies) where the response is down by 3dB from the level in the middle of the passband.

PATCH (Patching)

A process of routing or rerouting audio signals using patch cords. Also, in electronic music synthesizers, the combination of connections and control settings that produces a particular sound.

PATCH CORD

A short audio cable with a male plug on each end, and commonly used for audio signal routing (patching) between nearby electronic devices, or between various jacks on a mixer or patch panel (patch bay).

PHANTOM POWER

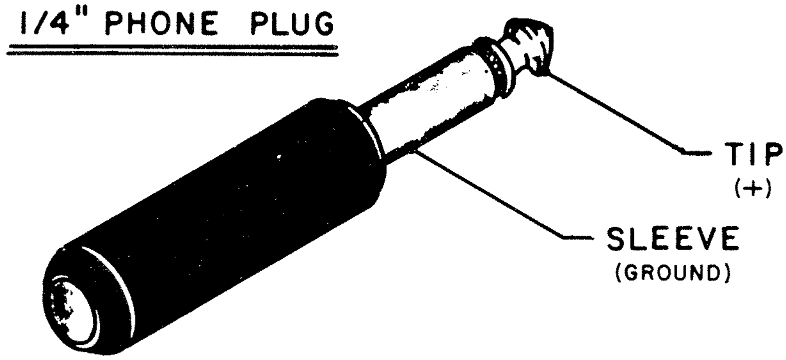
A method of remotely powering the preamplifier or impedance converter which is built into many condenser microphones by sending voltage along the audio cable. Phantom power is usually from 6 to 8 volts DC and is run along the same conductors that carry the mic signal. The DC is separated from the audio by using capacitors and special transformers.

PHASE

Phase describes the relative position of two sound waves with respect to one another, usually measured in degrees. One complete cycle of a sine wave is considered to be 360°, ½-cycle 180 °, etc.

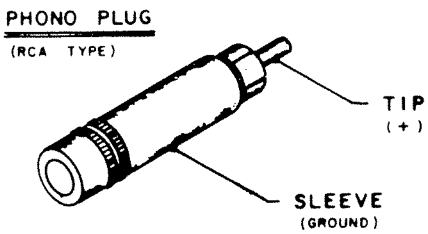
PHONE PLUG (Or jack)

Describes any of several varieties of 2-conductor or 3-conductor audio connectors having a ¼" shaft diameter. The term phone plug is derived from the original use of these connectors in early telephone switchboard equipment. The male connector is the plug and the female is the jack.



PHONO PLUG (Or jack)

A 2-conductor connector commonly used for unbalanced audio signal connections. Also known as an "RCA plug".



PLUG

Male connector for insertion into a female jack (receptacle).

POST

Refers to a signal input, output or routing point that comes after something. Usually used in conjunction with another term, such as post fader, post-EQ, etc.; if unstated, one may assume the term means post fader.

POST EQ

Refers to the signal point after the equalizer in a mixer or console. A circuit which is fed post EQ will be subject to any adjustments of that equalizer.

POST FADER

Refers to the signal point after the fader in the mixer or console. A circuit which is fed post fader will be subject to any adjustments of that fader.

POWER HANDLING CAPABILITY

In speaker systems, the maximum amount of power that can be safely accommodated without damage. The power handling capacity will vary depending on the frequency and length of time the signal is applied

PRE

Refers to a signal input, output or routing point that comes before (precedes) something. Usually used in conjunction with another term, such as pre-fader, pre-EQ, etc.; if unstated, one may assume the term means pre-fader.

PRE-EQ

Refers to the signal point preceding the equalizer in a mixer or console. A circuit which is fed pre-EQ will not be subject to any adjustments of that equalizer.

PRE-FADER

Refers to the signal point preceding (ahead of) the fader in a mixer or console. A circuit which is fed pre-fader will not be subject to any adjustments of that fader.

RESISTANCE

The opposition to flow of AC or DC electric current, measured in ohms.

SENSITIVITY

MIXER INPUT SENSITIVITY. The microphone or line input level which will drive the mixer output to the nominal rated line level. Usually measured in dBm, dBV or volts.

POWER AMPLIFIER INPUT SENSITIVITY. The input voltage which will drive the amplifier to its maximum rated output power. Usually measured in dBm, dBV or volts.

LOUDSPEAKER SENSITIVITY. The sound pressure level a speaker produces when fed by a given input power, measured at a specified distance on-axis (directly in front of the speaker). Usually specified in dB SPL at 1 meter, 4 feet or 30 feet, and with a 1 watt or 1 milliwatt input signal.

MICROPHONE SENSITIVITY. The voltage produced when the microphone is exposed to a specified sound pressure level. Usually specified in dBV in a 94dB SPL or 74 dB SPL sound field, measured with no load on the microphone. (EIA microphone sensitivity equation takes into account the impedance to which the microphone is connected.)

SHELVING

A boost or cut characteristic which has a response curve resembling a shelf. Maximum boost or cut occurs at the indicated frequency, and remains constant at all points beyond that frequency.

SIGNAL PROCESSING EQUIPMENT

Any equipment or circuit that is used to intentionally change the characteristics of a signal (other than overall level). Signal processors consist of such devices as equalizers, limiters, phasers, flangers, delay lines, etc.

SIGNAL-TO-NOISE RATIO (S/N)

The difference between the nominal or maximum operating level and the noise floor, specified in dB (decibels). Values may be weighted or unweighted. Sometimes equivalent to dynamic range.

IN A MIXER OR OTHER SIGNAL PROCESSING EQUIPMENT. Most commonly specified as the difference in dB between the nominal output level and the noise floor at the equipment output. Generally measured using one input assigned to one output, unless otherwise noted. Not equal to the dynamic range (which measures the difference between the maximum output capability and the noise floor).

SINE WAVE

The fundamental waveform of a pure audio tone.

SLEW RATE

The ability of an amplifier to follow a fast rising waveform. Usually measured in volts per microsecond.

SLIDER

Another term for a straight-line fader (see "fader"). Also may refer to any straight-line control, as opposed to a rotary control.

SLOPE

In a filter or equalizer, a description of the rate of boost or attenuation. Normally specified in dB/octave (6, 12, 18 or 24dB/octave slopes are most common). The steeper the slope, the higher the "Q" in a filter.

SNAKE

A multi-channel audio-cable intended for use with microphone level signals or line level signals.

SOLO (Also Cue)

A feature of many mixing consoles that allows one input signal, one buss, or a combination of "soloed" signals to be monitored exclusive of all other signals without disturbing the main output mix. For example, in the middle of the show with an 18 in X 2 out console, an engineer might listen to one microphone (perhaps to check for a problem, to see what it is picking up, or to adjust the input position's EQ). He merely engages that input position's solo button—the sound reinforcement and/or tape recorder feeds are not affected.

SPL

An acoustic measurement of the sound energy, or "sound pressure level", usually measured in dB SPL. Not the same as loudness, which involves subjective measurement based on the human ear's differing sensitivity at different frequencies and levels.

Also abbreviation for "Sunn Professional Line."

SOUND REINFORCEMENT

A general term for the amplification of sound for live performances; also called "PA" for public address.

STAGE MONITOR

A speaker system used on stage during a live performance so the performer can hear herself or himself and/or other performers.

SUB-MASTER

A level control or fader which adjusts the signal level on a given output channel, prior to the master fader (which is normally a multi-channel fader). For example, there might be a Left Channel Submaster, a Right Channel Submaster and a Stereo Master.

SUBSONIC

Actually means "below the speed of sound", but the term is generally used to describe " infrasonic", which means below audible frequencies: Infrasonic or subsonic frequencies of below 20Hz may not be audible, but they can cause audible distortion or damage in electro-acoustic equipment.

SUNN

A company dedicated to the design and manufacture of quality audio and lighting systems.

SYSTEM GROUND

The main ground point for any large recording or sound reinforcement system. The use of a single point ground rather than several ground points is an aid in the prevention of "ground loops", multiple paths to ground that can lead to hum, buzz and RFI.

TRANSFORMER

A transformer changes electrical energy at its input (primary winding) into magnetic energy in its core. This magnetic energy is transformed back into electrical energy at the transformer's output (secondary winding). Transformers are specified by input and output impedances (in ohms) or by winding ratio. If the secondary winding has more turns of wire than the primary, it increases the voltage (signal level); if it has fewer turns than the primary it decreases the voltage. There are many types of transformers designed for different purposes.

LINE INPUT. A transformer with a low primary impedance, typically 600 ohms, and a typical secondary impedance of 600 to 15,000 ohms. Designed to operate at line levels, to balance an input line and in some cases to provide voltage gain.

LINE OUTPUT. The output transformer of a line level amplifier, typically having a 1:2 through 1:4 voltage step-up ratio (1:2 to 1:4 ratio of primary to secondary impedance) to increase the output level. Sometimes an isolation transformer.

MATCHING. A transformer used to "match" unequal source and load impedances. The transformer's primary impedance is the same as the source impedance of the device which feeds it, and the secondary impedance is the same as the input impedance of the device to which it is connected.

MICROPHONE INPUT. A transformer with a low primary impedance, typically 1,000 to 1,500 ohms (for 150-250 ohm microphones). Designed to operate at low levels (microphone levels) and to increase the signal level.

TRANSIENT

An abrupt increase in signal level. In music, the result of a percussive or plucked instrument such as a drum head being struck or a guitar string being picked. It is desirable to preserve these musical transients. However, there are unwanted electrical transients such as the "click" caused by a switch, or the "pop" caused by a scratched record.

TREBLE

The high audio frequency range, normally considered to be above 5000Hz.

UNBALANCED

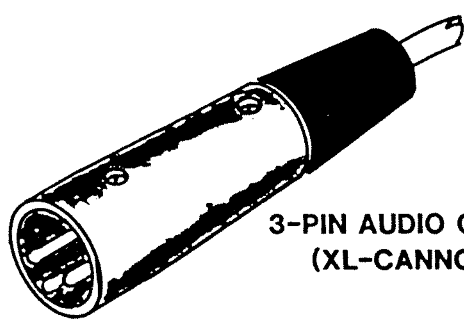
An audio circuit with 2 wires; one wire carries the signal, high (+) and the second carries the low (−) and also is connected to chassis or system ground. Since the signal leads are not of equal potential difference from the ground, they are considered to be "unbalanced" with respect to ground.

VU METER

A meter designed to measure audio level in volume units. VU meters may be calibrated to various "0 VU" standard levels (e.g., 0 VU = −10dBV, +4dBm, +8dBm, etc.) The VU scale may also be calibrated in percent, based on 100% modulation for broadcast transmission. True VU meters should always have a specified standard "ballistic" or pointer reaction behavior, chosen because it provides a good indication of average level as it corresponds to percieved loudness. Such meters, however, are relatively "slow" devices so brief transients often exceed the indicated level by 10 to 20dB.

XL-TYPE CONNECTOR (XLR)

Describes any of several varieties of audio connectors having 3 or more conductors plus an outer shell which shields the connection and locks the mating connectors. 3-pin XL-type connectors are commonly used to make balanced mic and line level connections in professional audio systems. The XL-type connector is sometimes called a "cannon" connector, so named for the original manufacturer although a number of companies now manufacture compatible 3-pin connectors.



3-PIN AUDIO CONNECTOR (XL-CANNON TYPE)

Y-CONNECTOR

A 1:2 connector that allows one signal source to be split and routed to two different inputs. Also called "Y-adapter".

Z

An abbreviation for "impedance". See "impedance".

Sunn would like to thank CAMEO for their assistance in assembling the glossary.