

SL24 SL40 SL56

Owners Manual

Revision 1 76-00024



Studio Live Consoles

Introduction

Sound Quality and In-Line Flexibility

There are a multitude of considerations that a sound engineer must weigh when choosing the right console. Whether the console will be used for live sound reinforcement or studio recording, ultimately the choice of a console always focuses on sound quality and signal routing flexibility. These were primary design considerations for the new S/L series consoles from Carvin.

Sound

Carvin's S/L consoles feature the ultimate Analog Devices SSM2017 mic/line preamp (on each channel). Other major console manufacturers sometimes offer this pre-amp as an upgrade option. In order to preserve the pristine signal quality of the S/L's preamp circuits throughout the signal path, Carvin utilized surface mount P.C. technology with expensive high quality surface mount components and high slew rate op-amps. The S/L's ingenious design features extremely large ground traces, double sided, plated-through FR4 military spec circuit boards and other design aspects normally found in only the most expensive world class consoles. As a result, the S/L series consoles offer some of the finest EIN, headroom, cross-talk and S/N specs of any console regardless of size or price! The S/L's 4-

band EQ section features two semi-parametric mid band and EQ in/out switching, as well as Lo-cut switching. The sweet sound of this EQ cannot be overstated. Carvin spent a great deal of time and effort to ensure that the delicate subtleties of the incoming signals are not disturbed by the EQ section, but that the character and flavor of the original signal be enhanced. The warmth & intensity of the new S/L consoles from Carvin are startling! Respective of any other console you may have been considering (for either live sound or recording), Carvin promises you will not find a better sounding console!

Signal Routing and Capabilities

Like any exceptional recording console, the S/L consoles offer advanced In-Line signal routing. The S/L consoles easily meet both pre and post production studio requirements, as well as offering additional ruggedness and special routing capabilities normally reserved for live sound mixing. If you are an experienced sound engineer, we invite you to review the signal routing and flexibility of this console for either application. Carvin is always eager to answer any questions you may have regarding the S/L console and how it will solve your specific needs. Even if you are just a novice, be assured the intelligent design and flexibility of the new S/L series consoles will meet all your sound needs and will allow you to discover aspects of your mixes you may not have even considered.

The channel format of the S/L consoles offers some unique features and capabilities.

Just a few (for your consideration) are as follows:

1) Phase Switch (per channel): Most mixes neglect the very important aspect of proper phase. When utilizing multiple microphones and signal sources, it is imperative that the final mixed signal be properly "in-phase". The sound in the room and on each microphone may be dynamic and lively, but when the final mix is accomplished the sound can become dull and lacking brilliance. The close proximity of multiple microphones recording the same source can result in a mix that sounds muddy or boxey. The channel phase switch on each mic channel ensures that your mixes will remain as brilliant and lively as the original instrument.

2) Channel Tape Switch - This function allows you to quickly configure your console for multi-track mixdown. By pressing the TAPE switch, your multi-track tape returns signal flow through the normal channel signal path for EQ, effects and basic post production mixdown.

3) Tape Selection for Buses 5 & 6 and 7 & 8 (depending upon the selection of the Input Tape Switch) allows interconnecting either the tape input signal or the mic input to these buses. Depending upon the configuration of your console, this switch will allow for individual bus routing for quick overdubbing while simultaneously setting up the stereo control room mix. For most recording situations, the input tape switch will be routing the mic signal through the channel and then (from the direct output) to the multi-track tape deck. The output of the multi-track tape deck is then fed back into the tape inputs of the channel and the tape switches will be depressed for buses 5&6 (for the studio headphone mixes) and for buses 7&8 for tape monitoring and stereo control room monitoring. In this configuration, punch-ins are a snap and a quick rough tape mix is automatic. The S/L consoles can accomplish just about any special signal routing requirement you may have. But, there is one aspect of the tape switch that is not normally considered. On the S/L console, you can use the channel tape input as a line input! You can then select the tape switch and route the signal to the stereo bus 7&8. From the 7&8 stereo bus output, you can select the L/R switch to route the signal back to the console's L/R master faders. This this feature will configure your 24 input console to a 48 input console (more if you have a 40 or 56 channel) with 24 mic pre's and 24 additional line -ins! Yet another fantastic and unique feature of the Carvin S/L series consoles!

4) Channel Solo & PFL - The S/L consoles are capable of both solo and PFL (Pre Fader Listen) monitoring. For live applications you may wish to configure your console for PFL (especially while setting up the system). But, you may decide that you prefer to have the function of the S/L's "solo-in-place" as you progress with your mix. Solo-in-place means that you can monitor the exact level as it is placed in the stereo field. The ability to quickly configure your console for either type of monitoring is both unique and yet another advanced feature of the S/L series consoles.

The above features are only a few of the unique options offered by the S/L console yet there are so many advanced capabilities such as:

Aux send switching to the XLR balanced outputs (for balanced signal sends especially useful for live sound reinforcement) Stereo Subgroup to L/R interconnects Control Room to Studio Output selection

Fully isolated power supply utilizing a heavy-duty Speakon[™] cable

We simply cannot explain all the configurations and capabilities of these extremely versatile consoles in this short introduction. We encourage you to read the manual and if you have questions call and allow our product experts help direct you to the advanced features available with the S/L series. Without a doubt Carvin's new S/L consoles represent a significant advancement in home studio as well as live sound mixing. The fact that they are value priced is just another bonus you receive from your new Carvin S/L series console.



RECEIVING INSPECTION—read before getting started

INSPECT YOUR MIXER FOR ANY DAMAGE which may have occurred during shipping. If any damage is found, please notify the shipping company and CARVIN immediately.

SAVE THE CARTON & ALL PACKING MATERIALS. In the event you have to re-ship your unit, the S/L console will only be accepted in our factory designed shipping carton or damage could result in which CARVIN and the shipping company are not responsible for.

SAVE YOUR INVOICE. It will be required for warranty service if needed in the future.

SHIPMENT SHORTAGE. If you find items missing, they may have been shipped separately. Please allow several days for the rest of your order to arrive before inquiring.

RECORD THE SERIAL NUMBER on the enclosed warranty card and on this manual for your records. Keep your portion of the card and return the portion with your name and comments to us.

For the new owner

Congratulations on your selection of a Carvin product "The Professional's Choice." Your S/L series console demonstrates Carvin's commitment to producing the highest quality & most sophisticated engineering in the audio industry today. Its wide acceptance and use by industry professionals illustrates the basis for Carvin's recognition as "The Professional's Choice."

Professionalism can only be measured from the results people achieve through their efforts and knowledge. It is not something that automatically happens when buying new or more sophisticated equipment. Rather, it's what you do with the equipment and how well you do it that ultimately makes the point. We are certain your new CARVIN console will deliver the performance necessary for you to achieve solid results, and ultimately enjoy a high degree of professionalism.

To compliment your new console and help you acquire that knowledge, we've included this manual. All of the information you need to be up and running is right here! You'll find using this manual easy and convenient. We've gone to great lengths to make it so. We've attempted to present the technical aspects of your new console accurately and in "plain English". But, if you have any questions that are not answered here, please call us at our toll free numbers. Our sales staff is well versed in the technical aspects of our products and are waiting to assist you with any questions you may have. We sincerely wish to ensure your complete satisfaction and enjoyment with your new console.

If you would like to comment on features or performance of your new console, please feel free to contact us. Comments from our customers have always helped us improve and further develop our products. We sincerely welcome any comments or ideas you may have.

Please, send in the warranty card. Although it is not absolutely necessary to ensure warranty protection, it will allow us to better know how you are using our equipment while keeping a ready reference for our files. Sending in the warranty card also helps us to mail literature and information that may be of interest to you as a professional musician. Let us know where you are so we can keep in touch!

In this manual there are diagrams and descriptions to aid you in understanding your new console. So, with this manual in hand, you hold the key to proper operation and the ability to achieve professional results.

May you enjoy many years of success and fun with your new CARVIN console!

TOLL FREE (800)-854-2235



For your records, you may wish to record the following information.

Serial No._____ Invoice Date_

Quick set up - Before power is turned on.

1) Be sure the SLP5600 power supply and AC cord is correct for your region (120 Volt or 230 volt).

2) Use only a grounded (3 prong) power outlet to prevent a shock hazard. This also provides the quietest grounding for your mixer.

3) Zeroing the console knobs and switches insures the current status of your mixer and reduces surprises when initially plugging in instruments and microphones.

Zeroing procedure:

- Channel Turn all EQ gain controls to their center positions. Turn the mic-pre gain controls to their full counter-clockwise position. Turn all auxiliary send controls counter-clockwise to "0". Place channel faders to the down position. Place all channel switches to the up position.
 Master - Place all master and group faders to the down position. Turn all control room levels counter clockwise to "0".
 - Turn all control room levels counter clockwise to "0" Turn all Auxiliary send master levels counter clockwise to "0". Turn all stereo return levels counter clockwise to "0".

4) The SL mixer series uses an external power supply, the SLP5600. AC power is connected to the SLP5600 and is in turn connected to the mixer via a 4 wire Speakon[™] cable. If an alternate Speakon[™] cable is to be used, be sure it is a 4 wire cable. Not all Speakon[™] cables have 4 wires.

5) After connecting your power supply and zeroing the console, turn on the mixer via the red power switch on the SLP5600 power supply. Test the mixer by plugging a microphone into a channel XLR input, turn up the gain control at the top of the channel and turn up the channels fader. While talking into the microphone, the channel meters should be showing signal. Some adjustment to the gain control and fader may be needed to see the signal on the channel meters. Now assign the channel to any of the master groups or Left-Right via the channel assignment switches 1-2, 3-4, 5-6, 7-8, or L-R. After adjusting the corresponding master fader, the master meters should be showing signal.

This is a very simple turn on/setup procedure. However, be sure to read the entire manual. Note; some features or setup connections may help in your situation because most of the S/L console features are not clear cut Studio or Live functions.



TABLE OF CONTENTS

Introduction
1. Mixer Overview
2. Installation
3. Input Channel Section
4. Master Sectionpg 11 Master I/O section Left, Right, and Groups Control Masters Stereo Returns & Tape IN/OUT Auxiliary Send Masters
5. Standard Studio Mixer Setupspg 14
6. Standard Live Mixer Setupspg 17
7. Using the Inserts
8. Meter Bridge
9. Power Supply
10. Cable Connections
11. Block Diagrampg 26
12. General Specifications
Appendixes
B. Schematics
C. Console Dimensions
D. Glossary of Terms

1. Mixer Overview

The S/L series mixing consoles are designed for Studio and Live performances. S/L series consoles allow the user to perform multi-track recording where the tracks equal the number of channels for the mixer; either 24, 40, or 56. This ability is due to the Dual In-line channel format used on the S/L series. Dual In-line channels provide a mixing path for the microphones or other input devices to be recorded, a mixing path for the individual channel outputs to a recording device's inputs and finally a mixing path for the output of the recording device. Live performances benefit from the S/L's AUX sends that can be used for monitor mixes or effects processing.

The SL consoles are equipped with :

- 24, 40, 56 channel frames.
- Low noise mic-pre amps EIN=128dB min.
- 4 band, dual mid sweep, Channel EQ.
- 8 auxiliary sends: 2 pre, 2 pre/post, 2 post, (stereo) pre/post fader.
- 4 stereo returns w/ Aux 1 & 2 sends, pan and level.
- 8 master groups w/ L-R, assignment and pan, mute and solo.
- Comprehensive master control room section with separate studio controls.
- Specially designed high current headphone outputs.

2. Installations

The S/L series console dimensions are in Appendix B for table size and custom installments. A good installation will keep in mind placement proximity to potential noise sources and positioning the mixer where the sound can be heard clearly from the audience's perspective.

The outboard SLP5600 AC power supply is also a consideration when installing a console of this caliber. The AC power for the mixer and the rest of the sound system needs to first of all, have the capacity to power the system. Be sure power amps and other high power portions of the system are on large enough and, if possible, separate breakers. It's possible to lose one or two power amps in a large system and deal with a 3-6 dB drop in output but if the power amp takes the mixing console with it, you've lost everything. Also, with AC power try to avoid using the same power outlets used by stage lighting. The performance lights typically require large amounts of power and produce heavy AC power spikes when switched on and off. These lights and many banquet room lights can be on solid state dimmer controllers, which produce noisy spikes on their AC power lines. These noises can only cause headaches for a sound system and nightmares for a recording studios because some of the spikes are emitted and can't be taken out by power conditioners.

The placement of the mixing console in live sound situations is often misunderstood. Especially when a promoter or banquet director sees the console as taking up valuable floor space or as an eye sore at a fancy show. The placement of the mixing console is very important in hearing the performance from the perspective of the listeners. The placement for routing cables is fairly obvious, avoid large power transformers, air conditioners and lighting rigs! Note: the best sound location in a room will be near the console because that's where the mix is created.

3. Input Channel section

1. 48V PHANTOM POWER SWITCH

This switch provides phantom power for condenser mics in groups of 8 channels. This allows other channel groups to remain non-powered for instruments and dynamic mics that don't require phantom power. The 48V switch should not be engaged when using sources that do not use phantom power.

2. CHANNEL DIRECT OUT

The DIRECT OUT is a 1/4" TRS (Tip/Ring/Sleeve) balanced output. The DIRECT OUT provides an output of the channel (post fader) unaffected by the pan or assignments. This is typically used for multi-track recording where each channel feeds a track on the recorder. The DIRECT OUT can also be used for sending a channel to an effects unit and to return the effects unit to another channel for mixing.

3. CHANNEL TAPE IN

The TAPE IN is a 1/4" TRS (Tip/Ring/Sleeve) balanced input. The TAPE IN input jack makes up the alternate input path in the dual in-line format of the S/L series consoles. This is typically used as a return from a multi-track recorder. The TAPE IN signal can be accessed via the tape switch, through the Aux 7-8 tape switch or through the Aux 5-6 tape switch.

4. LINE IN

The LINE IN is a 1/4" TRS (Tip/Ring/Sleeve) balanced input. The LINE IN connectors are for connecting balanced and unbalanced instruments, mics and line level sources such as drum machines or keyboards.

5. MIC IN

The MIC IN is an XLR balanced input. These Mic inputs are for connecting professional low impedance microphones. +48V Phantom power is provided via the grouped phantom power switch for condenser mics requiring phantom power.

6. INSERT

The INSERT is a 1/4" TRS (Tip/Ring/Sleeve) Tip send and a ring return break point. The INSERT jack is typically used for inserting an effects unit, compressor and/or equalizer at the beginning of the channel's signal path. In the S/L series, the insert point is located post mic pre-amp and tape in (depending on the tape switch position) and before the EQ and AUX sends. With an insert cable (1/4" TRS on one end and dual 1/4" TS "Tip/Sleeve" on the other ends), the TRS end is used at the mixer's insert jack and the corresponding tip TS (Tip/Sleeve) plug goes to the units input and the corresponding TS (Tip/Sleeve) plug goes in the units output. To use the INSERT jack as a direct output for an alternate multi-track recording output, use a 1/4" TS (Tip/Sleeve) cable and insert to the first "click" (1/2 insert). Then connect the other end of the cable to an input on a multi-track recorder. For detailed information on channel insert set up, please see section 7 **Using the Insert Jacks** on Pg 22



3. Input Channel section continued.

7. GAIN CONTROL

The GAIN control adjusts the gain of the channels high gain microphone pre-amplifier. This pre-amplifier is designed for professional low impedance microphones. Monitor the gain level with the PEAK LED. If the LED is constantly lit, decrease the amount of gain. If distortion is present, decrease your gain until it is eliminated. The GAIN control is also the level control for the LINE IN. Using both MIC IN and LINE IN inputs at the same time is possible with some deterioration in common mode rejection. To set the gain for optimum signal-to-noise, audition your audio source and turn the GAIN control up until the PEAK LED starts to flash. The PEAK LED should not be constantly lit but occasional peaking is OK. Repeat this with each channel. Different mics will require different levels of GAIN.

8. PHASE SWITCH

The PHASE switch reverses the polarity of the input signal to compensate for phase differences due to microphone placement or incorrect wiring of input cables. This switch should be released (up position) for normal operation.

9. TAPE SWITCH

The TAPE switch provides the Dual In-line channel format. It should be viewed as an input tape switch. In the up position, the MIC and LINE inputs are feeding the channel's EQ, sends and fader with assignments. The TAPE IN goes to the TAPE 5-6 and TAPE (AUX 7-8) switches. In the down position, the routing is "flipped" where the TAPE IN has the EQ, sends and fader with assignments and the mic line inputs are routed to the TAPE 5-6 and TAPE(AUX 7-8) switches. In a recording situation, the switch up position provides the tracking mode and the down position provides the mix down mode.

10. LOW CUT SWITCH

The LOW CUT switch is a 12dB/Octave high pass filter with a corner frequency of 75Hz. Using the LOW CUT feature helps eliminate unwanted low frequencies. Use the LOW CUT filter to reduce "rumbling" and "boom" noises picked up from mic stands or hollow body acoustic/electric guitars. This filter can also help tighten bass response by turning up the LOW EQ control while using the LOW CUT switch.

11. EQ HIGH CONTROL

The High control is an active shelving type of tone control with a corner frequency of 12KHz. The \pm 15 dB boost and cut provides an overall 30 dB range of powerful EQ control.

12. EQ HI MID LEVEL AND FREQUENCY CONTROLS

The HI MID level and FREQ controls comprise a semi-parametric or sweepable mid equalizer. The level control provides ±15 dB boost and cut with the middle position (center detent) at OdB. The frequency being manipulated is controlled by the frequency control. The FREQ control sets the center frequency for this "bell curve" designed tone control. The HI MID FREQ control ranges from 500Hz to 5KHz. Cutting and boosting frequencies improperly can cause an "un-natural" sound to your vocals or instruments. Only use these controls to overcome the limitations of the Microphones and room effects to achieve a natural sound for the entire performance or recording.

13. EQ LOW MID LEVEL AND FREQUENCY CONTROLS

The LOW MID level and FREQ controls comprise a semi-parametric or sweepable mid equalizer. The level control provides ± 15 dB boost and cut with the middle position (center detent) at OdB. The frequency being manipulated is controlled by the FREQ control. The FREQ control sets the center frequency for this for this "bell curve" designed tone control. The LOW MID FREQ control ranges from 80Hz to 2KHz. Cutting and boosting frequencies improperly can cause an "unnatural" sound to your vocals or instruments. Only use these controls to overcome the limitations of the microphones and room effects to achieve a natural sound for the entire performance or recording.

14. EQ LOW

The LOW control is an active shelving type of tone control with a corner frequency of 80Hz. The ± 15 dB boost and cut provides an overall 30 dB range of powerful deep bass EQ control.

15. EQ IN SWITCH

In the down position, the EQ IN SWITCH routes the audio through the channel's EQ (High, both Mid's, and the low controls). In the up position, the EQ IN SWITCH bypasses the EQ section but leaves the LOW CUT switch active in the circuit. The EQ IN SWITCH is great for comparing the "flat" to EQ sounds.

16. CHANNEL AUX 1 & AUX 2 PRE SENDS

The AUX 1 & AUX 2 controls are pre EQ, pre mute and pre fader sends. These controls are designed as uninterrupted monitoring feeds for stage monitor or in studio tracking. The AUX 1- AUX 2 PRE signal is post INSERT in order to get the benefits of an inserted channel effect such as a compressor.

17. AUX 3 - 4 PRE SWITCH

The Aux 3 - 4 PRE switch in the up position places AUX 3 and AUX 4 send controls post EQ and post fader (working the same as AUX 5 and AUX 6). In the down position AUX 3 and AUX 4 send controls, are pre EQ, pre mute and pre fader (working the same as AUX 1 and AUX 2).



18. CHANNEL AUX 3 AND AUX 4 PRE OR POST SENDS

The AUX 3 and AUX 4 level controls (with the Aux 3 - 4 PRE switch in) provides a total of 4 PRE monitor controls when used with AUX 1 and AUX 2 (4 monitor mixes). With the Aux 3 - 4 PRE switch out, AUX 3 and AUX 4 can be used with AUX 5 and AUX 6 as effects sends for a total of 4 post sends.

19. TAPE 5 - 6 SWITCH

The TAPE 5 - 6 switch is linked to the TAPE switch at the top of the channel. Here the alternate input (the input not going to the fader) can be sourced to AUX 5 and AUX 6. This is used primarily in recording during tracking to provide playback to the studio musicians.

20. CHANNEL AUX 5 AND AUX 6 POST SENDS

The AUX 5 and AUX 6 sends are post EQ and post Fader sends when the TAPE 5-6 switch is up. The AUX 5 and AUX 6 sends are designed for effects sends or any other fader tracking sends needed. See Tape 5-6 switch for it's tape input effects on AUX 5 and AUX 6.

21. CHANNEL AUX 7/8 TAPE SWITCH

The 7/8 TAPE switch is linked to the TAPE switch at the top of the channel. Here the alternate input (the input not going to the fader) can be sourced to the stereo pair of AUX 7 and AUX 8. This is used primarily in recording during tracking to provide playback mixing for the control room and to the studio musicians.

22. CHANNEL AUX 7/8 PRE FAD W/ EQ SWITCH

The PRE FAD w/ EQ switch in the up position places the stereo pair of AUX 7 and AUX 8 controls post EQ and post fader for use as effects sends. In the down position, the stereo pair of AUX 7 and AUX 8 controls are post EQ, pre mute and pre fader for stereo in-ear monitor sends or a stereo drum monitor send. This switch only works when the AUX 7/8 TAPE switch is in the up position.

23. CHANNEL AUX 7-8 PRE OR POST STEREO SENDS

The stereo pair of AUX 7 and AUX 8 level controls are post EQ and post fader sends when the TAPE and PRE FAD w/ EQ switches are up. The stereo pair is made up of the 7/8 level control and the 7/8 pan control which pans between the two send AUX 7 and AUX 8. The stereo pair of AUX 7 and AUX 8 sends are designed for effects sends or any other fader tracking send needed. See TAPE switch for its tape input effects on AUX 7 and AUX 8.

24. PAN

This is the channel's main PAN control. It allows the channel's signal to be place anywhere in the stereo field from left to right with the center detent being centered. The PAN control in the studio is typically used for adjusting the stereo field for each channel, but it is also used for assigning channels to individual buses. The later is typical for live mixing where certain channels are grouped to a single bus for sub group mixing. See **Standard Live** mixer set-ups for more details on sub mixing.

25. PEAK, SOLO, AND MUTE INDICATORS

The RED PEAK LED indicator is pre-fader and post EQ. A constantly lit LED indicates the signal probably needs a reduction in the GAIN control to prevent input overloading. The GREEN SOLO LED indicates the solo switch has been pressed on the channel. The RED MUTE LED indicates the MUTE switch has been pressed on the channel.

26. MUTE SWITCH

The MUTE switch turns off the channel at the fader. All sends which are pre fader will not be muted. This is extremely useful when you need to mute channels but can't afford to lose fader settings.

27. SOLO SWITCH

The SOLO switch allows the operator to monitor each channel. The SOLO switch can be set for either "PFL mode" (Pre Fader Listen) or "SOLO after fader" where the channel's pan position is heard when soloed. The switch to change the SOLO mode is found in the CONTROL MASTERS section of the console. See **Master Section** for more information.

28. ASSIGNMENT SWITCHES

These switches assign the channels' signal to the group faders (8 buses) for sub-mixing and to the Left & Right faders in the Master Section. Each channel can be assigned to the 1-2, 3-4, 5-6, 7-8, and L-R Faders in stereo pairs. This feature allows the operator to group certain channels (such as the channels used to mic an entire drum kit) and assign them to one pair of the master faders in the 1-2, 3-4 sub group. This sub-mixing feature decreases the number of channel faders that need to be adjusted. See Standard Live mixer setups for more details on sub mixing.

29. CHANNEL FADER

The channel FADER control is the final level control of the channel before the PAN control. The Fader has a 0dB marker for nominal setting of the channel. For typical mixing the 0dB position provides the lowest noise and highest headroom for the channel. The Fader has an added 12dB of gain for signals that need an extra boost in the mix, however pushing the fader above the 0db marker should be done in moderation. The channels have plenty of headroom but the proper adjustment should be made at the channel's input or GAIN control. Be aware of the signals and fader settings so you don't over-drive the bus or degrading the channel's signal by over-driving the faders. The meter bridge and direct outputs are sourced after the fader.



4. Master Section

Master I/O Section

Note: All XLR connectors follow the Standard of: 1- Ground 2- Non-inverting (hot) 3- Inverting (cold) All master 1/4" jacks are unbalanced TS (Tip/Sleeve) with the exception of the control room and Studio Right jacks used for PHONES.

1. POWER CONNECTOR

The SpeakonTM power connector is used to bring the DC power into the console from the 2 rack space SLP5600 power supply. A standard 4 wire SpeakonTM cable is required.

2. INSERTS LEFT-RIGHT, GROUPS 1-8

The Master Inserts are 1/4" TRS pre fader inserts. The TRS is the same as the channel inserts with Tip Send-Ring Return. These inserts are perfect when only a limited number of compressors are available. The Inserts are especially good for unbalanced graphic EQ's and other unbalanced gear desired on the outputs because it will not effect the XLR balanced outputs.

3. GROUP 1-8 XLR outputs

The Group 1-8 XLR outputs are electronically balanced connectors for the 1-8 Groups. The Group 1-4 XLR's are also used by AUX 1-4 via the AUX 1-4 FLIP switches.

4. LEFT and RIGHT XLR and 1/4" CONNECTORS

These XLR connectors are balanced outputs of the Right and Left master outputs. The 1/4" connectors are unbalanced outputs of the same L-R signals.

5. TAPE IN and OUT RCA's

Stereo RCA jacks are provided for connecting a tape recorder or CD player. The levels of these jacks are controlled by the corresponding TAPE IN and TAPE OUT controls in the master section.

6. AUX SEND 1/4" OUTPUTS

Eight 1/4" unbalanced output connectors correspond to the AUX SENDS controls in the master section. Aux 1-4 outputs are also used by the Group 1-4 masters via the AUX 1-4 Flip switches.

7. STEREO AUX RETURNS 1/4" inputs

Four 1/4" unbalanced stereo pairs (8 inputs) correspond to the 4 STEREO Return controls in the master section. If only one side (right or left) is used, the jacks are wired so the signal will be supplied to both the right and left path of the stereo return. This allows all the stereo features to work properly with a mono return.

8. CONTROL ROOM 1/4" OUTPUTS (CONTROL L-R, PHONES)

The 1/4" unbalance control room outputs provide individual left and right out for the control room section of the console. The right side is also configured as a TRS L/R stereo connection for headphones, when nothing is plugged into the left side. The PHONE connection is ideally suited for live situations. The individual outputs provide an ideal studio monitoring system connection. Both outputs are designed with discrete high current output circuitry for driving headphones and performance studio amplifiers.

9. STUDIO 1/4" OUTPUTS (STUDIO L-R, PHONES)

The 1/4" unbalance STUDIO outputs provide individual left and right out for the STUDIO section of the master. The right side is also configured as a TRS L/R stereo connection for headphones when nothing is plugged into the left side. The PHONE connection is ideally suited for live situations. The individual outputs provide an ideal studio monitoring system connection. Both outputs are designed with discrete high current output circuitry for driving headphones and performance studio amplifiers.

Left, Right and Groups

10. POWER LED Verifies power is being supplied to the mixer and the mixer is on.

11. LEFT and RIGHT FADERS

The LEFT and RIGHT FADERS get their signal only from the channel L-R assignment switches, Stereo Returns and the Group 1-8 L-R assignment switches. These faders are post L-R master INSERTS and provide the level adjustments for the left and right XLR and 1/4" outputs.

12. GROUP FADERS

Each GROUP FADER gets its signal from the corresponding channel group assignment switch. These faders are post Group master INSERTS, and provide the level adjustments for the corresponding Group XLR output.

Control Masters

13. GROUP MUTE'S

The Group MUTE switch mutes the Group signal pre Fader and post insert. This means the group output and L-R assignment are muted, but not the send on the insert. The corresponding RED Mute LED indicates the Group has been muted.

14. GROUP SOLO

The GROUP SOLO is an in place SOLO located after of the group PAN circuit.

15. GROUP L-R AND PAN

The Group L-R Assignment switch is located after the PAN circuit allowing the group output to be assigned to either the left, the right, or any PAN setting of both masters.

16. GROUP PEAK LED'S

The Group PEAK LED indicates the group is within 3db of clipping. The PEAK LED is located pre fader post insert in order to indicate the peaking level of the summing bus or insert return when the group insert is used.

17. SOLO LEVEL

The SOLO level controls the volume of the SOLO bus heard through the control room (and channel PFL when used). The SOLO level also affects the measured level seen on the L-R meters when something is soloed. This is why the SOLO is center detented to provide a unity gain location for the solo circuit in order to indicate the desired soloed material correctly on the meters. For normal solos the meters indicate stereo in place material, when using the channel PFL mode the meters indicate in mono the pre fader material on both meters.

18. SOLO LED

The SOLO led lights up when one or more solo switches have been pressed. The SOLO LED also indicates the control room and master L-R meters are monitoring the pressed SOLO switches.

19. CHANNEL LISTEN MODE & PFL MODE INDICATOR

The channel listen mode switch provides the option of changing the channel SOLO switches from normally post PAN, stereo in-place solo switches to pre fader PFL switches used in live applications. The PFL MODE led indicates the channels are in PFL mode.

CONTROL MASTERS

solo (18)

O SOLO

20. CONTROL ROOM LEVEL

The CONTROL ROOM level adjusts the level of CONTROL ROOM outputs. This is the master level when using the control room output for studio monitors or when using the control room right output for PHONES.

21. CONTROL ROOM SELECTION SWITCHES

Use these switches to select what is heard through the control room outputs. When selected these switches will be heard at the outputs. The switches perform as follows:

- AUX 7-8: listens to the final output of the AUX 7-8 masters.
- TAPE IN: listens to the post TAPE IN level input.
- AUX 1-2: listens to the final output of the AUX 1-2 masters.
 - MONO: creates a control room left/right summed mono output on the control room outputs.
 - L-R: listens to the final output LEFT and RIGHT masters.

TO STUDIO: Inputs the control room output to the Studio output.

22. STUDIO LEVEL

The STUDIO level adjusts the level of STUDIO outputs. This is the master level when using the STUDIO outputs for a studio playback system or when using the STUDIO right output for PHONES.

23. STUDIO SELECTION SWITCHES

Use these switches to select what is heard through the STUDIO outputs. When selected these switches will add together at the outputs. The switches perform as follows: (Also see TO STUDIO above under the control room selection switches.)

- AUX 7-8: listens to the final output of the AUX 7-8 masters.
- AUX 1-2: listens to the final output of the AUX 1-2 masters.
 - L-R: listens to the final output LEFT and RIGHT masters.



Note: You can use both the Control Room outputs and the Studio outputs for headphone monitoring.

STUDIO

(22)

Auxiliary Send Masters

24. AUX SENDS 1-6 CONTROLS & PFL

These are the Master levels for Auxiliary Sends. Each has its own PFL switch to monitor the AUX send buses pre fader.

25. AUX 1-4 FLIP SWITCHES

The AUXiliary FLIP switches on AUX send 1-4 are for the purpose of providing XLR balanced outputs for these auxiliaries. This is especially useful for balanced monitor sends. When the switch is down, the output of the auxiliary is sent to

the corresponding group XLR output and meter bridge feed. At the same time, the affected group output is fed to the auxiliaries 1/4" output jack. No other operation is affected.

26. AUX SENDS 7/8 CONTROL & L-R SWITCH

The AUX SENDS 7/8 CONTROL is a stereo master level for both the AUX Send 7 and 8. The output of AUX 7-8 level can be sent to the L-R masters via the L-R switch providing a route for secondary channel inputs to get to the main L-R mix.

Stereo Returns & Tape IN/OUT

27. STEREO RETURN LEVEL AND PAN

The stereo return level is the main level control for the return. The stereo return PAN works as a balance control for Stereo inputs and as a PAN control for mono inputs.

28. STEREO RETURN AUX 1 AND AUX 2 SENDS

The Stereo return AUX 1 and AUX 2 send controls allow the returns to be fed to the monitor mixes. These Aux sends are created from summing the left and right pre-level return signals. Because they are pre-level, they can be controlled independently of the signal going to the main left right mix.

29. STEREO RETURN MUTE AND SOLO

Each stereo return contains a MUTE and a SOLO switch. The MUTE is equivalent to a L-R assignment on the channels, except it works in the reverse action. The MUTE switch is post the PAN and LEVEL controls. The SOLO switch is an in-place stereo solo, placed in the same location as the MUTE (post PAN and LEVEL).

30. TAPE OUT LEVEL AND SOURCE SWITCH.

The TAPE OUT level provides an on board control for the RCA tape output. The level is limited to adding a maximum of +6db to the original signal going out in order to prevent overdriving the recording equipment used. The L-R or AUX 7-8 source switch is provided to allow the recording of a live performance from the main L-R house mix or an independent mix through the AUX 7-8 sends.

31. TAPE IN LEVEL

This adjusts the level of the signal coming from RCA TAPE IN jacks.

32. TAPE IN L-R AND AUX 1-2 SWITCHES

These switches allow playback of the RCA TAPE IN signal through the main L-R mix and/or the AUX 1-2 monitor mixes.





5. Standard Studio Mixer Setups

The following section describes two of the more common Studio recording setups. These setups are not the only setups you can use, many of the connections described here can be combined and customized to provide the best setup for you. The first Setup is based on the In-Line recording method and the second setup is based on the Split recording method.



In-Line Recording Setup.

The S/L series consoles are designed for In-line

recording. In-line recording is where each mixer channel can handle a mic input, a direct output to a multi-track and a tape return input from the multi-track. This is due to the dual in-line channel format of the S/L series console. The dual In-line channel is made up of two individual channel signal paths, a minor path for monitoring and a main, full featured path for tracking and final mix down. These two signal paths are sourced by two inputs at the MIC input (including LINE IN) and the TAPE IN. With the use of the TAPE switch, either of the two input paths can flow through either of the two signal paths.

When the TAPE switch is OUT, the MIC/LINE IN goes through the main signal path providing access to: the channel insert, the EQ section, all the Auxiliaries, the Fader, the DIRECT OUTPUT and Bus assignments. The tape input goes through the minor path, which can be



ments to the multi-track unit's individual tracks), where a mic is plugged into the channel's XLR

MIC input. Then via the channel DIRECT OUTPUT, the mic is recorded on one track of the recording unit. At the same time on the channel, the output of the recording unit can be monitored via the tape input and the Aux 7-8 TAPE switch.

When the TAPE switch is IN the inputs are flipped and the tape input goes through the main path



and the MIC goes through the minor path. This is the typical setup for mix down (where all the recorded tracks are mixed to a

stereo mix). The MIC input can be used as an input in the mix down to add more effects returns and/or midi tracks to the stereo mix. Figure 5-1 shows a block diagram of the channel and the modes of the TAPE switch. When the TAPE switch is fully understood, it can become a very powerful recording tool.

Figure 5-2 shows channel input connections for the In-line format. Here the mic and line inputs are both used in the example, but typically one or the other would be used. In the example the multi-track recorder is showing the inputs and outputs of one track. The rest of the tracks would be setup on the other channels in the same manner. One of the key points here is that these connections can be left connected in both the tracking and mix down modes because the TAPE switches re-patch the inputs for the two modes internally.



Split Recording Setup.

The Split recording method has many different looks depending on the mixing consoles setup and features. The basic concept of split recording is simply using a different channel or bus for mixing than was used for tracking. The complication comes with how the features of the mixer are used. If the mixer has direct outputs, then the channel used for tracking goes direct to the recorder's inputs and another channel is used for the recorders playback through the mixer. If the channel does not have direct outputs, then the channel output goes through the busses to the recorders inputs. If there are more tracks and inputs than channels, then the channels have to be patched for mic and instruments to track. For playback and mixing, the channels are patched for tape outputs.

With S/L consoles, some split recording is done when there are less tracks on the recorder than on the console or multiple channels are desired to be recorded to one or two tracks. In these cases, the Groups are used to sub-mix banks of channels down to one or two tracks. The same In-line setup is used, except that the direct outputs going to the desired tracks to get the sub mixed signals are patched to the outputs of the groups being used for sub mixing the channels. Even in this setup patching is greatly reduced with the S/L console because the tape returns can remain in the channels and only 8 direct outs and 8 recorder inputs need to be going through a patch bay for patching to the group outputs. The best recording setup has the least possible patching and interconnection.

Figure 5-3 shows a simple combination of in-line and split recording setups using a patch bay like the Carvin PB48 patch bay . The setup shows a typical 16 channel recording system. The first 8 channels are connected with the in-line method; where the mixer's TAPE IN's are receiving the multi-tracks 1-8 outputs and the mixers DIRECT OUT's are going directly to the multi-tracks inputs 1-8. Tracks 9-16 are setup using a patch bay with the normal connection (with nothing plugged in the front) producing the same setup as with tracks 1-8. Using the patch bay allows the busses to be used instead of the direct out's for inputs to the multi-track recorder. This allows multiple channels to be mixed down to one or two busses using less tracks. With the patch bay, compressors, gates or effects can also be patched (optional to using the insert jacks) covering multiple channels with one unit. The final channel setup shown in *Figure 5-3* shows the same 16 channels with the inputs from the mics and instruments being recorded.

see figure 5-3 on page 16



6. Standard live mixer setups

The following section describes aspects of using the S/L console in live sound reinforcement. Because live sound reinforcement is so diverse in set-up requirements and connections, this section will only discuss aspects of a live system setup and focus on the features of the S/L series console. The features covered in the following are:

Auxiliary sends Recording a live show to 2 track Sub mixing with groups A Live sound system The Sound Check

AUXILIARY SENDS

The S/L consoles feature 8 auxiliary sends with several switchable routing possibilities. For live sound, the main tasks for the auxiliary sends are stage monitor sends, effects sends and specialty sub mixes.

For stage monitor sends, the S/L console is capable of having 6 pre-fader sends. This is made up of the pre-fader AUX 1 and 2 sends, the AUX 3 and 4 sends in pre fader mode (via the PRE switch down) and the AUX 7/8 sends in the pre-fader / post EQ mode (via the PRE w/EQ switch down). In many cases, the AUX 1-2 sends and AUX 3-4 sends in the pre fader mode will provide the typical 4 monitor sends. Here the 4 monitors would break down as 3 individual front-of-stage monitors (center, left and right) and a monitor mix for the drummer (rear stage). The Auxiliary 1-4 sends are especially designed for this setup with the PRE switches on the channel and the FLIP switches on the AUX SEND 1-4 masters. The FLIP switches, on the AUX SEND 1-4 masters make the 1-4 Group's XLR balanced outputs available to these sends. The XLR balanced outputs provide better noise



4 MONITOR MIXES USING AUX 1-4

rejection and better Signal level when using long cable runs, like with XLR "snakes" going from the mixer to the stage. In turn for using these XLR's for the SENDs, the Group 1-4 outputs are now available at the 1-4 send 1/4" outputs.

For effects sends, S/L consoles can provide 6 post fader effects sends. These are made up of the AUX 3 and 4 sends in post fader mode (via the PRE switch up), AUX 5 and 6 post fader sends and AUX 7/8 send in the post fader mode (via the PRE w/EQ switch up). In most systems with 4 monitor mixes, AUX 5 and 6 will provide the typical 2 effects sends with AUX 7/8 open for two more effects sends.

For specialty sub mixes, the TAPE inputs can provide additional inputs for live shows when channeled through the AUX 7/8 send (via the AUX 7/8 TAPE switch in the down position) and in the master section by the AUX 7/8 L-R switch. These inputs, in this configuration have no EQ, so it is best these input signals are used with the stereo returns.

RECORDING A LIVE SHOW TO 2 TRACK WITH AUX 7/8

For specialty sub mixes, like recording a live show to a 2 track recorder, the S/L consoles feature a stereo send pair. The stereo send pair of AUX 7/8 provides (via the PRE w/ EQ switch down) an independent stereo mix with PAN for live recording. This independent post EQ send allows a completely separate mix to be sent to a 2-track recorder. Use either the AUX 7/8 Sends or the RCA TAPE OUT for the connection of a two track recorder. The TAPE OUT switch determins which outputs will receive the stereo mix. The AUX 7/8 PAN can also be extremely useful to send two discrete submixes to the L-R audio inputs of a video recorder. Panning hard left and hard right can separate the main speakers and other important channels from background music. This is useful in video post production.



SUB MIXING WITH THE GROUPS

For live sound reinforcement, the 8 Groups are mainly used for mixing sub groups of channels and providing alternate master outputs for surround sound mixes, center clusters and extension speakers systems. Sub mixing with the Groups is where several channels with a commonality are assigned on the channel to a Group (or Group pair for stereo). Then the Group is assigned to the L-R masters. The advantage is being able to adjust the group with one fader (or two faders for stereo). This would be used for a drum set using multiple mics. In sound check, the 4 individual microphones are adjusted for each drum, then adjusted as a group for leveling each drum for a total drum set sound. Later when the full band is playing, the drum set can be brought up and down with the one Group fader without disturbing the channel faders.

Also, sub-mixing can bring the advantage of one Group insert for control over several channels. This is where one compressor can be inserted to cover a full drum set reducing the number of compressors needed.

The Groups are also equipped to perform as master outputs. Here is where the Groups help support center clusters, extension speakers for reaching distant listeners and surround sound speakers for effects when the L-R masters are used for the Left and Right speakers.

A LIVE SOUND SYSTEM.

Figure 6-4 shows a typical live setup with the S/L console. All of the outboard gear is shown here in order to show its integration into the S/L console. The inputs are mostly mics, but because the console is distant from the stage, these mics and other inputs will be traveling through a "snake" cable. A snake is a cable carrying all connections to and from the stage to the console. Typically, a snake will only accept XLR balanced connections. This is why the Keyboard is shown connected through a "direct box" (Carvin DB3) which converts the unbalanced 1/4" outputs of the keyboard to balanced XLR outputs for the console. Often the "direct box" will also "ground" isolate the instrument from the console in order to reduce hum and ground noise.

The next section of *Figure 6-4* is the "front-of-house" rack usually containing compressors, effects units and graphic EQ's. These units are used at the inserts of the channels and at the master inserts for compressing vocals and drums, adding EQ or gates to background vocals and drums. Also, in the "front-of-house" rack will be the main graphic EQ's for room equalization on the house speakers. The 6 XLR outputs connected at the 1-4 XLR GROUP OUTPUTS and LEFT/RIGHT master outputs are for four stage monitor mixes via the AUX 1-4 FLIP switches. The LEFT and RIGHT master FADERS control the left and right main speaker clusters. The main mix may be in stereo or mono with the left and right control for room balancing. Typically a room's seating arrangement cannot be not setup properly for a true stereo performance. But having control over the two sides of the stage and even a separate feed for subwoofers can help the sound engineer control room acoustics and feedback. The four monitors are pre fader AUX 1-4 stage monitors going through the on-stage rack of graphic equalizes for feedback reduction. Often in simpler systems, the monitors do not get the attention needed to help the performers do their best on the stage. Whenever possible, test out the monitors and be sure the power level of the monitors is adequate for the expected stage level of the performance. A good monitor system makes the performers comfortable and therefore more focused on their best performance.

THE SOUND CHECK

The sound check takes some skill but mostly patience from the performers and especially you the system operator. If you get frustrated during the sound check, the performers can lose confidence and the sound may suffer due to things missed in the sound check. The basic sound check follows this format: First test all microphones and other input devices (direct boxes, etc.) before the performers are included in the sound check. A good thing to also check here is feedback in the monitors from the microphones. Good positioning of the monitors and the use of a graphic equalizer solves most major monitor feedback problems. Now for a sound check with the performers. First set the level of each performer individually and in cases where a performer has multiple microphones, such as with drummers, set each drum mic individually then the drum set as a whole. This is also a good time to make some channel tone control adjustments to tailor the sound of the individual performers and instruments. After setting each individual, have the performers run through a song. Don't hesitate to stop the performers if something needs to be heard solo again. Remember the sound check is not a rehearsal, but a system check. It is always a good idea for the mixer operator to have a microphone to inform the performers of what is needed during the sound check. If a monitor system is being used, the mixer operator's microphone should only be heard through the monitors when addressing the on stage performers, especially if something needs to be checked during the show. If the sound check is allowed to run through its full course, the system should run smoothly at show time.

Figure 6-4





7. Using the inserts

Full insert

The insert jack is a Tip/Ring/Sleeve (TRS) 1/4" phone jack where the tip is the send, the ring is the return and the sleeve is the ground. When used as an insert point or in full insert mode, the channel is opened up to allow an external piece of equipment to be inserted into the channels signal path. The channel signal coming from the TAPE switch, whether it is the MIC (up position) or the TAPE IN (down position), will be forced to go through the external equipment before it can continue back through the channel. The signal "reenters" before the low cut filter and continuing on to the channel EQ controls. Most external equipment is not set up for the TRS plug directly, so a special insert cable is required. The insert cable will have on one end the TRS (stereo) plug and two mono (TS) plugs (or jacks) on the other end. The two plugs each have the ground connected to the sleeve. One has the return on its tip and the other has the send on its tip. This allows the send to be connected to the input of the external equipment and the return to its output completing the insert loop back to the channel. One advantage of using the insert jack is with compressors. With a compressor at this point in the channel signal path, the input signal can be compressed reducing the need to adjust the faders during a show.



Half insert (insert Direct out)

The half insert connection creates a send signal without breaking the channels signal path. The insert in this mode is no longer used as an insert but it becomes what is called an "insert direct out". The advantage of an insert direct out is the signal at this point has only traveled through one amplifier stage resulting in the quietest and lowest distortion possible in the console. This is popular when recording tracks in the studio because the EQ and fader may not be needed and can be added in the mixdown. If an insert is needed on the same channel, special cabling is required to perform both functions. The half insertion connects the tip of the plug being inserted to the ring of the jack. (See the fig.) If the jack is fully inserted to where the tip of the plug connects to the tip of the jack, the internal jack switch will open and the channel's signal path will be broken. The connection will still function as a direct out but the channel's signal will stop at the insert and not continue on to the rest of the channel and the masters. The result of the half insert is multiple outputs for use in multi-track recording.



9. SLP5600 power supply



The SL series consoles use the SLP5600 external power supply. The SLP5600 power supply is a two space rack mountable power supply which provides the required DC voltages for the SL series.

Power Requirements

It is best to supply the SLP5600 with its own dedicated circuit. This reduces possible loss of power to the console if other circuit breakers are over loaded and tripped. The SLP5600 is supplied with a 6 AMP circuit breaker located on the rear panel. Should this breaker trip, turn off the power switch on the front panel and press in the black and white tab on the breaker. The circuit breaker is a thermal metal breaker, so it may not turn on immediately after tripping. Continue pressing every few seconds until the breaker latches. Now turn on the power again, if the breaker trips again, check for the proper cable connections.

Safety Ground

The SLP5600 is supplied with a third wire safety ground connected to the power cord. This should never be defeated. Along with providing the best grounding for the console, the safety ground prevents the possibility of electrical shock if a live wire comes into contact with the chassis. If grounds need to be lifted due to ground loop noises, only lift grounds on signal cables not on the AC power cords.

SPECIFICATIONS

120 VAC

DC OUTPUT

+48 Volt .5 Amps +18 Volt 3 Amps

- 18 Volt 3 Amps

Mixer Connection

The SLP5600 is supplied with a ten foot four conductor Speakon[™] cable. This can be replaced with any 16 ga. or higher 4 conductor Speakon[™] cable 25 ft or under. The Speakon[™] connector is a twist and lock system providing a secure connection at all times. AC INPUT

Fan Cooling

The SLP5600 is internally fan cooled. The fan cooling system automatically turns on when needed for cooling. On the larger consoles this fan system may remain on continuously for maximum cooling and uninterrupted operation.

Power Indication

There are three forms of indication that the power supply is on and functioning normally. First is the lighted power switch which lights when turned on. Second is the

green LED in the master section of the mixer labeled "power" which indicates the mixer is being supplied with the proper voltages. Third is the internal power supply LEDs which can be seen through the front panel vent holes. These LEDs are Red for $\pm 18V$, Green for $\pm 48V$ and Yellow for the fan on mode.

10. Cable connections used on the S/L series console

Balanced XLRs are used for the channel MIC inputs, the group outputs and the master left right outputs These all follow the standard: PIN 1 is ground PIN 2 is positive (in phase) PIN 3 is negative (out of phase or inverting)

Balanced 1/4" TRS phone jacks are used for the channel direct outputs, the channel tape inputs, and the channel line inputs.

These follow the standard of:

TIP is positive (in phase)

RING is negative (out of phase or inverting) SLEEVE is ground

Unbalanced 1/4" TS phone jacks are used for the master auxiliary send outputs, the left control room outputs, left studio output, the left and right return inputs and the master left and right 1/4" outputs. These follow the standard of: TIP is positive (in phase) SLEEVE is ground

1/4" TRS phone jacks are used for the channel inserts and the master/Group inserts.These follow the standard of:TIP is send (mixer output)RING is return (mixer input)SLEEVE is ground

1/4" TRS phone is used for the control room right output and the studio right output. If the left output is in use these right outputs will function normal as TS outputs like the left outputs. If the left has nothing plugged into the jack then these right jacks operate as headphones with: TIP is right RING is left SLEEVE is ground





11. Block Diagram

SL BLOCK DIAGRAM

All specs measured on (56 Channels Frame) 20-20kHz, Rs=150 Ω , RL=600 Ω . unless otherwise noted.

Total Harmonic Distortion

Channel Fader @ nominal Master Output = +14Dbu

Frequency	20 - 20kHz	@ 1kHz
+10DBu Mic Input	<.006 %	.002 %
0 DBu Mic Input	<.006 %	.002 %
-20 DBu Mic Input	<.008 %	.003 %
+10DBu Tape Input	<.006 %	.006 %
0 DBu Tape Input	<.015 %	.015 %
MIC IN to Direct out +	14Dbu <.003 %	.002 %

Frequency Response

Channel mic in to Master out = 10Hz - 60kHz ±1DB Master out > 10Hz - 100kHz +1DB -3DB

Hum and Noise

(input gain @ maximum, input sensitivity -55DB)				
Equivalent input noise	<-128 DBu			
Residual output noise	< -106 DBu			
Master fad nom. one ch nom.	< -70 DBu			
Common mode rejection	< -76 DBu @1kHz			

Crosstalk

Adjacent channels	< -80DBu @ 1kHz
Mic - master output L to R	< -65DBu @ 10kHz

Voltage Gain

Mic in to master output > 84DB Tape in to master output > 30DB Mic in to direct out > 77DB

Channel EQ

Low Cut	12DB /octav	e75Hz		
HIGH:	Shelving	±15DB	@ 12kHz	
HI MID:	Peaking	±15DB	@ 200 - 5kHz,	Q = 0.67
LO MID:	Peaking	±15DB	@ 80 - 2kHz,	Q = 0.67
L0 :	Shelving	± 15DB	@ 75Hz	

Channel Peak indicators

Full brightness @ -3DB < clip

Phantom Power

+48V available at all balance XLR microphone inputs

Accesories

AN24 Touring case AN40 Touring case AN56 Touring case G12V 12V XLR light SLP5600 120∨ Spare Power Supply SP10 10' Speakon™ power cable SP25 25' Speakon™ power cable SL-88001 & channel input card SL-88005 main meter bridge card SL-88005 channel meter bridge card

contact Carvin at 800-854-2235 to order, or visit www.carvin.com

B. Schematics

S/L Channel Schematic







































D. Glossary

AFL (After Fade Listen)

A function that allows the operator to monitor the post-fade signal in a channel independently of the main mix.

Assign

To route a signal to a particular signal path or combination of signal paths.

Attenuate

To reduce signal level.

Auxiliary Send

An output from the console comprising a mix of signals from the channels derived independently of the left, right and group mixes.

Balanced

A method of connecting audio signals using 3 wires. Two of the wires carry the audio signal out of phase from each other allowing common mode rejection of in phase noise that may be picked up on each wire. The third wire is the ground. The common use is with the XLR connectors on mics and mixer outputs. Also the 1/4" phone TRS (Tip/Ring/Sleeve) system is used for making balanced connections.

Bandwidth

The band of frequencies that pass through a device with a loss of less than 3dB, expressed in Hertz.

Cardioid

Cardioid refers to the heart-shape sensitivity pattern of some directional microphones.

Chorusing

An effect available in some digital delay effects units and reverbs. Chorusing involves a number of moving delays and pitch shifting, usually panned across a stereo field.

Clipping

The onset of distortion in the signal path, usually caused by the peak signal voltage being limited by the circuit's power supply voltage.

Condenser

Another term for the electronic component generally known as a capacitor. In audio, condenser usually refers to a type of microphone that uses a capacitor as the sound pickup element. Condenser microphones require electrical power to run internal amplifiers and maintain an electrical charge on the capacitor. They are typically powered by internal batteries or "phantom power" supplied by an external source, such as a mixing console.

Console

A term for a sound mixer, usually a large desk-like mixer.

Cueing

In broadcast, stage and post-production work to "cue up" a sound source (a record, a sound effect on a CD, a song on a tape) means to get it ready for playback by making sure you are in the right position on the "cue," or making sure the level and EQ are set properly. This requires a special monitoring circuit that only the mixing engineer hears. It does not go out on the air or to the main mixing busses. This "cueing" circuit is the same as the PFL feature on a mixer.

DAT

Digital Audio Tape, a tape-based digital recording format.

dB (see decibel)

A ratio of two voltages or signal levels, expressed by the equation dB=20Log 10 (V 1 /V 2).

dBm

A unit of measurement of audio signal level in an electrical circuit expressed in decibel referenced to 1 milliwatt. The "m" in dBm stands

for "milliwatt." In a circuit with an impedance of 600 ohms, this reference (0dBm) corresponds to a signal voltage of 0.775 VRMS (because 0.775 V across 600 ohms equals 1mw).

dBu

A unit of measurement of audio signal level in an electrical circuit, expressed in decibel referenced to 0.775 VRMS into any impedance. Commonly used to describe signal levels within a modern audio system.

dBv

A unit of measurement equal to the dBu but no longer in use. It was too easy to confuse a dBv with a dBV, to which it is not equivalent.

dBV

A unit of measurement of audio signal level in an electrical circuit, expressed in decibel referenced to 1 VRMS across any impedance. Commonly used to describe signal levels in consumer equipment. To convert dBV to dBu, add 2.2dB.

Decibel (see dB)

The dB is a ratio of quantities measured in similar terms using a logarithmic scale. Many audio system parameters measure over such a large range of values that the dB is used to simplify the numbers. A ratio of 1000V:1V=60dB. When one of the terms in the ratio is "an agreed upon standard value" such as 0.775V, 1V or 1mw, the ratio becomes an absolute value, i.e., +4dBu, -10dBV or 0dBm.

Delay

In sound work, delay usually refers to an electronic circuit or effects unit whose purpose it is to delay the audio signal for some short period of time. Delay can refer to one short repeat, a series of repeats or the complex interactions of delay used in chorusing or reverb. When delayed signals are mixed back with the original sound, a great number of audio effects can be generated, including phasing and flanging, doubling, Haas-effect positioning, slap or slap back, echo, regenerative echo, chorusing and hall-like reverberation. Signal time delay is central to many audio effects units.

Detent

A point of slight physical resistance (a click) in the travel of a knob or slide control.

DI(direct injection)

The practice of connecting an electric musical instrument directly to the input of the mixing console rather than using a microphone into the console.

Direct output

A post fader line level output from the input channel, bypassing the summing amplifiers, typically for sending to individual tape tracks during recording.

Doubling

A delay effect, where the original signal is mixed with a medium (20 to 50 msec) delay.

Dry

Usually means without reverberation or without some other applied effect like delay or chorusing.

Dynamic Microphones

Refers to the class of microphones that generate electrical signals by the movement of a coil in a magnetic field. Dynamic microphones do not require external power.

Dynamic range

The range between the maximum and minimum sound levels that a sound system can produce. It is usually expressed in decibels as the difference between the level at just before peak clipping and the level of the noise floor.

Echo

The reflection of sound from a surface. Reverberation and echo are terms that can be used interchangeably, but in pro audio a distinction is usually made: echo is considered to be a distinct, recognizable repetition (or series of repetitions) of a word, note, phrase or sound, whereas reverberation is a diffuse, continuously smooth decay of sound.

Effects unit

An external signal processors used to add reverb, delay, spatial or psychoacoustic effects to an audio signal.

EIN (Equivalent Input Noise)

A specification that helps measure the noise generation of a gain stage.

Equalizer (EQ)

A device that allows the boosting or cutting of selected bands of frequencies in the signal path. In the form of tone controls or sliders.

Fader

A linear control providing level adjustment

Feedback

The "howling" sound caused by bringing a microphone too close to its associated loudspeaker.

Flanging

A modulation effect. Before digital delay effects units, the flanging effect could be accomplished by playing two tape machines in synchronization, then causing one tape to become slightly out of synch by rubbing a finger on the tape reel.

FOH

An acronym for Front Of House. Refers to the main stage to audience sound system.

Frequency

The number of times an event repeats itself in a given period. Frequency is measured in units called Hertz (Hz). One Hertz is one repetition or cycle per second.

Gain

The measure of how much a circuit amplifies a signal.

Gain stage

An amplification point in a signal path, either within a system or a single device.

Graphic EQ

A graphic equalizer uses slide pots for its boost/cut controls with its frequencies evenly spaced through the audio spectrum.

Ground

Also called earth. Ground is defined as the point of zero voltage in a circuit or system, the reference point from which all other voltages are measured. In electrical systems, ground connections are used for safety purposes to keep equipment chassis and controls at zero voltage and to provide a safe path for errant currents. This is called a safety ground. Maintaining a good safety ground is always essential to prevent electrical shock. Follow manufacturer's suggestions and good electrical practices to ensure a safely grounded system. Never disable the safety ground to reduce ground loop noise problems.

Ground loop

A ground loop occurs when the technical ground within an audio system is connected to the safety ground at more than one place. Two or more connections will allow tiny currents to flow in the loops created, possibly inducing noise (hum) in the audio system. If you have noise in your system due to ground loops, check your manual for wiring tips or call technical support. Never disable the safety ground to reduce noise problems.

Haas effect

A psychoacoustic effect in which the time of arrival of a sound to the left and right ears affects our perception of direction. If a signal is presented to both ears at the same time at the same volume, it appears to be directly in front of us. But if the signal to one ear, still at the same volume, is delayed slightly (0 to 5 msec), the sound appears to be coming from the earlier (non-delayed) side.

Headroom

The available signal range above the nominal level before clipping occurs.

Hertz (Hz)

Unit of measure for frequency of oscillation, equal to 1 cycle per second.

Highpass filter

A filter that rejects low frequencies.

Impedance

The A.C. resistance/capacitance/inductance in an electrical circuit. Impedance may not be purely resistance but is mesaured in ohms and is usually indicated by the omega symbol (Ω).

Insert

A break point in the signal path to allow the connection of external devices, for instance signal processors.

Line level

A signal whose level falls between -10dBu and +30dBu.

Master

A control affecting the final output of a mixer. A mixer may have several master controls, which may be slide faders or rotary controls.

Mic level

The typical level of a signal from a microphone. A mic level signal is generally below -30 dBu.

Mic preamp (mic-pre)

Short for microphone preamplifier. An amplifier that functions to bring the very low signal level of a microphone (approximately –50dBu) up to line level (approximately 0dBu). Mic preamps often have their own gain control to properly set the volume for a particular source. Setting the mic preamp gain correctly is an essential step in establishing good noise and headroom for your mix.

Monaural (MONO)

Monaural has to do with a signal which, for purposes of communicating- audio information, has been confined to a single channel. One microphone is a mono pickup; many microphones mixed to one channe is a mono mix; a mono signal played through two speakers is still mono, since it only carries one channel of information. Several monaural sources, however, can be panned into a stereo mix. Monaural sound reinforcement is common for environments where stereo sound reinforcement would provide an uneven reproduction to the listener.

Monitor

In sound reinforcement, monitor speakers (or monitor headphones or in-the-ear monitors) are those speakers used by the performers to hear themselves. Monitor speakers are also called foldback speakers. In recording, the monitor speakers are those used by the production staff to listen to the recording as it progresses.

Monitor send, also (foldback)

A feed sent from the mixer back to the artist via loudspeakers or headphones to enable them to monitor the sounds they are producing.

Noise floor

The residual level of noise in any system. In a well designed mixer, the

Noise floor will be a quiet hiss, which is the thermal noise generated by bouncing electrons in the transistor junctions. The lower the noise floor and the higher the headroom the more usable dynamic range a system has.

Parametric EQ

A "fully" parametric EQ is an extremely powerful equalizer that allows smooth, continuous control of each of the primary EQ parameters (frequency, gain, and band-width) in each section independently. "Semi" parametric EQs allow control of fewer parameters. Usually frequency and gain (as they have a fixed bandwidth, but variable center frequency and gain).

Pan (pot)

Abbreviation of "panorama" controls levels sent to left and right outputs.

Phantom power, +48V

The phantom power supply, available at the channel mic inputs, for condenser microphones and active DI boxes that require power.

Peaking

An equalizer response curve affecting only a band of frequencies (i.e. based on a bandpass response).

PFL (pre-fade listen)

A function that allows the operator to monitor the pre-fade signal in a channel independently of the main mix.

Phase

A term used to describe the relationship of two audio signals. In-phase signals reinforce each other, out-of-phase signals result in cancellation.

Polarity

A term used to describe the orientation of the positive and negative poles of an audio connection. Normally connections are made with positive to positive, negative to negative. If this is reversed, the result will be out-of-phase signals

Post-fader

The point in the signal path after the fader and therefore, affected by fader level.

Pre-fader

The point in the signal path before the master fader position and therefore, unaffected by the fader level.

Q

A way of stating the bandwidth of a filter or equalizer section. An EQ with a Q of .5 is broad and smooth, while a Q of 10 gives a narrow, pointed response curve. To calculate the value of Q, you must know the center frequency of the EQ section and the frequencies at which the upper and lower skirts fall 3dB below the level of the center frequency. Q equals the center frequency divided by the difference between the upper and lower –3dB frequencies. A peaking EQ centered at 10kHz whose –3dB points are 7.5kHz and 12.5kHz has a Q of 2.

RCA phono jack

An RCA phono jack is an inexpensive connector (female) introduced by RCA and originally used to connect phonographs to radio receivers and phono preamplifiers. The phono jack was (and still is) widely used on consumer stereo equipment and video equipment but is not generally used in the professional and semiprofessional sound world.

Reverberation (reverb)

The sound remaining in a room after the source of sound is stopped. It's what you hear in a large room immediately after you've clapped your hands. Reverberation and echo are terms that can be used interchangeably, but by professional audio definitions a distinction is usually made: reverberation is considered to be a diffuse, continuously smooth decay of sound, whereas echo is a distinct, recognizable repetition of a note, phrase or sound.

RMS

An acronym for root mean square, a conventional way to measure AC voltage and audio signal voltage. Most AC voltmeters are calibrated to read RMS volts. Other types of measurements include average volts, peak volts and peak-to-peak volts.

Send

A term used to describe a secondary mix and output of the input signals, typically used for foldback monitors, headphone monitors, or effects devices. Often mixers use the term Aux Send for Auxiliary Send.

Shelving

An equalizer response affecting all frequencies above or below the break frequency (i.e. a highpass or lowpass derived response.)

Solo

A solo circuit allows the engineer to listen to individual channels, buses or other circuits singularly or in combination with other soloed signals.

Stereo return

An input specifically designed to receive the output of effects or other external processing devices

Sweep EQ

An equalizer that allows you to "sweep" or continuously vary the frequency with a fixed Q.

Talkback

The operator speaking to the artists or to tape via the auxiliary or group outputs.

Tinnitus

The ringing in the ears that is produced with prolonged exposure to high volumes. *BE AWARE OF YOUR EXPOSURE OR PERMANENT EAR DAMAGE WILL RESULT!*

Transient

A momentary rise in the signal level.

TRS

Acronym for Tip-Ring-Sleeve, a scheme for connecting three conductors through a single plug or jack. This would include 1/4" and 1/8" mini phone plugs and jacks. Since the plug or jack can carry two signals and a common ground, TRS connectors

are often referred to as stereo or balanced plugs or jacks. Another common TRS application is for insert jacks, used for insert an external processor into the signal path. Typically, the tip is send, ring is return, and sleeve is ground.

ΤS

Acronym for Tip-Sleeve, a scheme for connecting two conductors through a single plug or jack. This would include $1/4^{"}$ and $1/8^{"}$ mini phone plugs and jacks. Sometimes called mono or unbalanced plugs or jacks. A $1/4^{"}$ TS phone plug or jack is also called a standard phone plug or jack.

Unbalanced

A method of audio connection which uses a single wire and the conductive shielding as the signal return. This method does not provide the noise immunity of a balanced input.

Unity gain

A circuit or system that has its voltage gain adjusted to be equal one or unity. A signal will leave a unity gain circuit at the same level at which it entered.

volume

Electrical or sound level in an audio system.



12340 World Trade Drive, San Diego, CA 92128 (858) 487-1600 (800) 854-2235 www.carvin.com