CARVIN

MX22

12340 World Trade Drive., San Diego, CA 92128
800-854-2235

INTRODUCTION
"ABOUT THIS MANUAL"

OVERVIEW

MX-22 series "Stereo" mixing consoles are specifically designed to address live sound reinforcement as well as a host of other requirements targeting professional sound companies and users. This manual covers all MX-22 series consoles from six channels to 24 channels. If there are any differences between models, the differences are appropriately noted and explained. The MX-22 units (without the suffix "S") are designed to be used along with power amplifiers. Some units have built-in power amplifiers designated by suffix "S" in their model names. All MX22 series consoles may be used with crossovers (active or passive), signal processing equipment, recording equipment and speakers - those made by CARVIN or other manufacturers. Like any mixing console, the performance of the MX-22 series consoles depends upon the system design and installation. So, the content of this manual centers around explaining the features and various connections that can be utilized on the MX-22 series consoles assuring you the best possible operation and performance from your system. Additionally, this manual explains how to use an MX-22 series console in setting up a sound system, special features of the console, technical aspects of the MX-22 series consoles and a glossary to help explain any terms you may not be familiar with.

CONTENT

We recommend that you read the entire Operator's Manual. However, if you are already familiar with professional consoles you may wish to skip to section (1) for the "MX-22 Series Control Descriptions." This section contains basic information regarding channel features and connections needed for operation.

"Getting To Know Your Console" (Section #3 of your manual) goes into a detailed and easy to read description of the front panel features and rear panel connections on the MX-22 series consoles. Each of the features explained are numbered and include a small picture of the feature described. Also, each number assigned to the feature corresponds to the numbers assigned in the "MX-22 Series Control Descriptions" (Section #1) for additional information.

"Special Features Of Your Console" (Section #4) describes important information on maintaining your console, how to use the various patch jacks on the console, and basic information that might be needed to achieve maximum performance from the console.

"Setting Up Your Sound System" (Section #5) will provide diagrams and illustrations as well as information on how to hook up a complete sound system using the MX-22 series console. Various cabeling considerations, shielding concepts and several other topics are described.

"Suggestions For Efficient Set-up And Quality Sound" (Section #6) addresses the various procedures and needs in planning for a performance. It includes information on previewing the concert hall, maintenance or repair tool kits, powering up the sound system, and checking out the sound system for maximum performance.

The "Glossary" (Section #7) Describes the terms that are most commonly used in the sound reinforcement business relating to the use of the MX-22 series consoles.

The "Service Section" (Section #8) includes prices for circuit modules and accessories, technical specifications, circuit schematics, part listings and placement, wiring diagrams, harness diagrams and block diagrams. This section is designed to be a detailed overview of the technical aspects of the design and construction of the MX-22 series consoles.

INSPECTION & PRECAUTIONS

INSPECT YOUR CONSOLE AND THE SHIPPING CARTON FOR ANY DAMAGE which may have occured in shipping. If damage is found, notify the shipping company & CARVIN immediately and obtain a **DAMAGE INSPECTION REPORT** from the shipping company. Send a copy of the damage inspection report to CARVIN and return the goods to Carvin. This will allow CARVIN to process any damage claim with the shipping company and provide you with the fastest return of new goods. All goods must first be received back at CARVIN prior to exchanging or shipping a new item to you. This is both for your's and Carvin's protection. If you file a "Damage claim" you will have to settle directly with the shipping company. And, upon receiving your settlement you will then have to reorder a new replacement.

SAVE THE CARTON & ALL PACKING MATERIALS. In the event you have to reship your console, always use the original carton and packing material. This will provide the best possible protection for your unit during shipment. Both CARVIN and the shipping company will not accept liability for damage caused by improper packing.

SAVE YOUR INVOICE. It will be required for warranty servicing of your unit. Always check your invoice against the items you have received. If you find some items missing it may be that they were simply split up during shipment. Please allow several days for the rest of your order to arrive before inquiring. If you determine (after allowing an appropriate amount of time) you have not received all your items, please call CARVIN in order that we may file a tracer and take the proper steps to assure that you receive all the items in your order.

CAUTION - TO PREVENT ELECTRIC SHOCK DO NOT DEFEAT THE SAFETY GROUND ON THE POWER CORD.
WARNING - TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE TO RAIN, MOISTURE, EXPLOSIVE ATMOSPHERE OR INSTALL AN IMPROPER FUSE.

TOLL FREE NATIONAL (800)-854-2235

CONTENTS

"ABOUT THE MX-22 SERIES CONSOLE"

The MX-22 series consoles represent some of the mostadvanced, up-to-date, and uncompromised engineering in professional live reinforcement consoles today. The MX-22 series consoles have been used in virtually every aspect of professional sound reinforcement and have been praised by top artists world wide. The basic concepts surrounding the design of the MX-22 series consoles are centered on "real world" needs for a reliable and flexible console, required for today's demanding live performances. Every detail of the MX-22 series project received careful scrutiny. In depth research including surveys of the professional sound community regarding the most important features, the most efficient metering, monitoring, and interface systems, new and innovative circuit designs, highly sophisticated production techniques and all aspects of the MX-22 series project were well thought out. This ultimately allowed the MX22 series project appropriately address the requirements for a "total mixing system" for concert, broadcast, theater and recording.

The MX-22 series consoles were subjected to an extended program of rigorous laboratory and field testing. This testing has allowed Carvin to fine tune the features and performance of the MX-22 series consoles. The result. . .cleaner and quieter performance than the competition. Every aspect of this testing was scrutinized in great detail. The panel layout, color scheme, exterior dimensions and aesthetics were all important considerations in helping to make the MX-22 series consoles a "natural" for professional live mixing. The MX-22 series consoles, although extremely sophisticated and complex in function, are uncluttered and sensibly organized; mixing with the MX-22 series consoles is a breeze!

The input channels of the MX-22 series mixers utilize ultra-low noise differential mic pre-amps with mic/line switching and extremely low noise op-amps. Careful attention to gain structure and circuit design provides absolutely quiet performance. The residual output noise is below -90dBm! Each channel features (3) band equalization with fully sweepable midrange. The clarity and flexibility of this equalizer is exceptional! Each channel also has two monitor sends and one effects send. The "Effects Send" control on each channel controls the amount of internal "built in" reverb for that channel. Carvin put extensive effort into designing the reverb system for the MX-22 consoles. Extensive listening tests and research went into providing the smoothest, most natural sounding reverb possible. It incorporates the Hammond 3-spring reverb tank and specially designed driver circuitry to provide a quiet yet responsive reverb. An additional control allows reverb to be returned to the "MON 1" mix so that performers on stage can have the benefit of reverb in their monitors.

All MX-22 series consoles feature two built-in graphic equalizers. These equalizers are patchable and can be used in any signal path. They feature the same advanced circuitry and construction found in Carvin's highly praised EQ-2029 1/3 ocatve graphic equalizers!

After looking at the way typical power amplifiers reproduce music program material we were struck by the fact that audio power amplifiers cannot even approach their rated output power before audible clipping occurs on signal peaks. For example, a typical 100 watt amplifier cannot deliver more than 10 watts of average power before audible distortion occurs on the peaks of recorded music. This means that only about 10% of the available amplifier power can be used before an amp is driven into excessive distortion. For live music production this situation can be even worse (because of the extremely dynamic nature of live sound). At Carvin, we decided to look for a way to make better use of the power available in our powered mixing consoles. We found that by adding a very carefully programed dynamic range compressor at the input of the power amp we could significantly increase the average output power of our amps before audible distortion. This resulted in a large increase in real output power.

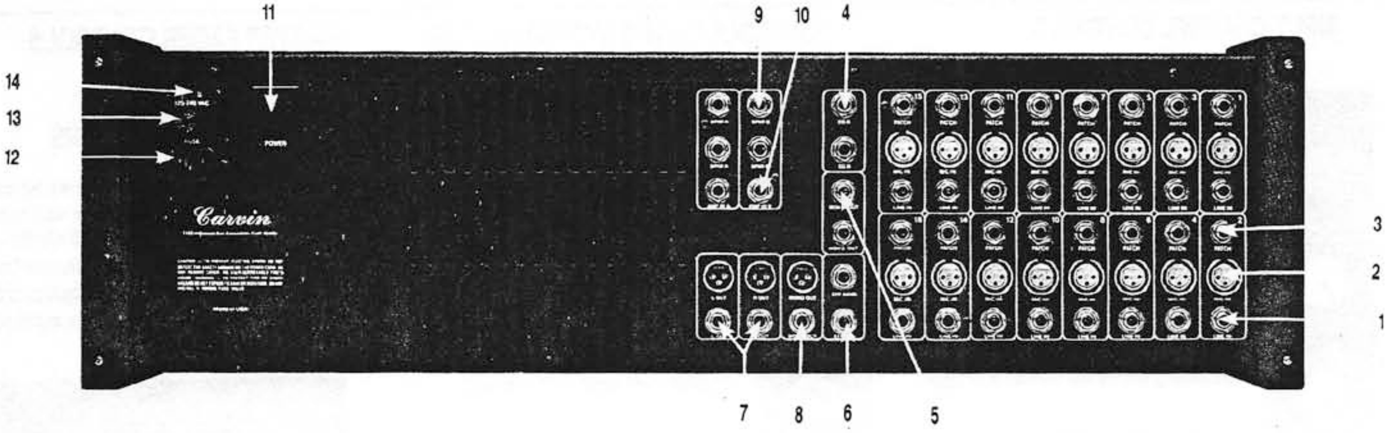
SECTION #1-2

It effectively allows the amp to achieve volume levels that could only be reached by conventional amps rated at over twice the power! When you compare the power amp specifications of the MX-22 series consoles to other products remember that the nominal power rating of 300 or 400 watts will not reflect the actual loudness that these mixers will achieve with real world program material. Also, because of the nature of Carvin's exclusive "Power-Track" circuit it should be noted that amplifier power cannot be measured accurately without first defeating this circuit. The power track circuit coupled with the high power MOSFET amplifiers provide an unbeatable combination of performance and reliability. The power amp sections in the MX22 series consoles will deliver over 800 watts!

The MX-22 series consoles are built with only the finest components. Electronically, all circuitry is designed for ultra-low noise performance and extreme reliability. An internal heavy duty power supply spares the expense and inconvenience of a separate outboard supply and is positioned in such a way that system noise is not affected. Not only are the MX-22 consoles quiet internally, but they also effectively reject external noise. Highly effective bi-polar differential inputs and outputs reject common mode noise, while extensive shielding and RFI suppression circuitry provide immunity to radio frequency interference. Each channel of the MX-22 features a separate printed circuit card for modularity and reliability. These channel cards are connected with highly reliable modular connectors and can be removed easily and quickly. Careful attention to signal routing and gain levels through the computer ribbon-type interconnect cables allow the MX-22's to achieve extremely low crosstalk when compared to other consoles. (Typically -65dB at 1Khz/-55dB @ 10Khz)!

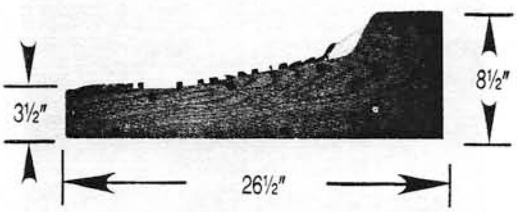
The Carvin MX-22 series consoles are an ultra-reliable state-of-the-art console. They are backed by Carvin's commitment to excellence and experience in building thousands of professional consoles as well as speaker systems, power amplifiers, crossovers, equalizers, etc. The MX-22 series consoles were well thought out and are at the forefront of modern mechanical and electronic design. Carvin is very proud of the MX-22 series consoles and sincerely wishes you the enjoyment and the satisfaction that comes from knowing your own "The best:" - "The MX-22 series console!"

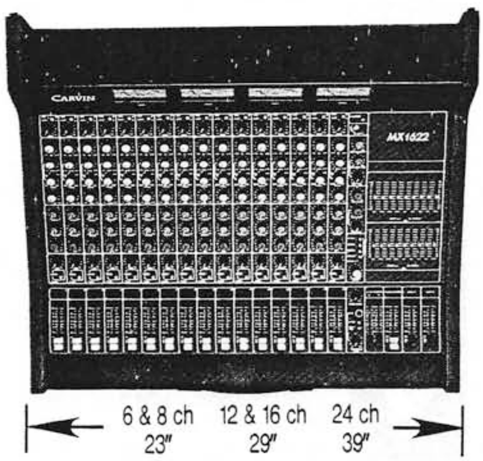
REAR PANEL VIEW



1. LINE INPUT JACK (Unbalanced)
2. MIC INPUT CONNECTOR (XLR Balanced)
3. CHANNEL PATCH JACK (Send/Return)
4. GRAPHIC EQUALIZER PATCH JACKS
5. MON 1 and MON 2 OUTPUT JACKS
6. EFF SEND AND EFF RTN JACKS
7. TWO-TRACK LEFT AND RIGHT OUTPUT CONNECTORS (Balanced and Unbalanced)
8. MONO OUTPUT CONNECTORS
9. POWER AMP SPEAKER JACKS (Powered Models Only)
10. POWER AMP INTERRUPT JACKS (Powered Models Only)
11. POWER ON/OFF SWITCH
12. POWER CORD CONNECTOR
13. AC LINE FUSE
14. 120/240V SWITCH

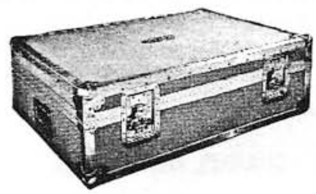
MODULAR CONSTRUCTION





MX22 DOCUMENTATION

The MX22 100 page documentation is complete with schematics, parts lists, parts placement diagrams, specifications, and operating guide. For a preview, you may send $15 for the MX22 Service Manual No. 5006.



Anvil™
Flight Approved
Case.

CONTROL DESCRIPTIONS

INPUT CHANNEL CONTROLS

MIC/LINE SWITCH

This switch is located at the top of each input channel and selects either the microphone preamp or the line preamp as the feed to the channel.

INPUT GAIN CONTROL

This control adjusts the gain of both the mic preamp and the line preamp in 41 steps.

CHANNEL EQUALIZER

Each input channel has a three band equalizer (EQ) with sweepable midrange frequency control. The high and low EQ bands have a shelving response. The mid EQ is more sophisticated in that it can be set to either boost or cut a band of mid range frequencies while a second FREQUENCY control allows the exact range of action to be varied anywhere from the lower mid tones to upper mid tones. This sweep action allows you to quickly locate problem frequency ranges and make your EQ adjustments. The range of EQ action is as follows:

High: ±15 dB @ 10kHz
Mid: ±15 dB w/200 Hz to 4kHz sweep
Low: ±15 dB @ 80 Hz

MONITOR AND EFFECTS SENDS

Each channel signal can be sent to the two monitor mixes and the effects mix by raising the MON 1, MON 2, and/or EFF controls. The monitor and effects mixes are independent of each other and independent of the two-track mix as well. Monitor sends are prefader—post EQ. Effect send is post fader—post EQ.

CHANNEL PAN CONTROL

The channel PAN control allows the channel signal to be panned across the two-track mix.

CHANNEL SOLO SWITCH (POST FADER)

When this switch is depressed the channel signal is switched into the phones in place of the normally selected audio. If more than one SOLO switch is depressed then the combined solo signals will be heard.

PEAK WARNING LIGHT

The channel PEAK light flashes to warn the operator whenever signal peaks come within 6 dB of overdriving any stage in the input channel.

CHANNEL FADER (CALIBRATED)

CONTROLS ON THE SYSTEM MASTER STRIP

PHANTOM POWER SWITCH

Depressing the PHANTOM power switch causes a standard phantom power voltage (+48 VDC) to be applied for condenser mics.

MINI-LAMP CONNECTOR

A BNC connector is provided for connecting an optional 6V mini-lamp.

REVERB RETURN GROUP

These three controls allow reverberation from the built-in studio reverb system to be returned to both the MON 1 mix, and the two-track mix. The pan control can be used to pan the reverb across the two-track mix.

EFFECTS SEND AND RETURN CONTROLS

The control labeled "EFF SEND" sets the overall level of the signal at the "EFF SEND" jack on the rear panel. This signal is usually routed through an external effects processor and the output of that processor is then connected back into the mixer at the effects return ("EFF RTN") jack. This effects return signal can be added to the two-track mix by raising the EFF RTN control located below the EFF SEND control. The EFF PAN control allows the effects return signal to be panned across the two-track mix.

PHONES INPUT SELECT AND LEVEL CONTROL

The headphone volume is set by the PHONES LEVEL control. Located above this control are four switches and the master solo light. The input to the phones amp is selected at the switch group. Any combination of switches may be selected (2 TRK, MONO, MON 1, or MON 2).

SOLO LEVEL CONTROL

A SOLO LEVEL allows the volume of solo signals to be set different from the normal phones audio (12, 16, & 24 CH only).

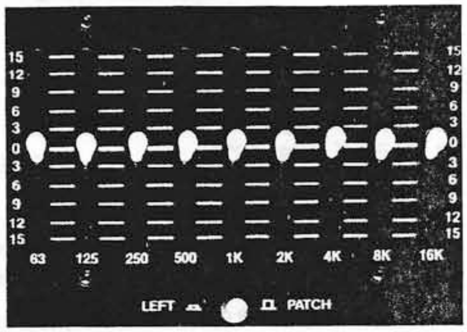
TALKBACK SWITCHES AND LEVEL CONTROL
(12, 16, & 24 ONLY)

The MX22 talkback system employs a built-in condenser microphone to allow the mixer operator to talk through either the monitor or the main outputs.

MASTER FADER CONTROLS

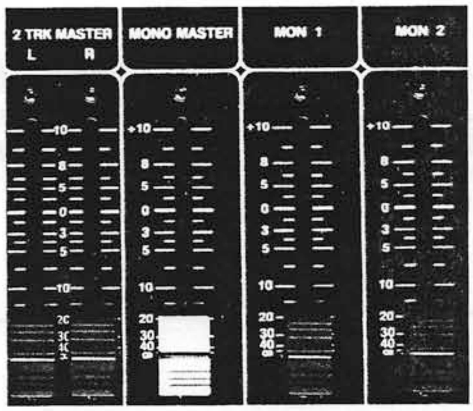
GRAPHIC EQUALIZERS

All models of MX22 series mixers are equipped with two nine-band graphic equalizers. Below each equalizer is a switch which connects EQ's A and B into the left and right outputs respectively when the switches are depressed. When the equalizer's switch is not depressed then the EQ is available to be patched into any other signal path by way of the rear panel patch jacks.



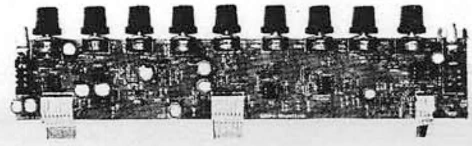
MASTER FADERS

Located below the graphic EQ's are the master faders for the two-track, mono, and monitor 1 and 2 outputs. The "2 TRK MASTER" faders are a close spaced pair of controls which set the left and right output levels. The MONO MASTER fader sets the level at the MONO output.



MODULAR CIRCUIT CARDS

The MX22 series mixers incorporate totally modular circuitry. Each modular card features precision connectors that are easily disconnected for quick card changes in the field. Computer type ribbon cable not only enhances performance by reducing crosstalk, but aids in the simplicity of removing and installing cards.



GETTING STARTED USING YOUR NEW CARVIN CONSOLE

"A SPECIAL MESSAGE TO THE NEW OWNER"

Congratulations on your selection of CARVIN products "The Professional's Choice." Your new "MX" 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 by people from the results they achieve through their efforts and knowledge. It is not something that automatically happens when buying a new or more sophisticated console. 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 professional gain & enjoyment.

To compliment your new console and help you acquire that knowledge, we've included this manual. All the information you need to be up and running is right here! You'll find using this manual easy and pleasant. 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, please call us at our toll free numbers. Our sales staff is well versed in the technical aspects of our products and are eagerly willing 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. Criticism and comments from our owners have helped us improve and further develop our products and our business. 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. And, it helps us mail out 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 plenty of diagrams & descriptions to aid in understanding your new console. So, with this manual in hand you hold the key to proper operation your new console, and to achieve truly professional results.

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

TOLL FREE NATIONAL (800)-854-2235

FEATURES OF THE MX-22 SERIES CONSOLES

- Three Band Channel EQ with Sweepable Midrange
- Two Nine Band Graphic Equailzers with Patch Points
- Built-in Reverberation System
- Two Independent Monitor Mixes
- Headphone Monitoring of 2-Track, Mono, Mon #1, or Mon #2 Outputs
- Solo on all Input Channels with Headphone Interrupt
- Effects Send from Channels (Post Fader)
- Effects Return with Level and Pan Controls
- Channel Peak Warning Indicators with "Peak Stretching"
- Patch Points on all Channels
- Microphone Phantom Power (+48 VDC)
- Totally Modular Internam Construction
- Input Noise of -127 dBv
- 4dB or -10dB Operating Level
- THD Less than .05%
- Eleven Step Gain Controls For Easily Repeated Set-up
- Outlet for 6V Mini-Lamp (Optional Accessory)
- MOSFET power amplifier (800 watts) powered models

SECTION #3-2

GETTING TO KNOW YOUR CONSOLE
FRONT PANEL CONTROLS
INPUT CHANNEL CONTROLS

Each channel of your console has several knobs. And, each of these knobs controls a specific function in your console. What this section will do is describe each function and the best way in which to use it. You may wish to refer to the glossary at the end of the manual for any terms that you may need to have better clarified. Hopefully, by understanding the operating parameters of each of the controls of your console, you will be better able to blend the right tones and levels to provide the most natural sound and mix of the various instruments. Although simply understanding what each of the knobs do will not ensure absolutely perfect mixes, it will provide the basis upon which you can expand and further your creativity. So, here we go;

1) Mic/Line Switch

* This switch will select between the microphone input and the line input at the rear panel of your console. Remember, the line input will accept signals many times the strength of the mic input, and this switch must be properly selected depending upon which input best suits your signal source. Mic inputs typically accept signals from -60dB to -20dB. Line inputs can accept "pre-amp" signals (i.e. line outputs of tape decks, etc.) of up to 30 volts. You may have both the line inputs and the mic inputs hooked up at the same time. This is a very effective way in which to select between a tape input or a mic input. For instance, if you wish to play a tape back during a break, you may wish to have your tape deck inserted into the line inputs of an input already dedicated to mic input. This will allow you to switch to the tape source by simply depress the Mic/line switch. The gain control directly below this switch controls the relative gain from either the mic or line source to the channel.

2) Input Gain Control

* The input gain control is very important for establishing the proper signal-to-noise ratio performance of the console. This simply means that proper adjustment of this control will offer the lowest possible "hiss" or background electronic noise. The best way to use this control is to rotate it fully clockwise to the position marked as "10". This affords the maximum gain for the input. If the signal is too strong from either your microphone or line input the red LED "Peak" indicator just above the fader will light. This LED indicator will indicate a strong red light for a short period of time whenever the signal is too strong. Rotating the "Gain" control (one click at a time) counterclockwise until the LED "Peak" light just goes out will properly attenuate the signal to deliver just the right level to the channel. Rotating the Gain control excessively counterclockwise beyond the point at which the "Peak" light goes out will drop the signal too much and you may not have enough gain, or the floor noise (Hiss) level will become more apparent. The best rule of thumb is to rotate the "Gain" control all the way to "10" and look to see if the "Peak" LED is lighting while the source is active. If it is not, then no further adjustment is needed. However, if it is, then it is best to rotate the gain control counter clockwise until the Peak LED just goes

out. Once this adjustment has been made you should not need to again touch this control through the rest of your mix. You might want to experiment with this control a little bit to become familiar with its feel and operation. it is an excellent tool for properly matching the different gains of various microphones to your console.

3) <u>Channel Equalizer</u>

* The Channel Equalizer is a very precise tone control. The High Frequency of "Treble" control is at the top of the Channel Equalizer controls. The Mid-range tone (Controls) are in the center and are shown connected by a line drawing them together on the chassis markings. And, the Low Frequency or Bass control is at the bottom of this array.

How a tone control works is basically similar to a volume control. The difference being that a tone control literally controls the volume of a specified frequency range. For instance, the "Hi", or treble control, when turned up will change the volume of the high frequencies (at a 10K frequency center). Likewise, with the Bass Control, it adjusts the volume at an 80Hz frequency center. The Mid-Range control allows you to select and adjust a particular mid-range frequency. Using the frequency adjustment knob you can select a particular frequency center you wish to boost or cut. Then using the "upper" level adjustment control, you can dial in the amount of boost or cut you desire. With mid-range controls you may adjust the volume of a range of frequencies sweepable from 200Hz up to 4kHz. This is especially useful as the mid-range is usually the most critical "problem range . By being able to properly "zero in" on the desired mid-range frequency you wish to boost or cut, you are able to very effectively adjust this problem area. Again, experimentation is in order. It is good to note that radical adjustments of your tone controls should never have to be made. Usually these controls are used as a means of compensating the response of various microphones in order to achieve the most natural response of the instruments you are mixing. If you are making excessive adjustments with these controls you may be best advised to use a different microphone or use a different "Micing" technique. The ability to make an instrument sound as natural as possible through the use of your Channel Equalizer is part of the overall art of professional mixing and recording.

4) <u>Monitor & Effects Sends</u>

* The Monitor send controls are pre-fader volume controls that are usually dedicated to sending a signal to the stage monitors. Each channel has (2) Monitor send controls (Mon #1 & Mon #2). These control the individual channels volume that will be sent to its respective monitor. Remember, the monitor sends are all "pre-amp" signals and are designed to drive a power amplifier that will subsequently drive your monitor speakers. Each monitor level control on each channel adjusts the relative amount of volume that that channel will send to the main monitor output. So, it is entirely possible that you could have an entirely different stage "mix" or monitor mix than the main mix. For instance, you could have a vocal "pumped up" or louder in the monitor for the performer to concentrate on while feeding a relative soft output to the main mix to de-emphasize its level. Since stage monitors are typically right next to the microphones, usually they are the most susceptible to feedback during a show. It is highly recommended that you use caution when turning up and down monitors during a live performance. It will take

a certain amount of "feel" for addressing the proper monitor mix, however, experimentation and practice are again the key to the most consistent results and most professional performance.

The "Effects Send" control is a "Post Fader" control. This means that when the channel fader is off, so is the effects control. This is done because this control is usually normalized to a reverb effect, or delay effect that should vary in intensity with the variations of channel volume. This control also feeds the internal reverb system. By increasing the level of this control you will increase the amount of volume from the respective channel sent to the internal reverb tank.

Note: *In order to hear reverb from your system you will also have to turn up the "REV RTN to 2-Track" or "REV to MON 1" control located on the "System Master Strip".*

5) Channel Pan Control

* The channel pan control sets the relative volume of the channels output to the "Left & Right" outputs (2-Track mix). It is used to control the amount of volume you wish to have in either the left or right field. By "Panning" the channel fully to the left - sound will only be heard from the left speaker. By panning the signal to the right - sound will only be heard from the right speaker. This is assuming you are mixing in a stereo field.

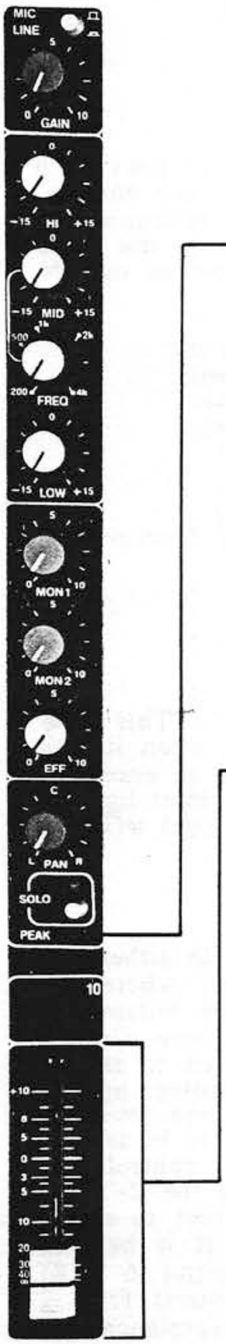
Since the (2-Track Master) controls feed the "Mono Master" control, you may use your Pan control to feed a "Sub-Mix". What this means is simply that you are fully panning one group of channels to "2-Track Master/Left" and fully panning a group of channels to "2-Track Master/Right". Then, whenever you bring up the level of either of the (2-Track Master) controls you will be controlling the level of those channels assigned to the respective 2-Track Master controls. This effectively allows you to control the level of several channels with one single fader. In a "Sub-group" situation (such as the above) you will usually be mixing in "Mono", therefore you will not be concerned about "Left or Right" but simply which channels will be affected by the sub-group faders.

Example: *You may pan your vocals to left, and instruments to right. The left output fader will control all channel faders assigned to left, and likewise with all faders assigned right. Since the "Mono" output is a sum of "Left" and "Right" your main mix will reflect only the relative levels of "Left" and "Right" fader settings.*

6) Channel Solo Switch

* The channel solo switch allows you to audition each channel in the headphones without affecting the main mix. What this means is that by depressing this switch you will be able to hear only that particular channel in the headphones, even though you may be using several inputs on your console. This is an extremely useful feature that will allow you to fine tune your mix during a performance. By being able to listen to one particular channel impervious to the rest of the channels you are able to focus your attention specifically to the sound and level of a particular channel as it relates to all the other channels you are mixing. You are able to combine solo's. This means that you may depress one or more solo switches in order to listen to

just certain instruments to be sure their relative levels are correct and that they are properly blending. Regardless of what is selected or what material you are listening to it the headphones, when you select a solo switch on your console only that channel or the combination of channels soloed will be heard in the headphones. Also, the solo has a "Solo Level" control located on the System Master Strip of your console (except MX-622 and MX-822 models). This will allow you to adjust the level in the headphones of any inputs that you may have soloed. Whenever you solo a channel a green "LED" indicator light will come on to indicate that a channel is soloed. Also, whenever a channel is soloed, a green LED indicator will light on the System Master Strip to indicate that you are in a solo mode.



7) Peak Warning Light

* The Peak Warning light will flash whenever any signal is exceeding the threshold level within the console. This light is used to warn the operator whenever signals are becoming too strong and will distort within the console. Whenever you see the "Peak LED" flashing, you should rotate the "Gain control" (see #2) counterclockwise until the LED just stops flashing. It is just as important to be sure not to rotate the "Gain" control too much counterclockwise (attenuating the signal). Too much attenuation will not allow you to achieve the best signal-to-noise performance your console is capable of. It is good to note that this indicator will show whenever "any" level is too strong on a channel. Whether the signal is too strong at the input, the output of the equalizer, or the output of the channel fader, the "Peak LED" will indicate the best performance. This indicator along with the "Gain" control work hand in hand as a first step whenever setting up your console for a mix. Proper use of these controls will deliver the best overall quietness and performance of your console.

8) Channel Fader

* The Channel Fader controls the volume of each channel. It is properly calibrated and adjusts the level of a channel as it is sent to the 2-Track Master. A proper setting of the channel fader would be between -5 and +8 on the fader markings. This means that usually you will be operating your channel faders quite high when compared to your 2-Track Master faders. Doing this will assure you the most quiet performance and best overall sound from your console.

GETTING TO KNOW YOUR CONSOLE CONTROLS ON THE ***SYSTEM MASTER STRIP***

1) Phantom Power Switch

* The Phantom Power switch (when depressed) will provide a +48V D.C. voltage to pins #2 & #3 of the mic inputs of your console. This voltage is required to operate "condenser" type microphones. Without "Phantom Power" you simply would not be able to use these types of microphones with your console unless you purchase an outside power supply to provide this voltage.

Note: *If most of your microphones are dynamic (not requiring phantom power) and you are operating several condenser type mics, you will need to depress the phantom power switch. This will not affect your dynamic microphones or their operation.*

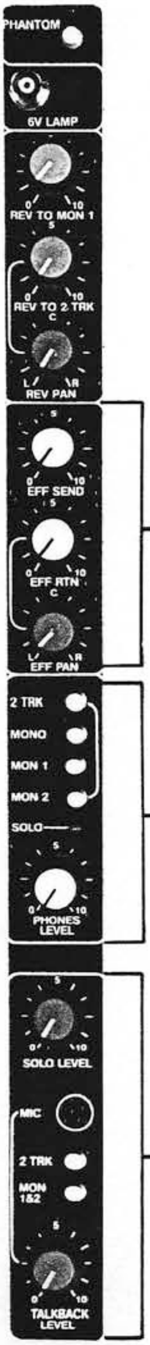
Note: *Before using the Phantom Power switch with wireless microphones check to be sure your mic is tolerant of Phantom Power. It is best to consult with the microphone's manufacturer to be absolutely certain.*

2) Mini-Lamp connector

* This is a receptacle for a "BNC" style "Little-Lite". This little light is used to provide some illumination to your console when it may be used in dark areas. It operates on 6 volts and provides an excellent source of light where house lighting is kept low. The mini-light is offered by CARVIN and may be purchased for $25.00 Model #G-12.

3) Reverb Return Group

* These controls adjust the overall level of reverb in either the Monitor or 2-Track Master mix. It is good to note that where the "EFF" control shown on each of the channels controls the volume of reverb with respect to the overall channels, the "Reverb Return Group" controls the overall reverb of all the channels as it is fed to the 2-Track Master outputs or the Monitor #1 Output. By turning up the level of the "REV to Mon #1" control you will adjust the level of reverb at the levels "Already pre-selected" by the channels to be sent to the Mon #1 output. By turning up the "REV to 2-Track" control you will be adjusting the overall level of reverb to appear at the 2-Track Master outputs. Also, you are able to "Pan" the reverb effect to either the left or right side of the 2-Track Master outputs. It is best to adjust the reverb level of your channels fairly high (around 6 to 8) where as you should usually adjust your "REV return" controls from 2 to 5. This will afford you the best signal quality and performance. It is best to experiment with these controls both on the channels and on the Master Strip. Reverb is a form of delayed ambience to provide a life to the mix in an otherwise dull sounding or dead room. Some rooms may require relatively large amounts or reverb whereas others may require little or none at all. You really have to experiment to find the best possible setting for your application.



4) <u>Effects Send & Return Controls</u>

* The "Effects Send" control enables you to adjust the level of signal that you will send to your outboard effect. The most commonly used effect is a digital delay although there are a host of other effects devices that may be used. A digital delay offers additional ambience and is usually used along with the internal reverb system. What the send control does is; it takes the relative levels already adjusted per channel by the EFF control and feeds the overall level of these signals to your Send Jack. Using the "Effects Send" control you are able to properly drive your effects with enough signal to operate it and provide the best performance from your effect. After you have fed the signal to your effect unit, adjusted its levels, and dialed in its proper parameters you will have to return the effects output signal to your console. So, you return your effect to the "EFF RTN" jack on the back of your console **(see #6 Rear Panel Connections)** and adjust its overall level in the mix using the "EFF Return" control.

* The "EFF RTN" control operates exactly the same as the REV Return control. It adjusts the level of your outboard effect as it is returned to the main mix. By using the EFF RTN and the REV to 2-Track controls properly you will be able to adjust the overall levels of both reverb and your effect as it appears at your 2-Track Master outputs. And, like the Reverb Return, you are able to Pan your effect to either the left or right side of your 2-Track Master output. Again, experimentation with your effects device along with the internal reverb system will best dictate the proper use for your application. Its a lot of fun mixing delays and other effects as well as reverb and can really spice up a mix. So, experiment and have fun!

5) <u>Phones Input Select and Level Controls</u> (12, 16 & 24 channel models only)

* These controls on the System Master Strip allow you to listen to various outputs of your console. By selecting the respective output listed you will be able to hear that particular signal in your headphones. Then you can adjust the level in your headphones using the "Phone Level" control labeled appropriately. Remember: When the "Solo" LED is lighted you will be listening (in the headphones only) to those inputs you have soloed. You will have to adjust the level of the soloed channels using the "Solo Level" control located directly beneath the "Phones Level" control, (except on MX-622 & MX-822 models).

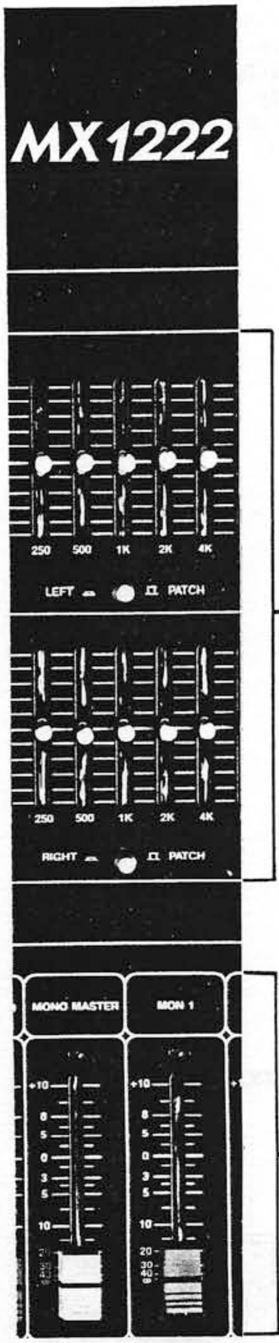
Note: Use only professional type headphones with impedance of 100 ohms or greater, 8 ohm headphones are not recommended.

6) <u>Talkback switches & Level Control</u> (12, 16 & 24 channel models only)

* The Talkback system is featured only in the 12 channel and larger consoles. It is not provided in the 6 and 8 channel models. This feature allows an operator to talk either through the monitor or main outputs of the console. For instance, if the operator wishes to talk to the performers on stage and he is remotely operating his console from a distance, he will simply depress the "MON 1 & 2" switch and adjust the "Talkback Level" control. He will then speak into the built-in condenser microphone and the signal will be fed to the Monitor #1 & #2 outputs that the performers will hear through their stage monitors. Likewise, if the operator must speak to the house, he will simply depress the "2-Track" button and speak into the built-in microphone. The signal will be fed to the 2-Track mix and

subsequently heard in the main house mix. This feature is excellent for narration, or setting up systems where the console is situated a great distance from the stage.

GETTING TO KNOW YOUR CONSOLE
ADDITIONAL FRONT PANEL FEATURES



1) Graphic Equalizers

* Graphic Equalizers provide a graphic representation of the overall levels of volume at various frequency bands. Instead of calling each "band" bass, midrange, or treble, each band is listed according to its exact frequency reference. Therefore the bands are listed as 63 Hertz - (63Hz), 125Hz, 250Hz, 500Hz, (1000 Hertz - 1K), 2K, 4K, 8K, and 16K. These numbers refer to the exact band in which you will be adjusting volume in the frequency spectrum. Because there are a few more controls than a standard (3) band equalizer, setting up a graphic equalizer may appear more difficult. However, at this point the best way to set your equalizer would be to experiment with each of the sounds of the different bands. Set each of the bands at "0" and boost and cut each band one at a time to become familiar with their respective sounds. Then experiment with the overall sound of all the bands of the equalizer until it sounds good to you. Let your ear be the best judge. Your ear is a very sensitive instrument and usually adjusting your equalizer to what best suits your ear is the best overall guide. Although there are more sophisticated electronic means of setting your equalizer, usually the final judge is the ear, and proper ear training will offer you the most consistent results when mixing various rooms. Again, experimentation and practice is the key. The graphic equalizers are usually used to properly adjust the response of the main speakers (controlled by the 2-Track Master Faders). So, the graphics are mainly used to "equalize" the response of the main speakers to provide the best sound for a given room. You are able to switch the graphics in or out for an instantaneous evaluation of how they are affecting your main speakers. And, should you desire to use these graphics for any other application (such as additional E.Q. for a channel) you may do so utilizing the patch jack for the graphics at the rear panel of the console. (See Section #4 item #1 for diagram of Graphic Patch Jack.)

2) Master Faders

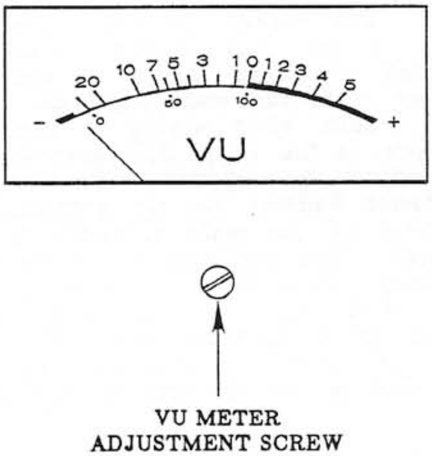
* The Master Faders control the overall volume levels selected by your channel controls. Usually the best operating parameters for your "Master Controls" is to operate them from approximately -20 to -3 on the markings. This in addition to a higher channel fader setting will provide the most quiet mix possible, minimizing background electronic hiss.

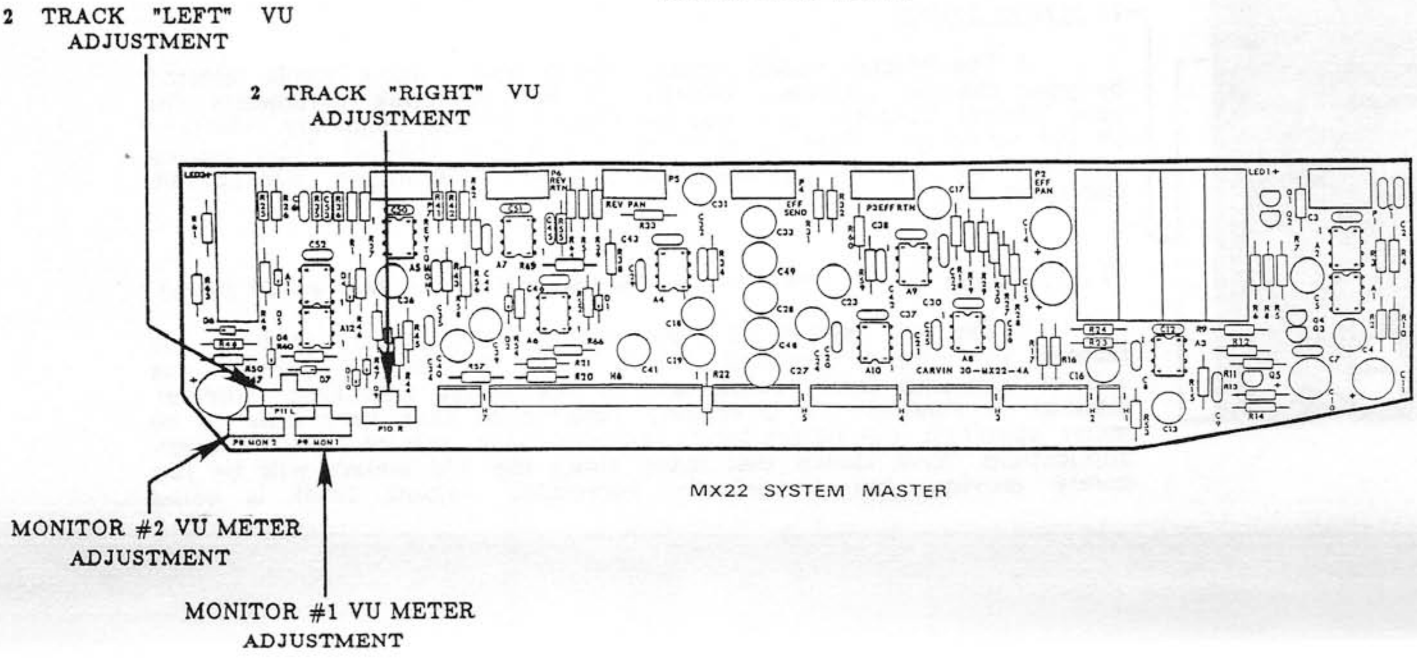
3) VU Meters (12, 16 & 24 channel models feature Mon #1 & Mon #2 VU meters)

* The VU meters on your console are pre-calibrated at the factory for proper operating levels. The VU meters indicate the relative pre-amp levels being sent to the power amplifiers whether internal or external. It is entirely possible to have very little or no meter operation but be producing an acceptable volume level. Church applications have shown that many times the VU meters will be just barely moving, but a perfectly acceptable volume level is being

reached. This is normal, and the VU meter is simply indicating that you are using very little of the available gain in the console. So, there is little need in this instance for metering. On the other hand, however, a rock band may produce levels that would indicate from -10 to +3 VU. Although it is perfectly normal for high level audio to indicate on the meter up into the "red" +3dB zone, you should be sure to keep the meter reading comfortably at or around "0" VU as a maximum value. Each VU meter has a small screw directly underneath it. This is a mechanical needle adjustment used to set the zero stop of the meter. By turning this screw you are able to properly "Zero" each of the VU meters. This adjustment is pre-set at the factory and should not require any adjustment, however, after a lot of transporting and vibration your VU meters may eventually require a slight adjustment to "Zero" them. It is an easy adjustment to make and can be done by simply using a small screwdriver to turn the adjustment screw until the meter rests at -30 VU.

Electronic Sensitivity Adjustment - There are trim pots located on the System Master P.C. cards. They permit the sensitivity of the meters to be adjusted from a -30dB to a 10dB reading with a 1.4 VAC output. This adjustment will allow you to set your meters to any standard. The factory setting is +4dB at the balanced outputs for a "0" VU indication. **(See "Special Features" section #3 for proper adjustment procedure.)**





GETTING TO KNOW YOUR CONSOLE
REAR PANEL JACKS AND WHAT THEY DO

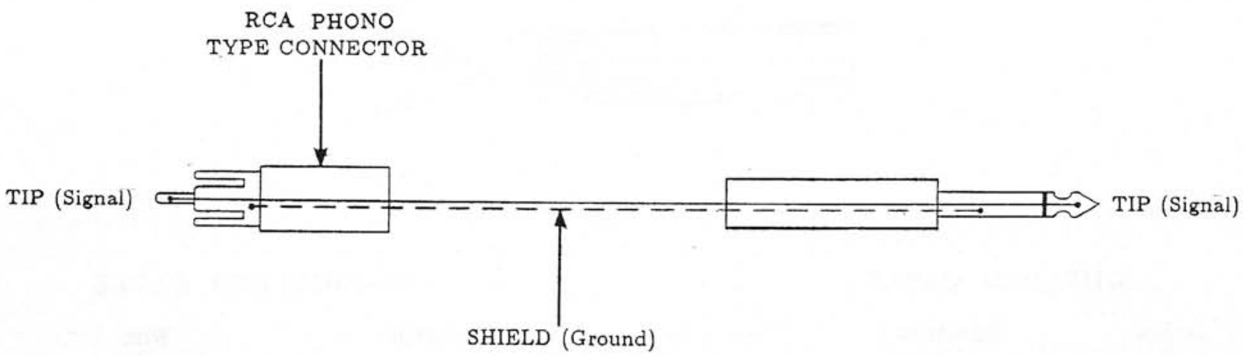
Although the back panel of any console can look rather complex and confusing to a novice, actually many of the "jacks" are redundant and quite simple to understand after a simple explanation. In the next few lines we hope to give you a good understanding of each of the functions of the jacks on the rear panel of your console. Please, don't be afraid to experiment a little with some of the examples. Our experience has shown that the best way to learn how to use each of the rear panel jacks is simple experimentation. If you follow the examples, you should have a lot of fun plugging things into your console.

In some sections there may be terms that you are unfamiliar with. Please refer to the glossary at the end of the manual if you are at all unfamiliar with any term.

Each of the inputs and outputs of the "MX" console are labeled according to the diagram in section 1-2. Simply follow the number to its corresponding jack and read up on the applications and use of that jack. Have fun!

1) <u>Input "Line In Jack"</u>

* This jack is a "Line Input" connection. It is designed to accept signals that are normally too strong for the mic input (See #2 this section) just above it. Line level or (Pre-amp) signals are typically from 200 millivolts to 2 volts. This input will accept signals from any line level source such as the line output of your tape deck, keyboard or instrument amplifiers. This connection wants a "Guitar type 1/4" phone plug," the type of jack found at the end of a guitar cord. Most tape decks use an RCA phono plug connector, and you may need to use an adapter to plug the outputs of your tape deck into this input, however, this is perfectly acceptable.



2) XLR Mic Input

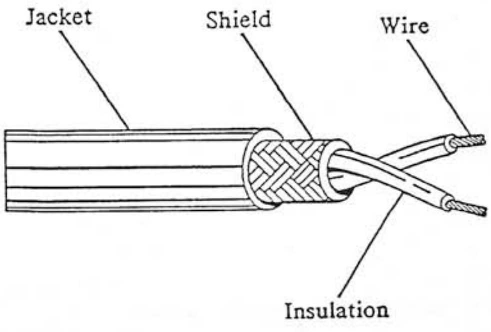
* This is also an input, however, with far greater sensitivity. Line input jacks like to see input voltages of typically 200 millivolts to 2 volts, whereas mic inputs typically want to see 10 to 100 "Millivolts." We also like to call this type of input a "high gain" input. Because this input typically is used with microphones and is designed to amplify very weak signals there are (2) main differences between a mic and line input which is why this connector is so much different than the "Line In" connector.

A) This "XLR" input is designed for low impedance signals. Low impedance simply means a lower resistance to the AC current sent from the microphone. Low impedance mics offer very little resistance to this current which allows for much longer mic cords without loss of signal. This lack of line losses through long cable runs make low impedance mics the best possible choice for use with snakes or mic cable extensions. Low impedance mics generally offer greater signal strengths which result in lower overall noise and higher performance from the mixer. Low impedance mics will offer the best overall response and output for the highest quality performance.

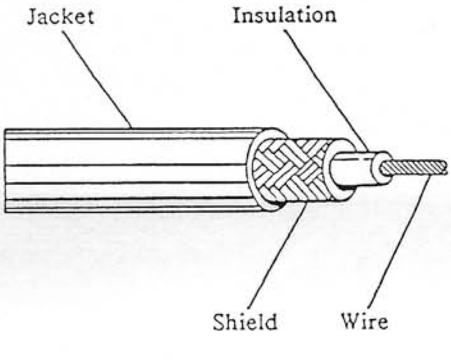
B) The input is "balanced." This term works closely in hand with the term "Common Mode Rejection" for the mic input of your console. Since the signal from a microphone is so low in level, as that signal is transmitted down its mic wire to the "Mic input" of your console, there is a good chance that other voltages "Hum" can be injected into the cable that could be heard when later amplified by your console. What balancing does is eliminate much of this "Stray Field" by canceling it at the input of the console. The act of this happening is called "Common Mode Rejection."

All microphone cables currently sold are "shielded." This means that the signal wire within the mic cord is surrounded by a tightly braided or solid "shield" or ground wire. This is done so that any potential injected signals "stray field" as from light sockets, etc., will be first be passed to ground prior to ever reaching the signal wire at the center of the cord. All mic level or "pre-amp" level cables should be shielded for the lowest possible noise. By this time you should be deducing that a "Balanced, Low Impedance" mic is the best to use with this input. And, with a properly shielded cord, it will provide the best overall gain, lowest noise and highest quality sound.

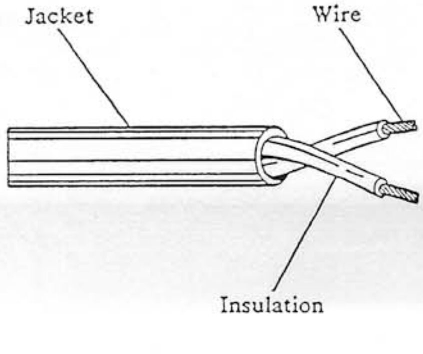
SHIELDED XLR TYPE "BALANCED" CABLE



SHIELDED CABLE



NON-SHIELDED CABLE



3) <u>The Channel Patch</u>

* The channel patch jack allows you access to that channel for inserting different effects or signal processing equipment. Usually this jack is used with such signal processing equipment as compressors, limiters, delays, EQ's, etc. These devices can help with many problem situations requiring special attention. For instance, if you have a vocal input requiring a very precise equalization (tone shaping) you may wish to "patch in" a more elaborate equalizer than the standard tone controls found on the channels. This would allow you the ability to affect that particular channel without affecting adjacent channels. And, you achieve your objective of fine tuning that particular vocal. **See the wiring diagrams for proper cable wiring of this patch point using a stereo phone plug connection. (Under "Special Features of Your Console" section).**

4) <u>Graphic EQ Patch Jacks</u>

* These jacks are also "Patch Jacks." They allow you to use your graphic equalizers for other uses rather than equalization of your console's main outputs. These jacks effectively enable you to break out the graphic equalizers from your console and patch them anywhere you may need them. So, if you wish to use your graphic equalizers for a particular problem channel you may do so using the channels patch point and inserting it into the Graphic Equalizer Patch Jacks. **Remember, these jacks use a <u>"stereo"</u> 1/4" connector. Please see the diagrams listed under "Special Features of Your Console" for making your own cables.**

5) <u>Mon 1 & Mon 2 Output Jacks</u>

* The MON 1 and MON 2 jacks are output jacks. These jacks are not "powered. " This means that they must be connected to a power amplifier in order to properly drive monitor speakers. Because these jacks are not a powered signal, they are said to be a "pre-amp" signal, meaning "Pre" amplifier. (Sometimes called "Line Level"). So, these signals are usually returned to the stage to feed the monitor amplifier inputs, with the monitor amps subsequently driving the monitor speakers. **Remember, with all pre-amp level signals it is best to use a good quality shielded cable.**

6) <u>Effects Send & Effects Return Jacks</u>

* The jacks are both output jacks and input jacks respectively. These jacks are used to drive outboard signal processors such as digital delays, reverb units, chorus effects, etc. The Effects Send jack is used as an output from your console to drive the input of the effect you desire to use. This is a "pre-amp" output and will properly drive the pre-amp or "line input" of and particular effect unit you wish to drive.

The Effects Receive jack is an input to the console and is used in order to receive the output of your effect device. Using the front panel controls of your unit you can vary the output of the Effects Send to drive your effect with as much, or as little signal as required for optimum performance. The Effects Return Jack is designed to receive the effected signal and to have control over the overall amount of the effect you wish to hear in your mix. The maximum loading of this jack to any effect is 22K ohms. This prevents any overloading of your effect device. This Effects Return input also has a front panel control labeled EFF Receive, and will allow you to adjust the amount of effect you wish to hear in the mix. Remember:

Effects Send ....Connects to ....Input of your effect
Effects Receive ....Connects to ....Output of your effect

7) Two-Track Left and Right Outputs

* These two jacks are "pre-amp" outputs from the 2-Track Master controls. The same mix (Stereo Left & Right) appears at these outputs at a (low pre-amp) level as appear at the speaker A & B outputs. These pre-amp outputs may be used to drive a tape deck for stereo recordings of practice sessions or performances while simultaneously using the speaker outputs to drive your main speakers (On powered mixers). There are two jacks per output. One is balanced and the other is a 1/4" (single ended) output. The preferred choice is always to use the XLR type connector because of better signal levels and lower overall noise. However, if your equipment only has (single ended) inputs just use the 1/4" connectors they will offer excellent "comparable" quality and performance.

The usual use of these jacks is to drive additional power amplifiers should you require them.

Note: *If you should require more speaker power than is available in your unit, you should use these jacks to connect to additional amplifiers (to subsequently drive additional speakers). Although, you may use the power amplifiers in your unit along with any additional power amplifiers, you cannot take the output of one amplifier to drive the input of another to achieve more power. Doing this could cause damage to both amplifiers. You also cannot take the outputs of two separate amplifiers and use them to drive (1) speaker. This also could result in damage to both amps!*

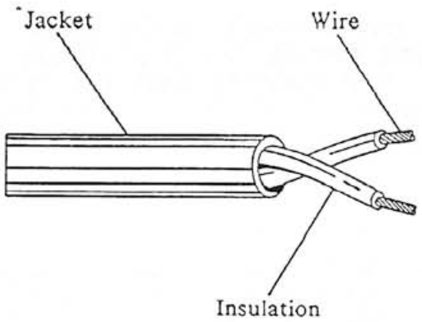
8) Mono Output Connector

* This is a "pre-amp" monaural output. It can also drive tape decks or other amplifiers and it derives its signal from the summing of the Left & Right outputs. Therefore, any signal that appears at the Left & Right outputs will appear at the Mono output only in "Mono." This jack is particularly useful if you are utilizing the speaker "Amp in" patches in order to obtain a powered main and monitor system from the internal power amps. (See Power Amp"In" Connections #10 this section) for description on how toachieve a patch. You could also use the Left & Right (Pre-amp) output jacks to drive your tape deck for a stereo recording while using the Mono output jack to drive your selected power amplifier to power your main speakers. Either way, the Mono output is very useful. Many churches and larger installations will find this jack extremely useful to send signals through telephone lines for remote broadcasts (using an appropriate phone coupler), or as a remote level feed to alternate rooms.

9) Power Amp Spk Connections (Powered Models only)

* These jacks are output jacks used to drive speakers. It is very important to note that on all speaker cords you should use only non-shielded cables. This is the only place in your system that you will use this type of cable. Shielded cables such as guitar cords can cause high performance amplifiers such as the one in your console to oscillate from excessive cable capacitance. Oscillation is a form of electronic feedback and can ultimately damage your amplifier. See the diagram below to properly determine if your cables are shielded or not.

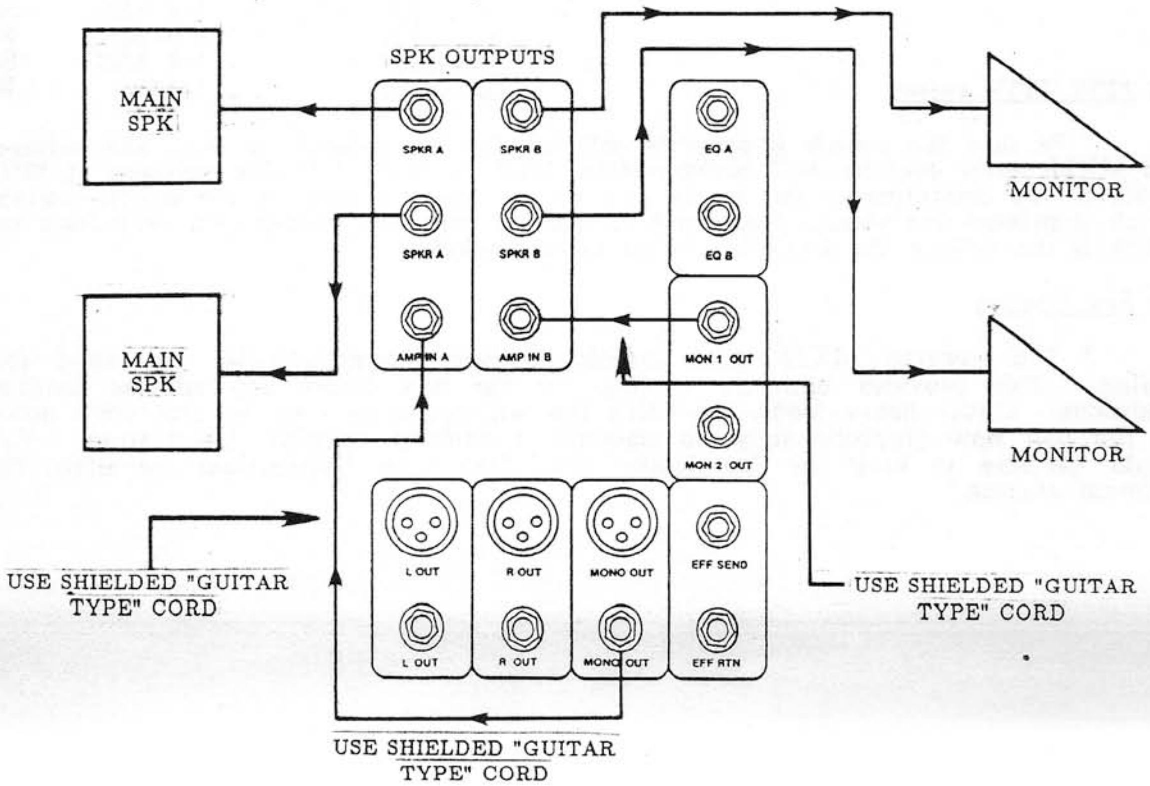
NON-SHIELDED CABLE



For speaker wire lengths up to 100' it is recommended that you use a 16 gauge or larger wire. For speaker runs up to 300' it is recommended that you use a 14 gauge or larger wire. This will provide the best proper delivery of power to your speakers. For speaker run lengths any longer than this we recommend that you call CARVIN for a proper recommendation. Caution: You should always turn your unit off when connecting or disconnecting your speakers.

10) AMP IN "A" & AMP IN "B" JACKS (Powered Models Only)

* These jacks are access inputs to the two power amplifiers you have in your unit. Normally the two power amplifiers in your unit are dedicated to driving your main speakers at the left and right sides of the stage. This consumes the use of both power amplifiers. However, if you need only a "Mono" main speaker system, it will only require (1) amplifier to drive that system. By taking the "Mono" line output (discussed in section #8) and plugging it into this "Amp input," you will use that power amplifier to drive your main speakers. This will then allow you to do a similar patch from the "Monitor #1" output to the remaining amplifier to drive the monitor speakers. The end result is that you are able to (within your mixing console) internally power both your main speakers and your monitor speakers. Of course, you will have less power for your main system. But, these jacks make the power amplifiers in your unit extremely versatile. It allows you to use them for virtually any speaker power requirement you may have by simply plugging any "pre-amp" level signal into your internal power amplifiers to subsequently power speakers.



11) Power On/Off Switch

* This is the main AC power switch for the mixer. It is normal to hear a slight thump on both turning on and turning off your console. This "Turn On Transient" will not harm your speakers. It is always a good idea to have the main volume faders off when turning on the console. This will eliminate any possibility of feedback or excessively loud noises when turning on your console.

12) Power Cord Connector

* A standard "European Style" power connector is provided. This allows you to detach your power cord when transporting your console. Be sure to check you A.C. power source to be sure it is a 110V to 120V, 50 to 60Hz A.C. source. It is an excellent idea to get into a habit of placing your power cord in a place where it will always be transported with the console. This will save you from forgetting it or misplacing it and not being able to later power up your console. However, if you do misplace it you may always order another from CARVIN. They only cost $5.00 plus $2.00 shipping and we'll be ready to rush one to you should you need it.

13) AC Line Fuse

* The A.C. line fuse is a protective safety feature. If your console should have an electrical malfunction, this fuse will protect the console from further damage. You should never attempt to increase the value of this fuse from the value listed on the back of the chassis. To do so could cause damage to your console.

Note: *Be sure if you are ever in need of running your console from the A.C. power produced by a generator that you use a properly "regulated" generator that will eliminate A.C. power surges. Such surges could damage your console and result in blown fuses.*

Fuse values for different models.

FUSE VALUES FOR DIFFERENT MODELS

Model	Fuse Value	Fuse Type
MX-2422	1 ASB	Slo-Blow
MX-1622	1 ASB	Slo-Blow
MX-1222P	5 AGC	Regular
MX-1222	3/4 ASB	Slo-Blow
MX-822P	5 AGC	Regular
MX-822	3/4 ASB	Slo-Blow
MX-622P	5 AGC	Regular

14) 120V/220V switch

* Be sure this switch is properly selected for the appropriate A.C. line voltage. The MX22 series consoles will accommodate 120V or 220V A.C. line voltages at 50Hz or 60Hz. To determine if the switch is properly selected, look at the switch (on the switch is printed the voltage the switch is selected to). The voltage you read from the switch is the voltage the amplifier is set to accommodate.

15) Fan Cooling

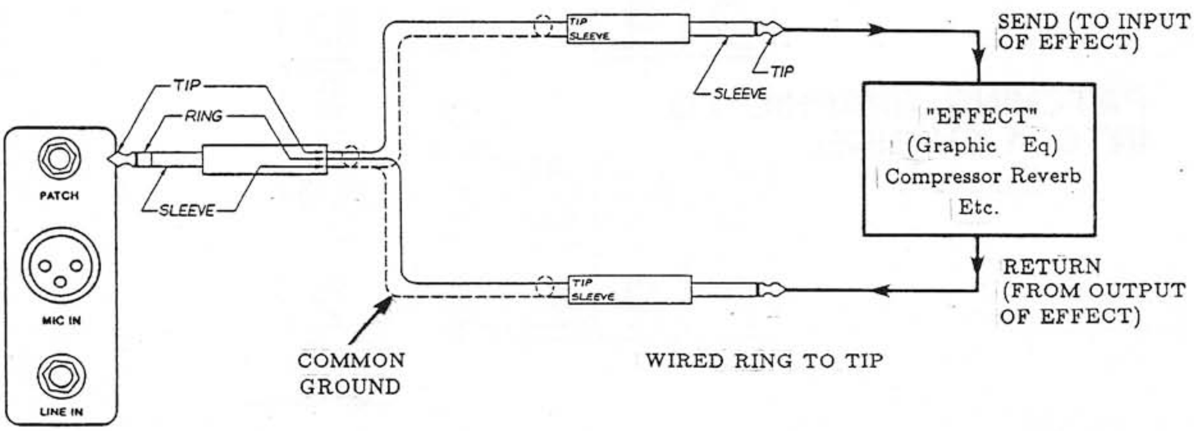
* The powered MX22 series consoles feature thermostatically controlled fan cooling. This provides optimum cooling for the unit under any adverse loading conditions. Under heavy loads, the unit's fan will speed up. As the unit cools down the fan will slow proportionately to maintain a uniform heatsink temperature. You should be sure to keep the fan intake duct free from obstructions to allow for optimum cooling.

SPECIAL FEATURES OF YOUR CONSOLE

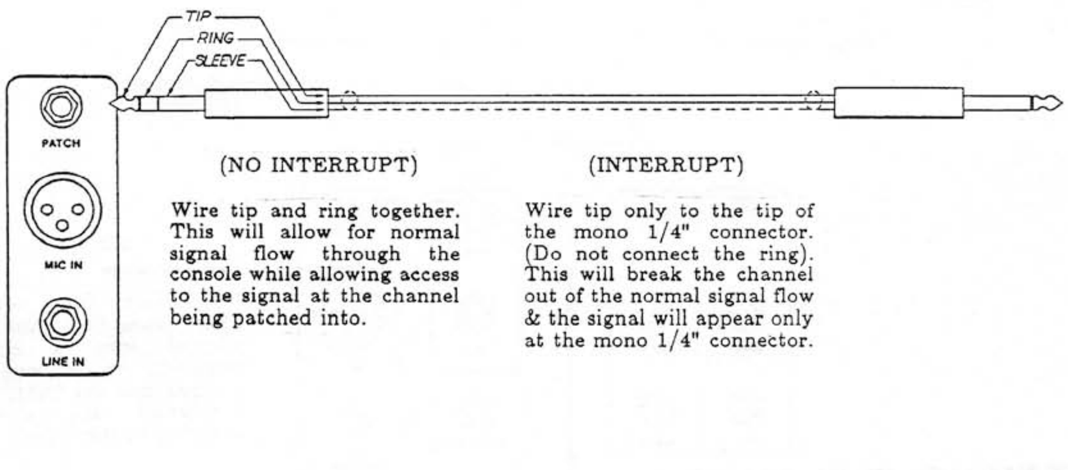
1) Patching Cables

* The channel patch jacks on each of the channels incorporates a "stereo" connection. It requires the use of a stereo 1/4" phone plug and can be wired according to the following diagrams for (Send & Receive), or (Send Only).

PATCHING AN EFFECT INTO A CHANNEL



CHANNEL (OUTPUT) ACCESS CONNECTIONS



(NO INTERRUPT)

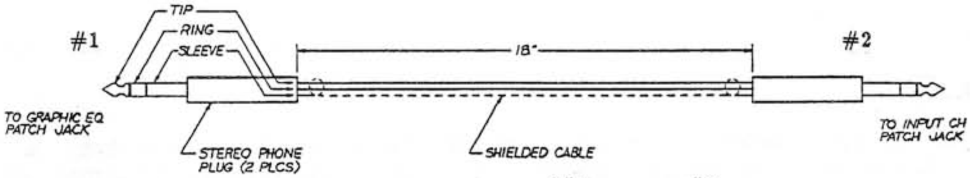
Wire tip and ring together. This will allow for normal signal flow through the console while allowing access to the signal at the channel being patched into.

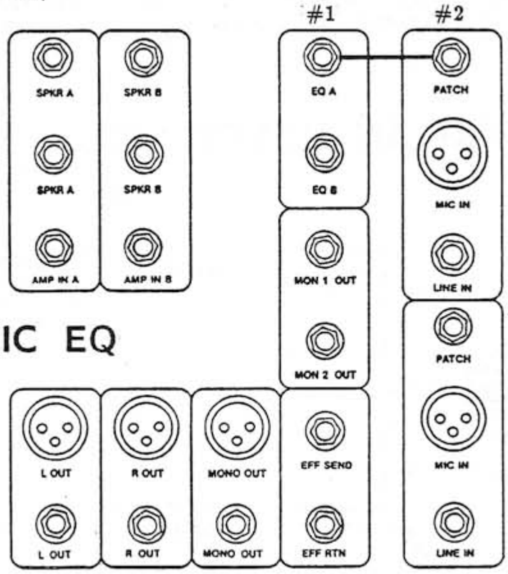
(INTERRUPT)

Wire tip only to the tip of the mono 1/4" connector. (Do not connect the ring). This will break the channel out of the normal signal flow & the signal will appear only at the mono 1/4" connector.

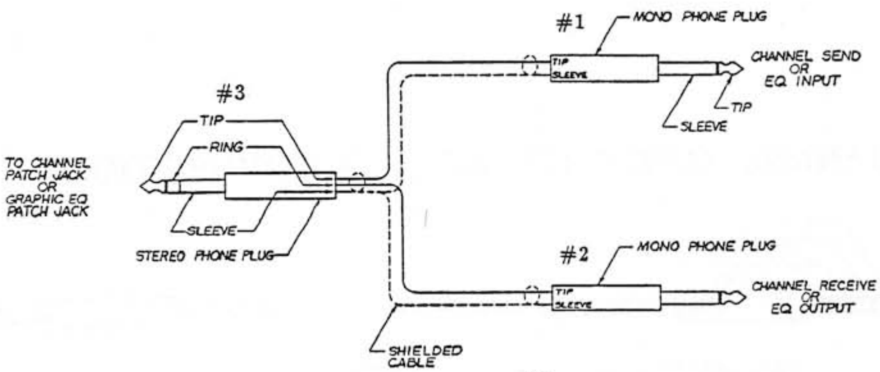
* The Graphic patch jacks also incorporate a stereo 1/4" connector and may be patched into any channel of your console using a shielded cable with a stereo 1/4"

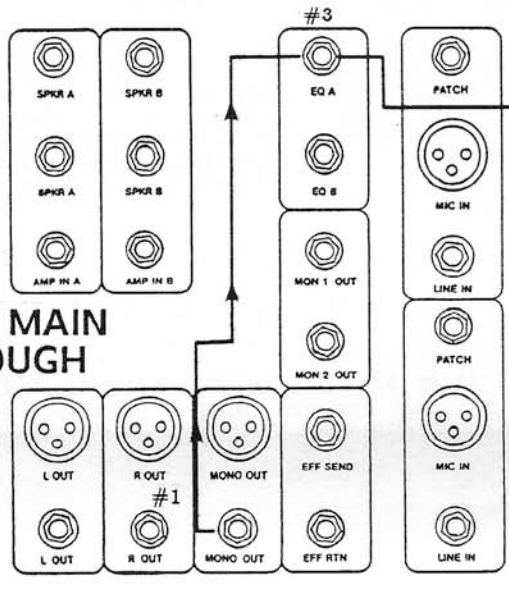
phono plug at each end. However, you can patch your graphic equalizers into any signal chain following the above diagrams for inputs and outputs using the patch connections.





PATCHING GRAPHIC EQ INTO A CHANNEL





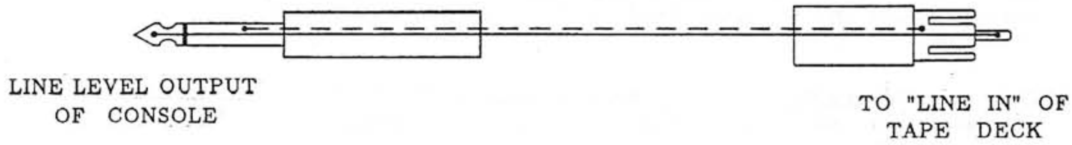
#2

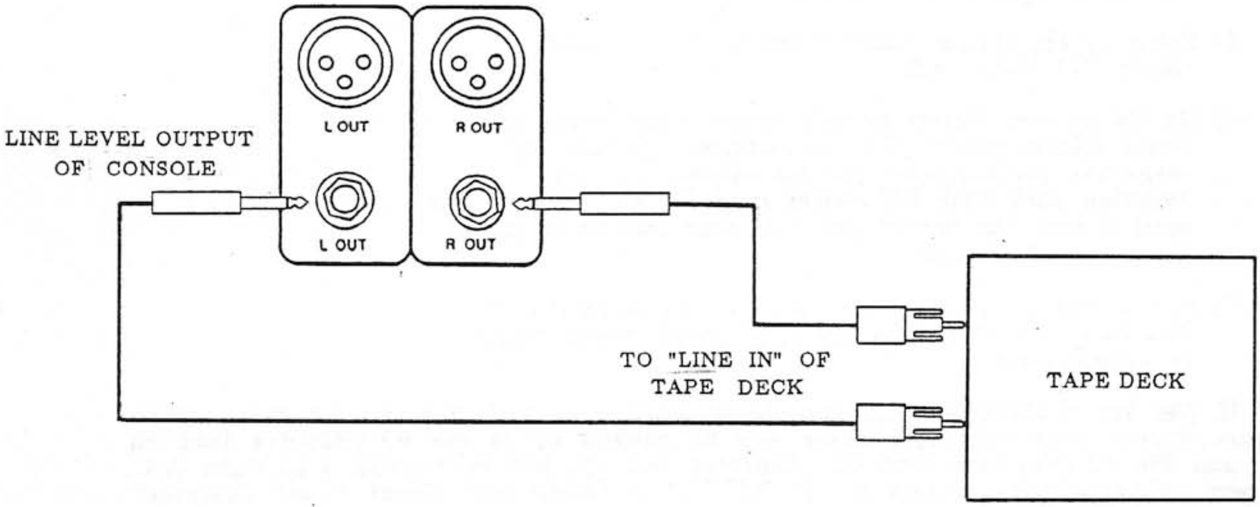
1/4" Phone Plug output. This connection may be plugged into a remote amplifier, or it may be plugged into the "AMP IN" connection for internal powering of this signal.

PATCHING THE MAIN OUTPUTS THROUGH THE GRAPHICS

2) <u>Recording Capabilities</u>

* The main point we want to stress within the scope of this manual is that you may record directly from your mixer to your tape decks. If you are mixing a stereo performance, you will want to take the (2) Track outputs at the rear of the console and plug these outputs into the "Line" inputs of your tape deck. Many times this will require a 1/4" phone plug to RCA phone plug adapter. There are many adapters to accomplish this connection, however, if you are wiring your own cords, the following diagram should be of help in determining the proper wiring configuration.





You may use either the balanced XLR type connection at the rear of your console or the 1/4" connection. The XLR connection produces a +4dB signal whereas the 1/4" connection produces a -2dB signal. This simply means that the relative voltage output of the (+4dB) balanced output is higher. If you are using recording equipment that is XLR compatible, it is advised to use compatible (RCA or 1/4" type connectors) you may be best advised to use the 1/4" outputs at the rear of your console.

3) <u>VU Calibration</u>

* The most often requested meter calibration is for a -10dB level. The easiest way to adjust, or re-calibrate your consoles VU meters are as follows:

1) *Remove the bottom panel of your console to obtain access to the system master P.C. board. You will have to remove the (3) screws in the left and right wood side panels as well as the small sheet metal screws on the bottom panel.*

2) *Connect a good quality Volt Meter to the single ended 1/4" phone plug) "Master Outputs" of your console. Set the Volt Meter to read A.C. voltage with the positive lead of the meter connected to the "Tip" of the 1/4" connector and the negative lead to the ground of the 1/4" connector.*

3) *Insert a good quality signal generator with a 400Hz sine wave into one of the input channels of the console.*

4) *Bring up the channel volume of the channel with the generator in it and assign the channel to all the main outputs. Be sure your pan control is centered to feed the signal equally to all outputs.*

5) *Bring up the Master faders until your volt meter reads .316 Volts (A.C.).*

6) *On the System Master printed circuit board there are small potentiometers (VU) adjustment. Adjust the respective potentiometer for the channel you are metering until your VU meters read "0" VU. Note: You must remove the bottom panel of your console to access the output P.C. cards.*

7) *Repeat this adjustment for each of your main outputs. You have just calibrated the 1/4" single ended outputs to -10dB levels.*

If you are calibrating your console to +4dB, you will perform the exact above procedure except that your volt meter will be hooked up to Pin #3 (Positive lead on meter) and Pin #2 (Negative lead of voltmeter) and you will be reading 1.23 Volts A.C. when you calibrate your meters to "0" VU. (See "Additional Front Panel Features" "VU Meter" #3 for diagram.)

4) <u>Speaker Impedance</u>

* On powered mixing consoles it is very important that you do not go below the unit's minimum impedance with your speaker system. What this means is that you cannot connect too many speakers <u>in parallel</u> to your powered mixing console such that it falls below the minimum load impedance your mixing console is rated to handle. On all Carvin powered mixers the minimum recommended impedance is 4 ohms per channel. In order to figure the impedance of your speaker system you should use the following formula:

FORMULA FOR PARALLEL IMPEDANCE

Take the rated impedance of each of your speakers whether it is 8 ohms, 4 ohms, 3 ohms, 16 ohms or whatever. Invert these numbers (i.e. make a fraction out of them). For instance 8 ohms would become 1/8, 4 ohms would become 1/4, etc. Add each of these fractions together and divide the denominator by the numerator. The result will be your load impedance produced by your speaker system. For instance, (2) 8 ohm speakers connected in parallel would provide the following calculation:

1/8 + 1/8 = 2/8
Inverted = 8/2 = 4 ohms total load impedance

Usually whenever you are simply plugging one or more speakers into the back of your unit, you are running this type of parallel connection. And, the above calculation will give you an accurate indication of what impedance your speaker system (in ohms) is loading your amplifier to.

Understanding that your powered mixing console's amplifier does have limits regarding how many speakers you can hook up to it, and that the limit is 4 ohms is insurance that you are operating your mixer properly and that it will be reliable. This information should be all that is required to properly determine the operating parameters of your mixer amp or any other amplifier. If you require additional speakers beyond the minimum load rating of your amplifier you will need additional amplifiers to drive them.

If you have any questions at all regarding how many speakers you can run off your mixer/amp or regular power amplifier, please do not hesitate to call Carvin. We will be more than happy to assist you in determining if you are operating your system properly.

MAINTENANCE

There is very little maintenance required by your unit. The best possible maintenance is preventative. The major causes of breakdown occur from dirt and heat. Vacuuming the front panel of your console regularly will assure that harmful dust and dirt does not accumulate in any of the electronics. A slip cover (manufactured by Carvin) is highly recommended and should be used whenever you are not operating the console. Always keep your cords in a clean and orderly manner. This assures you the most reliable connection and saves embarrassment from intermittencies caused by poorly maintained equipment. Vacuum your speaker cabinets regularly, and keep them wiped off to eliminate any build up of dirt. It is also recommended that you purchase a vinyl slip cover to properly protect your cabinets. And, you should be sure to cover you cabinets whenever you are not using them.

Always keep your mixing console well ventilated when using it. This will provide from proper cooling and will aid immensely in it's reliability.

SETTING UP YOUR SOUND SYSTEM

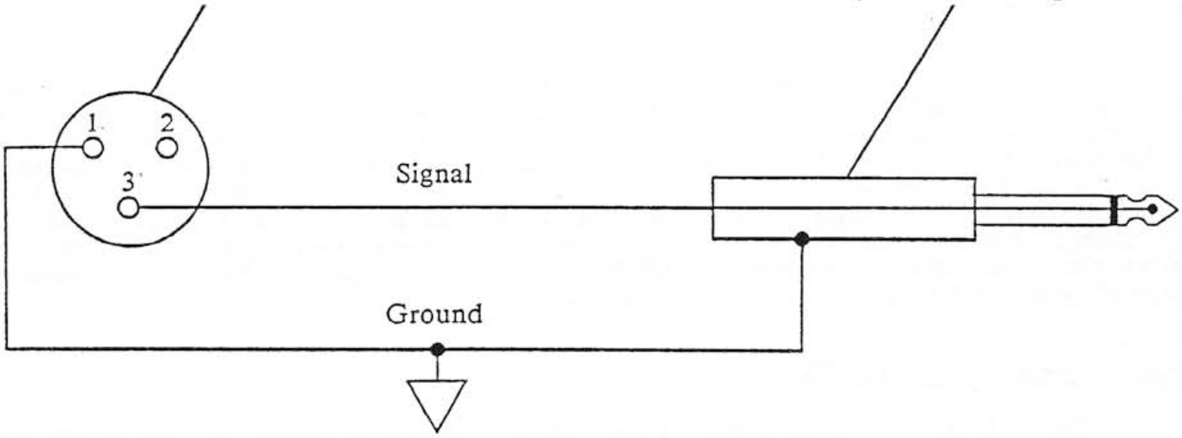
In this section you will be given a brief overview of what connections you will need to make in hooking up a sound system using your mixing console we cover some possible set-up suggestions, and some basics on how to mix live sound. We will even offer some suggestions for hooking up recording equipment and its proper operation. You should find this section both informative as well as enlightening, and we hope you will find this information a "head start" to operating your sound system properly. As always, experimentation is the key to success. Remember, after you have been given the basics and you understand all the controls, how you use them will ultimately expand your creativity as a sound system operator.

Input Connections From The Stage

For live sound reinforcement ("PA" Sound) the input signals to the mixer will come from the microphones and instruments on stage. Each microphone or instrument you wish to be amplified by the "PA" system must be connected to the inputs of the mixing console. It is preferred to have as many instruments as possible plugged into your mixing console. This allows you the best overall control of the volume of each of the instruments as they are amplified by the "PA" system and heard by the audience.

Many times the mixing console will be located a distance from the stage in order that the performance may be monitored and mixed from the audience's perspective. This usually employs a "Snake" cable (available from Carvin). Each of the microphones and instruments are plugged into the snake box at the stage and the snake cable carries all these signals out to the mixer where they then plugged into its inputs. All snake cables are numbered so that you can keep track of which microphones are being plugged into which channels. It is a good idea at this point to label each of the channels according to what instruments they will be controlling. This can be done with masking tape (Scotch brand #230 drafting tape) or another suitable "light" stick tape. This will give you a surface to write on and properly label your channels. It is good to note that all snake cables utilize (3) pin XLR type connectors. Therefore all your instrument and microphone inputs will have to be this type of format. The (XLR) balanced/low impedance format will ensure you the best possible performance and lowest possible noise when operating long cable lengths such as the snake. However, many times you may have a single ended output (1/4" phone plug type) from an instrument that you may need to plug into the snake or directly into the "Mic Input" of your console. This can be accomplished by a high to low impedance adapter (available from Radio Shack or other electronics outlets), due to the versatile capabilities of the Bi-polar differential input circuitry of the MX-22 series consoles, you may special wire a cable where pin #3 connects to the tip of the 1/4" phone plug pin #1 connects to the shield or ground of the 1/4" phone plug and pin #2 is not utilized. See the following diagram:

XLR TO 1/4" PHONE PLUG DIAGRAM



However, prior to performing any of these types of special connections it is recommended that you first consult the manufacturer of the instrument or device you will be making this type of connection to. If you are at all in doubt, we recommend using a low/high impedance adapter (as mentioned) or a "Direct box."

So, once you have made all the input connections to your console, have properly labeled them and have verified that all the connections are good and all mic's are connected properly, the next step is connecting your main amplifiers and speakers.

Connecting Your Main Amps & Speakers

Any of the Carvin "MX-22" consoles can be used for either "Mono" or "Stereo" sound reinforcement. All the mixer formats feature a (Number of Channels) X (Number of Outputs) format. Therefore the MX1222 console for instance is a (12 channel) X (Stereo) X (Mono) format. This means that 12 input channels may be mixed to (2) outputs or "stereo," which subsequently is summed together to feed a mono output. For the sake of simplicity we will show how to hook up a "stereo" system here.

You will be using either the Stereo or Mono Main outputs to drive your power amplifiers. The same snake that was used to feed the signals from the stage to the mixer usually has provisions for sending output signals from the mixer to the stage. You may plug the (Pre-amp level) main outputs from your mixer into the snake cable to send this signal to the power amplifiers left on the stage driving your speakers. Usually the XLR connectors employed on the main outputs of your mixer are used to make this connection.

Important: *With powered mixers (i.e. mixers with built-in power amplifiers) you cannot take the "Speaker Outputs" and feed these through the snake to subsequently power your speakers on stage. To do this could result in damage to the power amp in the mixing console. Only*

"Pre-amp" signals can be returned to the stage through the snake.

Once the signal has reached the stage via the snake cable, or alternate means of cabling, these connections are then made to the power amplifiers. The power amp outputs are then connected to the main loudspeakers using heavy gauge (16 AWG or heavier "Non-Shielded" wire).

Note: *Your speaker cables are the only cables that should not be shielded. All other cables in your system Mic" and "Pre-amp" level cables should be shielded. To have shielded cables connected to the outputs of your power amplifiers could result in damage to your amps.*

Since you cannot send speaker level signals ("Amplifier outputs" from powered mixing consoles) up the snake you will have to use separate speaker cables from the powered console to the speakers.

Connecting the Monitor Amps and Speakers

In a typical setup for live sound the "MX" series MON #1 & MON #2 monitor (auxiliary) busses will be used to provide monitor mixes for the musicians on stage. The MON #1 & MON #2 output signals will be routed to the stage just like the main output signals. Once the signals are at the stage either by direct connection at the rear of the mixer or via the "Snake" sends, you will plug this signal into the inputs of your "monitor" amplifiers for subsequent powering of your monitor speakers.

Remember: *All monitor sends are pre-amp level and are "non-powered." These signals are used to drive power amplifiers that subsequently drive your monitor speakers. You cannot drive loudspeakers directly from these outputs.*

Connecting Outboard Effects

Although the reverberation system built into the "MX" series mixers (except MX-1688 models) will meet the effects requirements of most users, the mixers are also equipped with an auxiliary (effects) buss for mixing other effects devices. Outboard effects can be controlled and returned to the stereo output (2-Track) mix with control over both level and panning.

Outboard effects devices can be connected to the mixer by connecting the "Eff Send" signal from the mixer to the "Line" input of the effects device and returning the effects output signal back to the mixer's "EFF RTN" jack. The interconnect cables will require 1/4" phone jacks for connection at the rear of the mixer and connectors appropriate to the effects device at the other end.

Note: *If you have need of more effects returns you may use the line input of an open or unused channel. An unused channel will offer the same capabilities as the EFF RTN jack with the added option of equalization on the return effect signal and routing of the effects signal to the monitors. This is a common connection and if you are using more than one effects device you may have to use a channel to return the effected signal.*

Because the effects send and return levels of all "MX" series mixers are variable it is possible to use various effects units with operating ranges of -20dBv to +4dBv.

This includes many guitar effects units ("Stomp Boxes") providing they have acceptable audio quality. The effects unit usually found most useful will be a delay unit; either an analog delay, digital delay, or tape echo delay.

Note: *These are all effects that are normally mixed back in with the direct or dry signal. This "Mix in" type of effect is different from an "In line" effect such as an equalizer, compressor or noise gate. "In line" effects are usually patched into the signal path and will affect the whole signal. "Mix in" effects combine a certain amount of the affected signal with the dry and unaffected signal. For example, you might normally add a little reverb to be mixed with the direct or dry signal from a vocal. Only the "Mix in" type effects are appropriate for use in the effects system of your mixer. All "In line" effects are usually used between the main outputs of the mixer and the power amplifiers, or at the channel patch points.*

Note: *Many of the "Mix in" type effects units will feature a built in control that allows you to mix varying amounts of direct signal with affected or processed" signal. When using the effects send and receive controls you should adjust your effect to provide only the total affected signal to be summed at the console (i.e. rotate the effects's mix entirely to the "wet" or processed signal). This will allow you the best control over the dry signal (sent from the channels directly to the outputs) to be mixed with the affected "Processed" signal summed to the outputs. Doing this will eliminate variations in volume when adjusting either the effects send or receive controls as heard in the main mix.*

Monitoring at the Mixer

The MX series headphone output can be used to allow the sound mixer to solo individual channels, to set up the stage monitor mixes, and to audition either the two-track or mono main outputs. The group of switches just above the phones level control is used to select the feed to the phones. Whenever a solo switch is selected, all the other signals regardless of what has been selected by the group of switches above the phones level control will no longer be heard. Only the sound of the signal source where the solo switch is selected will be heard in the headphones. For instance, if a solo switch is depressed at an input channel, the master solo LED (located just above the phone level control) illuminates to indicate that the solo signal has replaced the normally selected signal as the feed to the phones. When all channel solo switches are released the phones feed will automatically switch back to the signal selected at the switch group. Usually isolating or "closed" type headphones are the best choice for this use because they help block out some of the sound from the main speakers. This allows you to better listen to whatever you may have selected at your console oblivious to the surrounding ambient noise.

For phones monitoring of the the main outputs, turn the phones level all the way down, and plug a pair of stereo headphones into the jack at the front right of the mixer. Be sure your headphones are 8 ohms or greater for proper operation. Depress either the "2 Track" or the "Mono" switch located above the phones level control making sure none of the other switches are depressed. (Depressing the other switches will not harm anything, however, it will not allow you to concentrate on a specific selection.) Now raise the faders for one or two active channels and then raise the Two Track faders. This will raise the volume on the main speaker system. Now raise the Phones Level control for a comfortable volume in the phones. Now you can "solo" channels and hear the solo audio in the phones for the purpose of adjusting input channel equalization, tracking down noisy inputs, etc.

Setting Up the Main Mix

In order to set up the main mix you need to first have the input channels set up properly. An important control on the input channel is the Gain control. This control determines the overall "volume" of the signal from each channel. You should first set the input gain control fully clockwise to its highest setting. If the highest setting on this control results in your "peak" LED flashing you should rotate this control counterclockwise one "click" at a time until the peak light just stops flashing. As a rule the channel peak light should not be flashing if the channels are set up properly. Slight flashes from time to time are ok and indicate that you have probably set up your channels properly, since the LED light 6dB before actual clipping (distortion) occurs. You do not have to worry about brief signal peaks escaping detection because a special peak stretching circuit makes sure even the shortest over-level peaks will result in a strong flash of the LED. If the Gain control is set too low then there may not be enough signal available at the channel fader when you are adjusting the main mix. If this should happen after you have set up the main and monitor mixes, then you will need to raise the channel gain control to get more level. Be careful when you raise a channel gain control during a performance because you will be increasing the volume both at the two track mix and the monitor mix, and you may risk feedback, especially at the monitors!

With the input channel Gain controls set properly you are now ready to set up the main two-track mix. Start with a couple of channel faders at the nominal (0) setting and raise the "2 Track Master" faders to get the desired volume over the main speaker system. You should now hear combined audio from those channels with faders raised. Proceed to adjust the channel faders to create the mix of input signals that you would like. Try to keep the channel faders working in the upper half of their range of travel. The faders of unused input channels should be left down so that they do no contribute noise to the mix. If you are listening to the stereo (2 Track Master) mix then you can use the "Pan" controls to pan the individual channel signals anywhere between far left and far right. If you are mixing to a "mono" output, the pan controls will have no effect except for a slight volume loss at either far left or far right extremes. For mono mixing the channel pan controls are usually set at center. (See #5 "Input Channel Controls" for sub-group mixing using the Pan Control.)

Setting Up the Monitor Mixes

Each input channel of your console has two blue knobs labeled "MON #1" and "MON #2." These controls allow you to adjust the volume of either "MON #1" or "MON #2" at each input channel. They allow you to send two different monitor mixes at levels independent of your main mix. These two mixes (MON #1 & MON #2) are independent of each other and the main two track mix. The overall level of the MON #1 and MON #2 mixes is set by the two master "Monitor #1" and "Monitor #2" master faders at the master fader section of your console.

The monitor send signals at the input channel are "post" equalizer which means the channel equalizer will affect both the monitor and main mix. Also, the monitor controls are "post" the channel patch point which means that any effect you have patched into the channel will affect the monitor mix as well as the main mix. And, the monitor controls are always "pre" fader. This means that any channel volume setting controlled by the channel slider will not affect the monitor control. So, you can set your monitor levels completely independent of your main mix.

The MON #1 and MON #2 mixes can be auditioned in the headphones by depressing the appropriate switch just above the phones level control. Remember that if a solo switch is depressed, the solo audio will always override the signal feeding the phones.

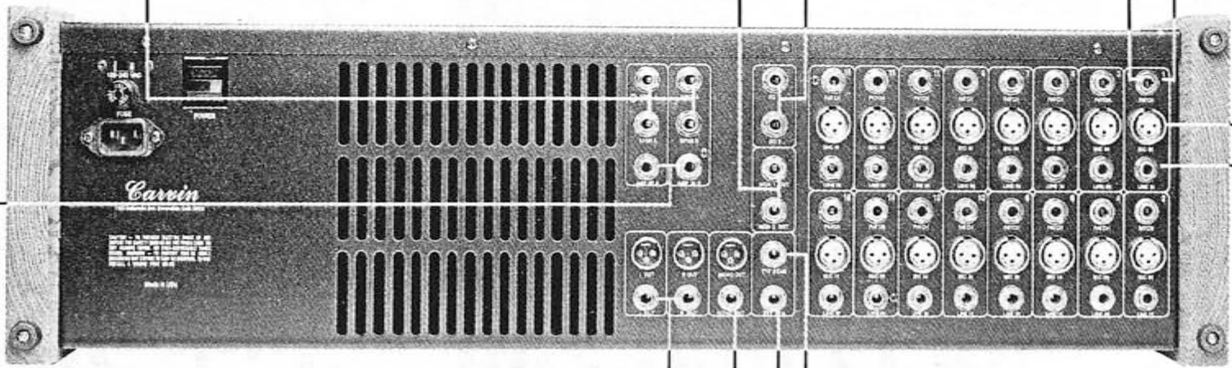
Amp In Jacks are used to access internal (built-in) power amps for amplification of other pre-amp level signals. When a 1/4" phone plug connector is inserted into this jack it will disconnect the normal internal connections driving the A & B amplifiers. The signal inserted at this point will subsequently be amplified. See section #3-15 for additional information regarding this feature.

Speaker Outputs (2 per channel wired in parallel). These outputs are used to drive the main system loudspeakers.

Mon 1 and Mon 2 outputs are "pre-amp" outputs used to drive power amplifiers dedicated to powering stage monitors. This output may be used to send monitor signals through a multi-channel snake to remotely drive power amplifiers at distances up to 200'.

EQ patch jacks (for use in patching the built-in equalizers to other areas of the console). See section #3-13 & #4-2 for additional information on the use of these jacks.

(Channel Patch) see sections #3-13 & #4-1 for additional information on how to use the channel patch jack. This is a stereo 1/4" jack that allows you to loop an effect through a single channel. Usually it is used with compressors, additional equalization, noise gates, etc.



Balanced Microphone Inputs

A & B main outputs are (stereo) outputs used to drive power amplifiers for the main P.A. system. (On powered units, these outputs are internally connected to the internal power amplifiers appearing as SPK-A and SPK-B outputs. These are line level sends and may also be used to drive the line inputs of a stereo recorder for stereo recordings of live performances and may be used to drive line level inputs of alternate equipment at the same time the SPK-A and SPK-B outputs are being used.

The Mono Output may be used as a monaural line level output signal used to drive power amps for a monaural mix. It may also be used as an alternate feed wherever you may need a monaural mix. It may be used to send music to lobbies or (in church applications) as a 600 ohm telephone feed for broadcasting.

The Effects Send & Return is used for looping effects that will be used in varying amounts common to each channel. The effects most commonly used are digital delays, other reverb systems, chorus effects, harmonizers, or effects mainly used to compliment the ambience of the internal reverb system.

(Line Inputs) may accept line level outputs from tape decks or other sources. Will also accept outputs from guitars, keyboards & other instruments.

MX-22 SERIES CONSOLES APPLICATIONS DIAGRAM

BI-AMPING & TRI-AMPING

DIFFERENCES BETWEEN CONVENTIONAL "PASSIVE" AND BI-AMPED/TRI-AMPED SOUND SYSTEMS

We have discussed how to set up a conventional sound system where a full range audio signal passes through one amplifier and feeds a high level crossover within the speaker. This crossover then divides the "lows" from the "highs" and the outputs of the respective signals are fed to the low and high frequency drivers. In "bi-amping", the system utilizes a low level or "active" crossover, receiving the output signal from the pre-amp outputs of your mixing console. Internally, the active crossover divides the signal into its high and low frequency parts. The low frequency outputs of the crossover are then fed to the amplifier that directly drives the low frequency drivers (woofers). The high frequency output of the active crossover similarly feeds the amplifier dedicated to high frequency amplification, which drives the high frequency drivers (horns). Bi-amplification requires the use of a power amplifier dedicated to high frequencies as well as another power amplifier for the low frequency drivers. It will deliver a clean sound with minimal distortion and will more efficiently drive the loudspeakers. Bi-amping will offer better control over the crossover points as well as the relative volume levels of the high and low frequency components.

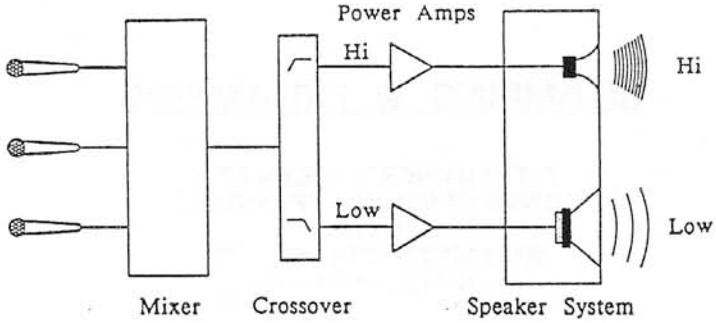
Tri-amping is the same as bi-amping except it utilizes a midrange output within a three-way system. In tri-amping, the output of the mixing console (full range audio signal) is fed to an active crossover that splits the audio into three frequency ranges. The outputs are fed to their respective amplifiers subsequently driving the high, mid, and low frequency drivers. Tri-amping offers exceptional control over the relative levels of each element's volume while offering selectivity for each of the two crossover points. Tri-amping is often used in high-quality high-level sound reinforcement applications. Please see the following block diagram for conventional versus Bi/Tri-Amped sound systems.

BENEFITS OF BI-AMPING & TRI-AMPING

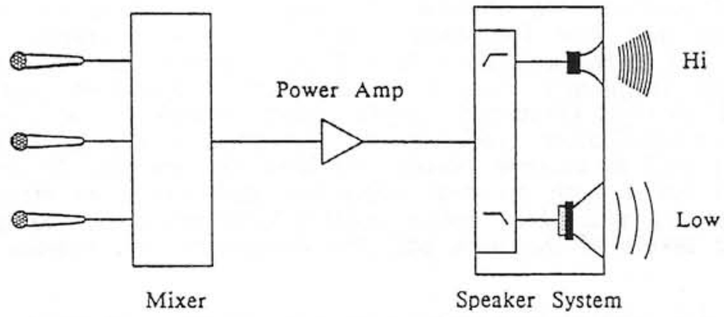
Bi/Tri-Amping provides a great degree of efficiency that is typically lost by a conventional "Passive" crossover. Conventional crossovers utilize inductors, resistors, and capacitors in their design. These electronic devices can affect the output response of the power amplifier or waste much of its available output power. Since Bi-Amping or Tri-Amping circumvents these problems a more efficient delivery of power from the amps to the speakers is achieved. This results in greater efficiency from the sound system. The components of a passive crossover are used "In line" with the outputs of the power amplifier and they affect the way in which the amplifier responds. This interaction can reduce the damping of the amplifier (See damping in Glossary). Bi-amping bypasses the passive crossover and offers a more direct output from the amplifier effectively improving the damping performance of the system.

Bi-Amping & Tri-Amping also provides real power output "Headroom" advantages. Higher frequencies tend to "Ride" on top of the higher energy low frequencies being amplified. As the output of the amplifier begins reaching its total output power capacity these high frequencies may begin to reach the "Peak output" of the amplifier before the low frequency material. This effectively clips the high frequency material. Since the human ear is very sensitive to high frequency distortion this type of "High frequency" clipping is very noticeable. By dividing the high and low frequency material prior to amplification by the systems power amplifiers this headroom problem

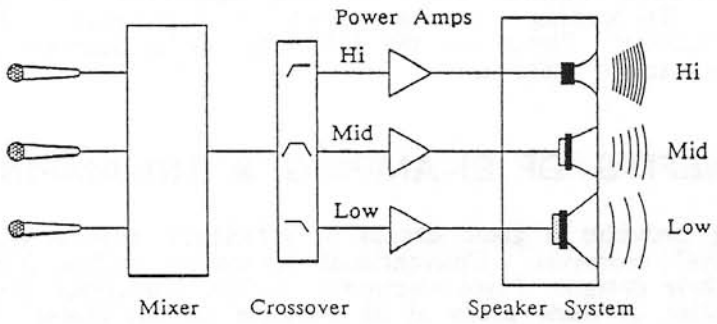
BI-AMPLIFIED SYSTEM WITH ACTIVE CROSSOVER (2 WAY)



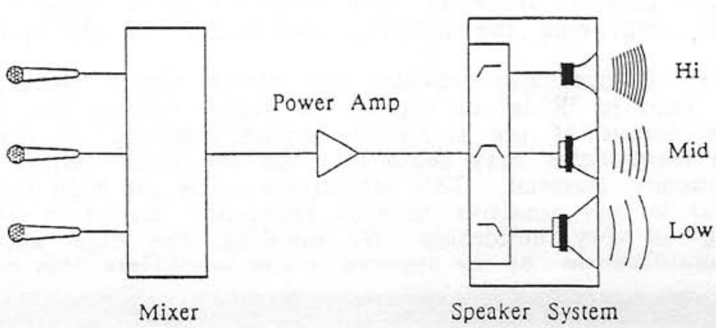
CONVENTIONAL 2-WAY LOUDSPEAKER SYSTEM WITH PASSIVE CROSSOVER SYSTEM



TRI-AMPLIFIED SYSTEM WITH ACTIVE CROSSOVER (3 WAY)



CONVENTIONAL 3-WAY LOUDSPEAKER SYSTEM WITH PASSIVE CROSSOVER SYSTEM



is minimized. Bi-Amping allows for more low frequency headroom and far greater high frequency headroom (when compared to passive systems), and offers increased output with less overall distortion at high volume levels.

WHEN TO UTILIZE A BI-AMP SYSTEM

Bi-Amped or Tri-Amped sound systems are best targeted towards high level sound reinforcement. The increased efficiency and headroom of these types of systems allow for higher volume levels, greater speaker protection and less overall cost in multiple speaker/amplifier systems. Because of the greater efficiency and protection of these types of systems a Bi-amped or Tri-Amped system will provide a higher degree of reliability for demanding "High volume" sound reinforcement. Bi-Amping or Tri-Amping is the choice among professionals and sound companies where continuous high level sound reinforcement is required. It provides greater simplicity in set-up and affords more control over the response of the sound system.

If high level sound reinforcement is not needed the best choice would be a passive crossover network system. In small to medium sized rooms, clubs or auditoriums you may not require the gain and output of a Bi-Amped or Tri-Amped sound system. Passive sound systems are overall less expensive and sound excellent for these types of applications. Carvin manufactures many speakers that feature a passive crossover (C-1204) that are excellent choices for main speaker systems or monitor systems, and each of these speakers may be optionally Bi-Amped should you require extra output and headroom from your system down the line.

HOW TO USE AN ACTIVE CROSSOVER IN BI/TRI-AMPED SOUND

Understanding the optimum operating frequency ranges and power handling capacities of the loudspeaker drivers is essential to properly setting up the active crossover in a sound system. This information will allow the system operator to select the right crossover points in order to maximize the response of the system utilizing those drivers. For example, a high frequency driver may indicate its optimum response range is from 1500Hz to 20Khz. Its power handling capacity within that range may be 30 watts. The best choice for a crossover frequency would then be 1500Hz and above. To choose a lower crossover frequency could cause excessive voice coil excursion and subsequent damage to the driver. Note: For every octave a drivers frequency range is lowered its power handling capacity is divided by four. So, dropping the crossover frequency to 750Hz would allow the above driver only appx 7.5 watts of power handling capacity. However, a higher crossover point could allow for increased power handling capacity and less distortion. If a higher crossover frequency was then desired (Say 2K or 2.5Khz) you would want to choose a woofer (Or mid-range) speaker to compliment this crossover frequency. So, you would look for a low or mid-frequency driver that maintained a liner response out to the desired crossover frequency.

Good quality drivers are very important in achieving a "Flat" or accurate responding system. Any "dips" or inconsistencies in the response of drivers within a sound system will reduce the accuracy of the sound systems reproduction of sound. Select drivers that have a smooth response throughout the range where they will be used. This will assure the best possible driver response from the sound system. Equally important is the active crossover and its overall capabilities and performance.

The Carvin XC-1000 crossover is an excellent representation of a quality crossover. It features 18dB/octave Butterworth filters that sum accurately. It offers sweepable parametric selection of crossover frequencies. Its high efficiency roll-off offers maximum protection to the high frequency drivers. As with any professional audio product, its inputs and outputs are balanced (and will accept high impedence sources). It will interface with any professional audio system, and addresses correct input and output impedences for maximum signal quality and performance. Carvin highly recommends the use of this crossover with any of our professional Bi-amp or Tri-amp systems.

Some of the recommended settings for Bi-Amping and Tri-Amping Carvin's speakers are as follows;

Speaker	Recommended Crossover Freq.
3000E&M	500Hz or lower
1200E&M	2kHz or lower
1330E&M	1500Hz or lower
R-540H&E (Horn)	1.2k or higher
980E&M	1.2k
960E&M	1.2k
850E&M	1.2k

SUGGESTIONS FOR EFFICIENT SET UP AND QUALITY SOUND ***LIVE SOUND REINFORCEMENT***

At this point we would like to make some general comments on setting up and operating a sound system. The most important point to emphasize is that a little planning before the day of the performance can prevent serious problems the night of the performance, especially if you are new to sound reinforcement work.

PREVIEW THE CONCERT HALL

Try to check out the concert hall before hand to determine where you can obtain power and how far you will have to run power extension cords. Is there enough current capacity to ensure that you will not blow all the circuit breakers on the opening note? To answer this question you need to know approximately what the total AC line current requirement of your sound system will be. One of the greatest problems of circuit overloading is "Flood or Spot Lights" plugged into the same circuit. Check to be sure any lighting systems are on a different circuit, or that the circuit they are on will handle the current requirement of your lighting system. You can easily determine the total AC current requirement of the system (that is, the total number of "Amperes" required from the AC line). To do this you should add up the total current requirements of each piece of equipment (in amps) that you will be using in the system. Or, you can add the total "Watts" (power capability) of all the pieces of equipment in the system and divide this number by 120 (the AC line voltage). This will allow you to arrive at the total current draw.

The current capability of the AC power circuits you use should exceed your total sound equipment's use by a healthy margin. As an example, let's say you have determined your total current requirement to be 15 amps. Then, for a good safety margin (25%) you should make sure that the circuits can supply 20 amps. In any event, make sure you power up the complete system well in advance of showtime so if there are any problems you can deal with them and have plenty of time to correct them prior to showtime.

MAKE AN EQUIPMENT LIST

A good way to prepare for the show is to start with a list of all the equipment that you will be using on the job. Later you can use this list as a check list when it comes time to load up for the performance. And, it will make sure that you do not lose pieces of your gear that could be both time consuming and frustrating later. Your list should include everything from the mixer down to the last interconnect cable. If you start with a block sketch of the sound system showing the mixer, snake, main amps, and speakers, monitor amps and monitor speakers, you can then draw in each interconnect cable and label each end as to the type of connector. This diagram will allow you to set up the system in less time because you will not have to stop to think what connects to what! It's no fun scrambling to build interconnect cables when showtime is a few minutes away! A complete equipment list and system diagram can help prevent equipment loss and help prevent such last minute emergencies!

HAVE A GOOD TOOL KIT

Put together a good tool kit and add it to your equipment list. Make sure you tool kit includes a generous assortment of connector adapters and enough spare connectors to repair each type of interconnect cable used with your system. Preventive maintenance is always the best way to assure the least amount of equipment related problems. Keep your cables in clean and ready state of repair at all times. And, be sure to clean any dirt or dust off your mixer, cables, and speakers. Such preventive maintenance will provide you with a much more reliable system. A good basic tool kit might carry the following items;

1) Spare cables
2) Spare connectors or adapters
3) Pliers & wrenches
4) Screwdrivers appropriate for any need
(Straight & Phillips - Small & Large)
5) Soldering iron/solder/solderwick

Of course, your tool kit can be as elaborate as you desire according to your technical ability. It is also a good idea to carry a flashlight or accessory lamp for those occasions where the house lights go down and you are left in total darkness groping for the faders!

POWER UP SEQUENCE

There is really just one rule for good practice regarding the sequence in which to power up and down the system. That rule is: "Turn the loudspeaker amplifiers on last and off first." This will prevent any excessive transients from getting to your loudspeakers and possibly damaging your horn drivers.

Besides the concern over damaging the loudspeakers, it is not a good idea to expose your audience to obnoxious pops, squeals or thumps if you want them to come back again. Make sure your audience hears only music from your sound system!

SOUND SYSTEM CHECK

After the sound system is connected and powered up, you will want to check each microphone and instrument connection to the mixer one line at a time. Have an assistant speak into each microphone so that you can confirm that it feeds the correct channel of the mixer and that the channel is properly identified on the writing strip. If your audience is seated at this time you can spare them from listening to this test by switching off the main speaker amps, (or, unplugging the speaker outputs on powered mixers) and monitoring the sound check over the headphones. At the same time, you can verify the stage monitor system by raising the monitor sends at each channel and then carefully raising the monitor output faders. Have you assistant verify stage monitor sound as he checks each mic and instrument. Finally, check the main system from one of the mics on stage. This completes the basic system check out.

GLOSSARY
OF AUDIO TERMS

"A"

AC (Alternating current)
An abreviation for the term "Alternating current." Alternating current is a current that periodically changes its polarity. U.S. standard is 60 times per/second or 60Hz. Some countries operate on 50Hz (or 50 times per/second).

ACOUSTIC ISOLATION
Any manner of device that prevents sound in one area from leaking into another area. Sound proof walls provide acoustic isolation to prevent sounds from one area being heard (leaking) in other areas. Baffles, sometimes called "Acoustic Flats" are another means of providing acoustic isolation. These are a "Flat" of sound deadening material that is typically placed between musicians in a studio to reduce bleed through of adjacent instruments into nearby mics.

ACTIVE FILTER (ACTIVE EQUALIZER)
This type of filter is characterized by its amplification qualities and ability to boost or cut a signal at defined frequency centers. This type of "EQ" utilizes electronics to alter the frequency response characteristics of audio or other electronic signals, without imposed losses on the signal. Passive equalizers (on the other hand) introduce losses because they do not contain amplification circuitry.

AMBIENCE
Refers to the reflective characteristics of a particular room. A room that contains many multiple reflections of sound waves from the walls, ceiling, floor, etc. are said to have much ambience. Rooms that are said to be "dead" lack ambience. Ambience therefore best describes the particular reflective qualities of a room.

AMPLIFIER
Refers to an electronic device used to increase the strength or level of a signal. It may increase (amplify) the voltage or current of a signal, and many times an amplifier may be used to isolate and control a signals level. Automated consoles and synthesizers may use voltage controlled amplifiers to automatically control signal levels. When an amplifier is used to isolate two parts of an electronic circuit to minimize any interraction it is said to be an isolation amplifier, or "Buffer amplifier." Power amplifiers are amplifiers that are designed to drive loudspeakers.

AMPLITUDE
This terms usually refers to the measurement from the crest to the trough of a waveform. May be associated with the relative measured level or "volume" of an electrical or acoustical signal.

ANALOG
A term which refers to an electrical signal whose characteristics vary in direct relationship with the acoustic waveforms. It may refer to level changes to produce asignal in direct proportion to the original signal.

ARTICULATION

Refers to a quantitative ability to understand spoken words. Percent articulation is a quantitative measure of the degree to which this understanding occurs. For the typical sound reinforcement system no more than 15% articulation loss is acceptable.

ATTENUATE

To reduce the strength of a signal. Sometimes refered to as reducing the volume level or loudness. May also refer to reducing the acoustic sound level or loudness through the use of sound absorption materials.

AUDIO FREQUENCY

The range of human perception of sound. Approximately 20Hz to 20kHz. (Hz = cycles per second).

AUX (Auxiliary Input)

Usually a "Line Level" input on a mixer, amplifier, or tape machine designed to accept a variety of line level sources, such as outputs of tape machines, signal processing equipment, amplifier line outputs, etc. Non-used inputs on consoles are sometimes used as "Aux" inputs accepting effects outputs or similiar line returns.

"B"

BAFFLE

A sound absorbing partition used in studios to provide sound isolation between different instruments.

BAFFLE (speaker)

Refered to as the board or surface that a speakers are mounted to within a speaker cabinet. It is used as an obstruction to the transmission of sound waves between the front and rear of loudspeakers within the cabinet ultimately allowing for best possible efficiency of bass frequency development.

BALANCED

Refers to an electronic circuit utilizing (3) wires. Two of the wires carry the signal at exactly equal and opposite polarities. One wire carries the signal at a (+) potential and the other carries the exact signal an equal and opposite (-) potential. The third wire is a "shield" or ground wire. Because the two signals are at equal potentials with respect to "ground" they are said to be "Balanced" with respect to ground. Balanced inputs in mixing consoles help minimize extraneous noise picked up by long cable runs. Any extraneous signals potentially injected on the wire will appear at equal potentials with respect to each other. When summed at the input of the console these signals will effectively cancel each other. This helps provide the lowest possible noise. Balancing is especially effective when using long cable runs.

BANDWIDTH

Refers to the range of frequencies within a frequency range through which signals can be transmitted. Bandwidth's will have an upper and lower limit in between which frequencies may pass. If a bandwidth is variable (ie. its range may be narrowed or widened) it said to have a vairable "Q" as with certain parametric equalizers. (See Parametric "Equalizer")

BASS

A defined frequency range (termed low frequency range) normally considered below 500Hz.

BIAMPLIFICATION

Biamplified sound systems utilize a low level "Active" crossover to divide the full range audio signals into their high frequency and low frequency parts subsequently feeding two separate power amplifiers that ultimately drive the high and low frequency loudspeakers of the sound system. The high frequency amplifiers drive the horns and high frequency drivers, and the low frequency amplifier drives the low frequency drivers or woofers.

In a conventional system the "Full range" audio is amplified through a single amplifier. This amplified signal then feeds a high level crossover (usually located on the loudspeaker enclosure) that divides the signal to the high and low frequency drivers within the enclosure. This type of system is sometimes called a "Passive" type system.

BITE

A term usually used to describe the attack or percussiveness of a particular instrument. A distorted guitar played with percussiveness and a considerable amount of edge may be said to have considerable "Bite." (ie. very loud with sharp staccato attacks).

BLOCK DIAGRAM

An overview of the overall layout of a particular electronic system. It usually does not show all the specific details, and shows basic "Schematic like" drawings illustrating the main parts of a circuit, showing overall signal flows but lacking smaller details of the circuit.

BOOM, (Microphone)

A pole-like attachment usually mounted with a swivel or pivot point to a mic stand allowing for horizontal as well as vertical extension beyond the normal capabilities of the stand. It allows for easier placement of microphones closer to the performers or instruments.

BOOMY

Sometimes refered to as "Tubby" sounding. It is a condition where excessive bass response exists, or a peak in the bass range exists, causing the sound system to sound unnaturally bass heavy. Characterized by a "Thumpy," or "Boomy" sound.

BREATHING

A situation in which too much attack or too much compression is used on program material such that background noise becomes louder in the absence of strong signals and becomes softer when a strong signal is present. It is an undesireable phenomenon and can be rectified by proper adjustment of either the attack, release, or levels of compression, using only the appropriate amounts of compression required.

BRIDGING

A situation in which the input impedence of an audio circuit is at least 10 times the output impedence of the signal source. Such a circuit imposes minimal loading to the signal source (greatly reducing the power taken from that circuit).

Bridging also refers to a technique where a stereo power amplifier may be used as a higher powered "Mono" amplifier. This is achieved by driving the inputs of the stereo power amp with the same signal but reversed in polarity. The amplified signal is then derived by using the (+) terminal of each channels output. This may cause damage to some amplifiers, however, all Carvin's DCA-800 & DCA-300 amplifiers are capable of this function. It is always recommended to consult you manual.

BRIGHT

A term for excessive amounts of high frequency intensity or energy in a sound reinforcement system, recording, or playback system. Sound which is too bright is considered to be shrill.

BUSS

A "Buss" within a console describes a circuit board trace or signal wire where a number of inputs may be connected, feeding one or more outputs. Switches or controls may regulate the assignment, or intensity at which a signal is summed to a particular buss, and the output of the buss usually feeds an output summing amplifier that subsequently drives one of the consoles main output connections. Bussing signals within a console provides a means of summing multiple source signals, combining (mixing) them and routing them to various outputs of the console.

BYPASS

Refers to the act of routing a signal around a given circuit. Bypasses may be accomplished by use of relays, switches, or hard wiring to allow routing of a signal directly from an intput to any output of a circuit. Sometimes called an "In/Out" switch.

"C"

CAPACITOR

A device consisting of two or more conducting plates separated from one another by an insulating material and used for storing an electric charge. Capacitors will pass A.C. currents, but will block D.C. They are also used to store voltages.

CENTER FREQUENCY

Describes a focus point at which the greatest amount of boost, or attenuation occurs in a peak or notch type filter. (ie. The top of the crest of a peak filter when boosting a signal, or the trough of the waveform when cutting (notching) a signal).

CHANNEL SEPARATION

Describes a quantitative measurement of the electrical isolation between signals picked up from adjacent channels. It is usually measured in dB. Good channel separation implies very little crosstalk or leakage of signals from a channel to an adjacent channel.

CLIPPING

The signal level at which an amplifiers capacity is exceeded and distortion occurs.

CLOSE MICING

A situation where microphones are placed very close to the instrument or source signal. This allows for greater intensity of the source signal and less leakage of sound from nearby sources. Close micing provides a great deal of presence and detail from sound sources significantly reducing leakage in ambient environments.

COAXIAL SPEAKER
A low frequency driver with a concentrically mounted high frequency driver integral to its construction, and possibly sharing the same magnet structure.

COMPLIMENTARY AMPLIFIER
A "Push-Pull" design amplifier utilizing both PNP and NPN or similiar "complimentary" devices in its output and driver circuitry. The amplifier uses the complimentary devices in both positive and symetrical negative amplification of signals.

COMPRESSION
A process whereby signal levels are limited within a particular intensity range. Signal levels of program material considered too strong are reduced and signals very weak are amplified and made louder.

COMPRESSION DRIVER
Refers to a high frequency transducer designed specifically to be coupled to a horn for maximum efficiency instead of directly radiating into the environment.

COMPRESSOR
A compressor is a signal processing device (amplifier) that attenuates the level of an input signal, effectively reducing the dynamic range of the signal. It may be set to attenuate signals at a particular set range, or it may be used to operate on signals above or below a set threshold.

CRISP
A descriptive word used to indicate the edge or bite of a recording, playback, or sound reinforcement system. The crispness of a particular sound also relates to its accuracy and intensity of high frequency energy. It may be desireable to have a crisp sound whereas too crisp may appear shrill and undesireable.

CROSSOVER FREQUENCY
The point at which a frequency split is made. The frequency point where the division between the low frequency material below the crossover frequency, and high frequency material above the crossover frequency is sent to the low and high frequency amplifiers or drivers of the sound system.

CROSSOVER NETWORK
An electronic unit (Passive or Active) that is designed to divide a full range audio signal into two or more frequency bands.

A low level (sometimes called active) crossover is designed to operate at line levels and usually has built-in amplifiers to avoid any signal losses. They are placed before the power amplifiers and are used in biamped or triamped sound reinforcement applications.

A high level (sometimes called Passive) crossover system is designed to operate on speaker level signals and are connected between the speakers and the outputs of a power amplifier. These types of crossovers impose certain losses in achieving the required frequency separation, and are the type of crossovers normally built into a speaker system.

CROSSTALK
Describes the leakage of undesired signal from one channel into an adjacent channel. The quantitative measurement of this spec. is usually described in dB. (See channel separation).

CUE BUSS

A term used to describe channel or sub-group routings in a mixing console to feed program material to performers on stage or to headphone feeds in a studio. Sometimes called a Foldback Buss.

CUE CIRCUIT (foldback circuit)

Cue circuits are designed to route program material for appropriate auditioning by either a performer, engineer, or for remote communications. In mixing consoles there are several Cue circuits that enable the operator to audition various inputs, and outputs as well as remotely feed signals to required areas.

The Channel Cue - allows the operator to audition a particualr channel prior to the channel fader in order to audition material without feeding program material to the outputs (channel fader down) for channel identifications and system check-outs.

Talkback & Communications Cue - describes a built-in signaling system utilizing amplifiers, microphones, signaling lights & tones for communicating to the stage, backstage, to lighting personnel, or other personnel. Not used for actual program audio.

Performer Cue - may be a stereo or monaural feed to audition performers on stage. In a recording studio this feed may drive performers headphones in order to monitor themselves and other musicians as well as listen to previously recorded tracks for overdubbing.

CUT SWITCH

Sometimes called a "Mute" switch. On a mixing console, when this switch is depressed any signal normally appearing at the input will be cut off so that it no longer feeds the mix. It allows channels to be cut off without adjusting fader settings allowing the operator to audition other channels eliminating the channel where the "Cut Switch" has been depressed.

CYCLES PER SECOND

The term Cycles Per Second (CPS) has been replaced by Hertz (Hz). It refers to the frequency of a periodic waveform. Usually concerned with defining a particular value of a frequency within a full range audio frequency spectrum. (ie. 400 CPS or Hz refers to a frequency at 400 cycles per second and is considered a mid-bass sounding frequency.)

"D"

DAMPING FACTOR

The ability of a loudspeaker to resist unwanted cone movements (usually at lower frequencies) is "Damping Factor." The damping factor of a loudspeaker is calculated by dividing the speakers rated impedance by the amplifiers source impedance.

dB (Decibel)

The dB is a unit of measurement relating a numerical expression of relative levels (at a logarithmic ratio) between an absolute reference and a measured signal. The dB is only meaningful when referenced to some absolute or actual value. (ie. dBv reference voltage is .775 volts, dBV references 1 volt, dBm is referenced to 0dBm or .775 volts across a 600 ohm impedence and is a measurement of power etc.).

D.C. (Direct Current)

An electrical current flowing in one direction, of uniform polarity as opposed to A.C, (Alternating current) which regularly changes its direction (60Hz) and polarity.

D.C. PROTECTION

D.C. protection is usually associated with the output of a power amplifier. Power amplifiers with no output transformers have the potential of delivering D.C. voltages to speakers. Such voltages will rapidly heat a speaker's voice coil potentially damaging it. To avoid this potential problem many amplifiers contain special circuitry, relays, capacitive networks, etc. This circuity is commonly called D.C. protection. Some passive crossovers also contain in line capacitors to help protect horn drivers from high level D.C.

DELAY LINE

A signal processing device that is used to time delay an entire audio spectrum by a certain length of time. Most delays today are capable of delaying signals from 0 to 2 seconds. Common uses of delays are for time aligning concert halls in sound reinforcement, or for echo, flanging, chorusing or other special effects.

DIAPHRAM

The moving surface of a loudspeaker transducer or horn that couples the mechanical motion of the device with the air to project sound. Diaphrams are usually used to describe the part of a horn that couples to the air, while a "Cone" is normally used to describe the coupling surface of a loudspeaker (woofer).

DIGITAL DELAY

Digital delay describes the manner in which a particular signal is delayed. In a digital delay an input signal (analog) is converted to a digital (binary) number format. The binary numers are stored for a period of time preset by the operator and are later recalled and converted back to analog format. The period of time the binary numbers are stored determines the time length of the delay. Digital delays typically have far better frequency response and signal to noise ratios than analog type delay units.

DIRECT BOX

A device that converts high impedance signals to low impedance signals. Usually a Direct Box will accept single ended high impedance sources (such as line level instruments, outputs of guitars and basses, etc.) and will convert them to balanced low impedance sources for subsequent feed to a remotely operated mixing console. Many direct boxes are active for little or no signal loss in performing the conversion. They allow for long cable sends of high impedance signals, and usually balance the signals for lowest possible noise and best gain.

DIRECT FEED

Refers to routing an audio signal via the most direct route to its intended destination. Taking a mixing consoles direct output from a channel and bypassing the output summation circuitry and connecting the signal to an input of a multi-track recorder would constitute the most Direct Feed of the audio program material to its intended destination.

DIRTY

Refers to a term describing excessive audible distortion in a recording, playback, or live sound reinforcement system.

DISTANT MICING
Describes a technique where one or more microphones are placed at a distance from there intended pickup source. Usually used for wide coverage or an "Overall Effect" covering a greater pickup area. (See also "Close Micing")

DISTORTION
Modification of an original inputed signal causing frequencies to appear at the output that are unwanted and did not appear in the original inputed signal.

DRY
Refers to the lack of ambience in a track, or program. Ambience may be reverb, echo or combinations of both.

DULL
A term describing a relative high frequency deficiency in a sound system. A recording, playback, or sound reinforcement system that lacks high frequency response is said to be "Dull" sounding when compared to a system with adequate high frequency reproduction.

"E"

ECHO BUSS
Sometimes called an effects buss. It is an auxiliary buss that is normally used to feed program material to signal processing gear, reverberation chambers or the like.

ECHO RETURN
This term describes an auxiliary input that is normally used to return effect signals to a mixing console. It is sometimes called an effects return or auxiliary return. On a console it is usually a line level input used for returning outputs of reverb, delay or auxiliary signals. Most consoles have provisions for level adjustments and assigning of the returned signal.

ECHO SEND
This term is sometimes called an effects send. On a mixing console the "Echo Send" is a level controllable output designed to send program material to signal processing gear. It is used to drive inputs of effects devices such as reverb systems, delay, and other signal processing equipment.

EFFECTS BUSS
On a mixing console the effects buss is identical in operation to the echo buss and is usually dedicated to driving special signal processing equipment. It may be used to drive reverb or delay processing equipment, however, typical signal processing "Effects" devices usually refers to flangers, chorus, harmonizers, gates, and the like.

EFFECTS RETURN
A line level, volume controllable and assignable input on a mixing console used to return effects signals. Identical in operation to an "Echo Return."

EFFECTS SEND

A line level output feed of program material from a mixing console designed to drive the input of an effects device, such as flangers, chorus, harmonizers, etc.

EQUALIZER

An electronic circuit or device designed to change the frequency response of signals in a pre-determined manner. An equalizer may amplify or attenuate the level of signal at certain areas of the audio frequency spectrum. There are several different types of equalizers mainly classified by their particular characteristics or modes of operation.

FIXED FREQUENCY - Describes an equalizer or equalizer circuit that is dedicated to a particular frequency range within the audio spectrum. It is capable of boosting or cutting the level of only that particular frequency range. A fixed frequency equalizer may have more than one conrtol, and hence may be able to operate on as many frequencies as controls provide for. For instance a (3) band equalizer may control bass, mid and high frequencies. However, the particular frequency that the bass, mid or high level control is pre-set and cannot be changed; thus "Fixed." Sometimes fixed frequency equalizers may be called 2,3, or 4 band equalizers, depending upon their number of controls.

GRAPHIC EQUALIZER - This type of equalizer utilizes many bands of peaking type fixed equalizers (filters). They are called graphic because they usually employ slider type controls on a graphic background. This provides a "Graphic" representation of preselected boosts and cuts of various frequency bands. Graphic equalizers have many pre-set filters and are usually structured according to an octave format. Each filter may be boosted or cut independant of each other. The standard format for graphic equalizers are octave, 1/2 octave, 1/3 octave & 1/6 octave. This simply means that the equalizer filters are positioned according to octave, 1/2 octave - etc. centers. So, a 1/3 octave equalizer will contain 3 times as many controls as an octave equalizer.

PARAMETRIC EQUALIZER - Unlike a fixed type equalizer the frequency centers are adjustable across a given range. With a Parametric Equalizer your are able to move the frequency center to a particular portion of the audio spectrum you wish to boost or cut. Parametric Equalizers may feature a variable "Q" where the slope rate of the filter is adjustable. If the "Q" is non-adjustable you may only "Sweep" a pre-set "Q" filter to other frequency areas within the range of the particular filter. Non-adjustable "Q" filters are sometimes called Quasi-Parametric or Sweepable Parametric equalizers.

PEAKING EQUALIZER - a Peaking type equalizer is characterized by a bell shaped curve or response when boosting or cutting a particular frequency center. So, whenever boosting a particular frequency center adjacent frequencies are boosted according to the relative slope of the bell shaped curve. Maximum boost and cut will occur at the exact frequency center while less effect will be heard at points further from the center, the reduction in level corresponding to the slope of the bell curve.

SHELVING EQUALIZER - This type of equalizer is characterized by its resemblence to a shelf in its response curve. Its maximum boost and cut appears at the corner frequency and will remain constant at all points beyond that frequency. So, a high frequency shelving type equalizer boosted at 2K would produce a boost from 2K out to 20K or the upper end of the audio spectrum. This is diffenent from a peaking type equalizer in that it would only boost frequencies in and around the 2K center.

"F"

FADE

A term used to indicate the reduction of intensity of a signal until it becomes inaudible. Fade in describes the gradual increase of signal until the signal becomes more or less distinct in level with respect to the mix.

FEEDBACK

There are several different types of feedback. All involve the interaction between the input and output of an audio system or electronic circuit. In acoustic feedback usually the output (speaker) system achieves a certain level feeding a microphone (input) creating a feedback loop. This results in a squeal or howl in the sound. In this case mic placement farther away from the output is required to eliminate this "Feedback loop." In electronic feedback the output of an electronic circuit may be of sufficient gain to feed a signal of sufficient intensity to the input of the circuit resulting in a loop. The result may be oscillation within the circuit or an audible squeal or high pitched hiss. Again, proper gain adjustment and alternate signal routing may be required. Note: Negative feedback may be used to provide better stability to an amplification circuit and lower overall distortion.

FILTER

A term losely describing a device that will alter certain portions of an audio spectrum. It is usually used to remove certain portions (unwanted) within a particular audio spectrum. Some typical filters are:

BANDPASS FILTER - A filter that cuts high and low frequency material and allows signals in-between the high and low frequency cut-off points to pass unaltered.

HIGH PASS FILTER - A filter that is designed to allow only frequencies above a certain cutoff frequency to pass. Sometimes called a low cut filter.

LOW PASS FILTER - A filter designed to pass frequencies only below a certain certain cut-off frequency. This filter is sometimes also called a high cut filter.

NOTCH FILTER - This is a filter designed to attenuate a very narrow range of frequencies while passing frequencies on either side of it. Notch filters are excellent tools in adjusting the response of speaker systems in critical environments.

FLAT FREQUENCY RESPONSE

Usually associated with a sound system that will reproduce signals that vary by +/- 3dB or less over its rated frequency response range. For instance a 20Hz to 20kHz frequency response +/- 3db means that the respnse of the system is within the +/- 3dB "Window" over the entire frequency range (20hz - 20kHz).

FLOATING

In a balanced circuit the signal leads (+) and (-) are isolated from ground and are hence "floating." When the signal leads are referenced to ground the circuit is balanced.

FOLDBACK

A term used interchangeably with "Cue" meaning "Monitor Send." It refers to the signal that will usually be "folded-back" to the stage for monitors or headphones to allow the performers to monitor program material.

FREQUENCY

The rate of change in voltage or current, or of air pressure as in an acoustic (sound) signal. Frequency is measured in cycles per second or (Hz) Hertz.
(1 CPS) = 1Hz (Hertz)

(20 CPS) = 20Hz
(1000 CPS) = 1000Hz or (1kHz)

FREQUENCY RESPONSE

Refers to the range of frequencies that an electronic device will reproduce. A second parameter must be indicated to establish the frequency response tolerance and is denoted by a +/- dB factor. This term is always associated with the response of a system or device. Example: 20 to 20KHz +/- 3dB.

FUNDAMENTAL

A term used to describe the prime or main tone of a harmonic series. Considered the basic pitch or a music note. Most instruments create harmonic overtones in addition to the fundamental tone that determines the character and timbre of the particular instrument.

"G"

GAIN

The amount of increase in the level of a signal. Usually expressed in terms of dB.

GAIN CONTROL

A circuit or device that will adjust the gain of an amplifier, usually a potentiometer.

GROUND

The potential in a part of a circuit, such as the chassis that is a zero reference with respect to other voltages and may or may not be at the potential of the earth.

EARTH GROUND - To place in connection with the earth as a ground in a circuit. A stake, water pipe, or other solid conductive connection to the earth.

SYSTEM GROUND - One point where the ground connections from various pieces of equipment are connected together and usually connected to the best earth ground.

GROUND LOOP

Where two or more paths to ground exist at slightly different potentials (unequal voltage levels) causing a 60 cycle hum or buzz within the system.

GROUPING (SUB-GROUPING)

A situation where several inputs are mixed to a single output, with one level control for the group of signals. Usually grouping is best used when many channel inputs are required to properly "mic" an instrument. All the channels are assigned a group fader which ultimately controls the volume of all of the pre-set channels simultaneously. This type of "Grouping" greatly simplifies a complicated multi-channel mix.

"H"

HARMONICS
The overtones or partials of a fundamental tone. Harmonics can be calculated as integer multiples of the fundamental frequency. The first harmonic is the fundamental frequency, the second is (2) times the fundamental, the third is (3) times the fundamental, etc.

HORN
A hollow tubular device terminating in a cone of varying cross section, for maximum collection of sound waves increasing the efficiency and better coupling the loudspeaker or driver to the environment. There are low frequency horns used to increase efficiency at lower frequencies usually below 500 Hz. High frequency horns reinforce upper, mid, and high frequencies usually above 500 Hz. There are many types of low and high frequency horns with the main purpose of the horn being to maximize the efficiency of a system by increasing output, dispersion, or projection.

"I"

I.D. STRIP
A small area located around the main and channel faders of a console to allow identification of the signals appearing at those channels. Usually a light stick masking tape is used as a backing for grease penciled labels of the various input instruments.

IMPEDANCE
The total opposition to an alternating current presented by a circuit. Impedance is measured in ohms. (See also: Resistance)

INPUT LEVEL
Usually referred to in dB, dBm, or volts that are acceptable levels to be connected to the input of a particular piece of electronic equipment. These levels are usually measured as a maximum (point at which distortion could occur) or minimum (point at which the signal is nominally driving the input circuitry).

ISOLATION
Refers to the absence of signal leakage from a channel or signal path to another within a given piece of electronic equipment. Also refers to the absence of sound leakage in adjacent acoustic environments.

"J"

JACK OR PATCH BAY
Refers to a series of single inputs or outputs for routing signals easily to or from equipment in a studio or equipment rack. It provides for maximum flexibility in routing signals and is extremely convenient for temporarily connecting equipment to a system.

JUNCTION BOX
Sometimes call a "snake box." It usually features several "XLR" type connectors attached to a multi-conductor shielded cable. Usually used to connect several microphones for subsequent routing to a remote mixer.

"K"

kHz
The abbreviation for 1000 cycles per second, or "Kilo-cyles."

"L"

LED
Light Emitting Diode. A solid state device which emits a light either visible or infrared when a D.C. current flows through it.

LEVEL
A term that usually indicates the relative volume or amplitude of a signal or sound. Usually based on a particular reference expressed in dB, dBm, SPL, etc.

LIMITER
A type of compressor with a high compression ratio typically 10:1 or greater that prevents a signal level from exceeding certain pre-set levels.

LINE INPUT
An input designed to operate at "Line levels" sometimes call "pre-amp" levels."
Some standard line levels are :
+8dBm (1.95 volts RMS) - Broadcast
+4dBm (1.23 volts RMS) - Pro-recording/sound reinforcement
-10dBv (.316 volts RMS) - Pro-recording and alternate equip.

"M"

MIC LEVEL

The signal level of a microphone usually expressed in dBV, dBv, or dBm. Mic level is generally considered to be approximately -40 to -60 dBv. In the presence of strong inputs certain mics may produce outputs of up to +4dBv.

MICROPHONE

A device for converting sound waves into electrical currents. There are several different types and classifications of microphones, each specialized for a particular application. Local studios are an excellent source of information regarding microphones and which types best suit a particular application.

MICROPHONICS

An audible noise induced by mechanical vibration or shock of electronic components.

MIXER

Sometimes calledmixing "consoles." A stereo mixing console would combine several inputs (usually 6, 8, 12, 16, 24, 32, or more channels) down to (2) outputs (Left and Right). The ability of the mixer to send each channel to its respective left or right output establishes its stereo capability.

MONITOR MIXER

A mixer or mixing console that is used primarily for balancing the sound fed to the monitor speakers or performer headphones. Usually this type of mixer is used for live performances where the "on-stage" sound and balance would be substantially different from the house sound. Many mixing consoles have a built in "Monitor" section for subsequent feed to the monitors. It is sometimes called a "Stage Mix" section.

MONO OUTPUT

The output of a mixer or electronic device where no stereo separation is present. Many mixing consoles feature stereo capability simultaneous to providing a "Mono" or combined output representing the sum of the left and right stereo outputs. This output is very usefull for feeding such mono sources as broadcast feeds, monaural tape recorders, or other devices requiring a monural signal.

MOTORBOATING

Low frequency oscillation usually resulting in a low frequency sound similar to a motorboat.

MUDDY

An expression referring to the lack of clarity or definition of sound, usually caused by distortion, excessive bass, deficient or underpowered high frequency components or the like.

MUTE SWITCH

A switch that typically stops the signal flow from a channel input or channel output of a console. Similar to a channel on/off switch or kill switch. It allows the operator to "Shut off" a channel without affecting the channels volume setting should it be required later in the mix.

"N"

NOISE

Any unwanted noise or random disturbance that causes the received signal to differ from what was transmitted. Characterized as unwanted signals such as hum, hiss, rumble, etc.

PINK NOISE - Random noise that is modified to have equal energy levels at each octave. Pink noise is commonly used to help analyze a sound system for equal frequency projection (volume of each frequency) for a particular room. Pink noise is amplified by the system and analyzed using a "Real Time Analyzer" at various parts of the room to determine if all frequencies are reproduced at equal volume levels at those analyzation points.

RANDOM NOISE - Unwanted noise usually manifested as a hiss or crackling within the system. Usually caused by minute fluctuations in electronic component tolerances (i.e. Thermal noise).

NOISE (EQUIVALENT INPUT)

The measure of how quiet a microphone pre-amplifier is. The basic calculation measures the output of a device and subtracts the amount of amplifier gain in dB. This along with other caculations regarding input inpedances and how the noise is expressed determines the overall figure for the noise floor (quietness) of the pre-amp.

NOISE FLOOR

The level at which noise exists in an electronic device without the presence of signal. It is commonly measured in the absence of signal.

NOISE GATE

An electronic device that turns off or severely attenuates an audio signal when the level of the signal falls below a pre-set threshold level. So, whenever a signal is just above the pre-set threshold, the signal is passed without any alterations. When the signal level drops below the threshold it is severely attenuated or turned off eliminating any noise, crosstalk, or leakage from being amplified.

NOISE (UNWEIGHTED)

Measuring the noise level of audio equipment using test equipment sensitive to a wide range of frequencies with equal response at all frequencies.

NOISE (WEIGHTED)

Measuring the noise level of audio equipment using any of several standard filters that restrict the response of the audio spectrum. Weighted noise specs are utilized to give a more "real world" approximation of the noise performance of a piece of electronic equipment as used in an audio system.

NOMINAL OPERATING LEVEL

The average signal level with which a particular circuit or piece of electronic equipment is designed to operate.

"O"

OHM

The unit of measure of electrical resistance, or impedance.

OUT OF PHASE

When two signals are offset in time either electrically or acoustically, or two circuits are reversed in polarity with respect to one another. A 180 degree out of phase signal when combined with its "In phase" counterpart will cancel one another.

OUTBOARD

Usually referred to equipment that is not incorporated in the console. Outboard equipment can be such items as compressors, delays, or other "outboard" signal processing equipment (i.e. external equipment usually plugging into the console).

OUTPUT CHANNEL

Usually referred to as the final output of any multi-channel audio device. Input channels from a console are usually combined or "mixed down" into output channels. Input channels can be routed to various output channels which in turn will drive individual channels of a recorder or power amplifiers.

OVERDRIVE LEVEL

The level at which distortion will occur. Usually the point at which the signal source is too strong for the capability of the electronic device it is feeding. The point at which clipping will occur.

OVERHEAD MIC

A microphone or series of microphones that are placed above the instruments or performers. It is usually used to increase either the ambience or overall pickup of the instrument or performer. Overhead mics can be used as a sole source for picking up an instrument or, more often, it is used in conjunction with close micing for ambience and smoother overall pickup of an instrument for a more natural sound.

OVERLOAD

A situation characteristic of attempting to deliver more power or output than an electronic device's power supply is capable of delivering. If loading a power amplifier down excessively with multiple speaker loads, an overload condition can occur. This usually results in distortion and could cause damage to occur to speakers.

"P"

PAD

A passive resistor network which is used to lower the input strength of a signal in order to accomodate a fixed input level of a pre-amp or other electronic device.

PAN

A term used to indicate the ability to send a mono signal to a stereo (left, right) image. A "Pan Pot" or potentiometer is usually used as the control for this action.

PASSIVE DEVICE

Passive devices are electronic components that usually induce losses in an audio circuit and do not possess amplification circuitry unlike active devices. Active devices require a power supply to drive them and have inherently more control than passive devices.

PATCHING

The process of routing and re-routing audio signals using patch cords. Patching can be done internally utilizing switches or "Patches" or externally using patch cords or such.

PATCH POINTS

The points in an audio circuit that can be brought out of the circuit and reintroduced to the circuit via patch cords or internal patches. Usually considered a point at which the audio circuit is interrupted for insertion of various electronic processing devices.

PEAK LEVEL

The maximum instantaneous level measured as either power or voltage of a signal.

PHANTOM POWER

A method of powering a remote pre-amp or impedance converter built into many condeser microphones. A voltage (from 6 to 48 volts D.C.) is sent along the audio carriers of the microphone cable remotely powering the condenser mic's. The Phantom voltage (D.C.) is separated from the audio at the input pre-amp on the console using D.C. blocking capacitors or special transformers.

PHONO PLUG

Sometimes called RCA plug or Pin plug. It is commonly used for unbalanced connections or "single ended" connections.

PHONE PLUG

Describes any of several types of 1/4" shaft diameter plugs. The male connector is termed the "plug" and the female counterpart is termed the "jack". A (TS) "Tip/Sleeve" phone plug is a mono single ended connection used for unbalanced audio connections. A (TRS) "Tip/Ring/Sleeve" phone plug describes a "stereo" phone plug used for balanced 1/4" connections or stereo headphone connections.

POLARITY

Refers to the relative position of the high (+) and low (-) signal leads in an audio system. If two signals of equal and opposite polarity are combined, each signal will cancel the other out.

POP FILTER

An open celled foam shield or similar material placed over or in front of a microphone in order to eliminate sudden bursts of sound or "Popping" from breath or other dynamic sources. Sometimes it may be an electronic filter (usually a band reject type filter with a center around 60Hz to 100Hz) filtering the frequency region where popping sounds usually center.

POST

Refers to a signal input or output or routing point that comes after something. It is a term usually used in conjunction with another term such as; Post Fader - after the fader, Post-EQ - after the equalizer or, etc. Usually this term, if not stated otherwise, means after the fader; Post fader.

PRE

Refers to a signal or routing point that occurs before something. Usually used connected with another term such as "Pre-fader,""Pre-EQ," etc. If not stated otherwise this term means pre-fader.

PRESENCE RANGE

This term refers to the frequency range most sensitive to the human ear as it affects the "presence" of sound. The range from 2kHz to 5kHz is generally considered the presence range.

PROXIMITY EFFECT

A low frequency boost that occurs when a cardiod type microphone is held very close to a sound source.
lead 14.4pt

"Q"

Q

A description of the width or sharpness of a filter. The characteristic of the boost or cut of a filter. The "Q" is usually associated with a parametric equalizer where this parameter is selectable enabling it to be used as a notch type filter or more broad band filter. (See "Equalizer" & "Filter".)

"R"

REAL TIME ANALYZER

A piece of test equipment with a display to indicate the relative energy or level of each frequency band. Audio real time analyzers have a range of from 20Hz to 20Khz and will usually allow the operator to observe the room equalization (or relative levels of each frequency band) within the defined range. It allows the operator to visually check the relative levels of different frequency bands thus enabling him to subsequently adjust each band for equal levels. A room where each frequency is at an equal intensity "energy level" is said to have a "flat" frequency response.

RESISTANCE

The opposition that a conductor offers to the passage of D.C. current, measured and expressed in ohms.

REVERBERATION

The audible persistance of sound caused by numerous reflections from the walls, floor and ceiling of a room. Also can be created by electronic means or through mechanical devices.

RMS LEVEL

A specific measurement of a signal level by mathematically squaring all the instantaneous voltages along the waveform averaging the squared values and taking the square root of that number. The term "RMS" literally means (Root Mean Square).

RMS POWER

A misnomer used to describe the output power capability of an amplifier. "RMS" power is actually referring to the RMS voltage squared and divided by the nominal speaker impedance. Properly called "continous" power.

"S"

SENSITIVITY

A reference level usually measured in dB, dBv, or volts that determines the relative strength of a signal to drive other electronic equipment to its proper operating level.

MIXER INPUT SENSITIVITY - Either the mic, or line input level which will drive the mixer output to nominal rated line level.

POWER AMP INPUT SENSITIVITY - The relative input level required to drive the amplifier to full output power levels.

LOUDSPEAKER SENSITIVITY - The sound pressure level measured in dB SPL (at a give distance from the speaker) when a given input power is fed to the speaker.

MICROPHONE SENSITIVITY - The output voltage measured in dBV where a microphone is exposed to a specific sound pressure level (usually 94dB SPL). Certain standards such as the EIA standards take into account the impedance to which the microphone is connected.

SEVENTY VOLT LINE

A method by which a large number of speakers can be driven at long distances with minimal line losses. It requires an amplifier with a relatively high output voltage swing (usually 70 volts). The signal is routed to relatively high impedance transformers (when compared to conventional low impedance speaker systems of typically 4, 8, or 16 ohms). The transformers in turn drive the loudspeakers allowing more speakers to be operated without excessively loading the power amplifier. It also allows greater flexibility in driving various types of speakers within a particular system.

SIGNAL PROCESSING EQUIPMENT

Any electronic equipment or electronic circuit that is used to alter or change the characteristics of a signal. Signal processors include devices such as equalizers, compressors, flangers, reverb, digital delays, etc.

SIGNAL TO NOISE RATIO

A term describing the measurement of the difference between either the nominal, or maximum operating level and the noise floor. It is usually expressed in terms of dB's, and may be measured according to different standards. (See noise - "Weighted" and "Unweighted".)

SLATE

A voice or talkback signal applied to each of the output channels or busses of a console in order to record that signal on tape, later allowing easy location of certain songs or recordings on the tape. Sometimes a 30 to 60Hz signal is added to the signal to allow the operator to easily locate a particular recording or "take" during high speed tape shuttling.

SLIDER

A term for a potentiometer configured as a straight line control as opposed to a rotary control.

SNAKE

A multi-channel audio cable used to carry many signals from the stage or studio to a remotely operated mixing console. It is intended to be used with microphone or line level signals only, however, some "Snakes" are available that will carry speaker level signals as well.

SOLO

A feature on many consoles where a single channel or group of channels can be listened to individually exclusive of all other signals. Soloing allows the operator of a console to listen to a particular channel without affecting the main mix thus allowing the operator to fine tune a mix during the performance.

SOUND PRESSURE LEVEL (SPL)

An acoustic measurement of sound energy. Usuaully measured in dB SPL.

SOUND REINFORCEMENT

A term that refers to Public Address or the amplification of sound by electrical means.

STEREO MASTER

Either (2) faders positioned close together or a (2) ganged fader controlling the left and right outputs that feed either a stereo P.A. system or stereo tape recorder. (Sometimes called the stereo buss output.)

SUBSONIC

A misnomer for the term "infrasonic" meaning frequencies below the audible range. Usually refers to frequencies below 20Hz.

SUB-WOOFER

A loudspeaker or louspeaker system designed to specifically reproduce the lowest of audio frequencies. Its normal range of operation is 300 Hz and lower.

SYSTEM GROUND

The main ground point for a sound reinforcement system or recording system. The use of a single point ground where all grounds go to one single point rather than multiple points, aids in preventing ground loops, hum, buzz, or radio frequency interference.

"T"

TALKBACK
A system built into a mixing console whereby the operator is allowed to communicate with performers either on stage or in a studio.

THRESHOLD
Usually referred to as the point where a compressor, limiter, or other signal processing device begins to act on a signal. The threshold is usually adjustable to accomodate any type of input signal level.

THUMP
A brief low frequency noise (transient) that is heard in powering up or down particular pieces of equipment within an audio system. Sometimes call an On/Off transient.

TONE CONTROL
A broad range equalization control. Referred to as Bass, Mid, and Treble in most systems.

TRANSDUCER
A device which converts energy from one form to another. loudspeakers are a transducers converting electrical signals to acoustic, whereas microphones convert acoustic energy to electrical.

TRANSIENT
A short duration signal. Dynamic music is filled with transients that add dynamic range to music. Transients within music are desirable as long as they fall within the dynamic headroom of the equipment and do not induce distortion.

TREBLE
The high end of the audio frequency range. Usually considered above 2000Hz. Boosting the treble refers to increasing the relative volume level of those frequencies above 2000Hz.

TRI-AMPLIFICATION
A tri-amplified system utilizes a low level crossover to divide the full range audio signal into a low frequency, mid frequency, and high frequency output. These outputs are then fed to amplifiers that specifically amplify those frequencies. The outputs of the low frequency, mid frequency and high frequency amplifiers then directly feed those speakers dedicated to those ranges within the speaker system. Tri-amplification differs from Bi-amplification in that it divides the audio signal into 3 separate frequency bands instead of two. (See Bi-amplification.)

TUNED PORT
In a bass reflex speaker system it is a vent that optimizes the response of a speaker cabinet according to the shape, size, and dimensions of the vent, and its interraction with the specific operating parameters of the speaker.

TWEETER
A transducer designed to reproduce the treble or high frequencies of an audio

system.

"U"

ULTRASONIC

Those frequencies above the high frequency limit of human hearing. Usually refering to those frequencies above 20kHz.

UNBALANCED LINE

An audio circuit having only two conductors. One is the high output or (+) signal and the other is the (-) signal and is usually connected to chassis or ground. Since these signal leads are not at equal potential when referenced to ground, they are considered "Unbalanced" with respect to ground.

UNITY GAIN

This term refers to a circuitwhere the input signal level is exactly the same as the output level. Its output is therefore "unity," with respect to the input signal gain. Also 0dB gain.

"V"

VIBRATO

A frequency modulation, or variation in the frequency of a sound at a regular interval. Usually that interval is between 2 and 15 times per second.

VOICE COIL

A coil of wire wrapped around a bobbin in a loudspeaker The voice coil is connected to the cone of the speaker and causes subsequent movement of the cone when producing sound. A "Blown" or "Open voice coil" indicates that the wire used in the voice coil has been burned apart and will no longer conduct an electrical current through it.

VOLUME

A term describing the overall level of sound, or the intensity of sound.

VU

A unit of audio level change. A change of one VU is the same as a change of one decibel for a sine wave. It is more specifically used to describe the overall level of more complex waves forms such as speech, or music.

VU METER

A meter used to measure audio level in "Volume units." VU meters may be calibrated to various standards depending upon the input levels of the equipment it will be hooked up to. VU meter balistics are relatively slow and brief transients often exceed the indicated level by 10 to 20dB.

"W"

WATT

A unit of measurement of electrical or acoustical power.

WET

A signal to which reverb, echo, or other ambience has been added as opposed to a "Dry" signal having no ambience or reverberation.

WOOFER

A loudspeaker or transducer in a 2-way or more complex speaker system that is used to reproduce only the bass or lower end of the audible frequency system. Usually associated with frequencies below 1000 Hz.

"X"

XLR

Refers to several varieties of 3 conductor or more complex connectors. These connectors feature locks and mating connectors most commonly used to make microphones, pre-amp or balanced connections. Sometimes called "Cannon" connectors names for the first company to manufacture them although there are a host of new companies making compatible alternates.

"Y"

Y-CONNECTOR

A connector or cord that splits one signal source to be routed to two different sources. Also call a "Y" adapter, or parallel "Y" connector.

"Z"

Z

An abreviation for the term "Impedance." (See "Impedance".)

DISASSEMBLY FOR SERVICING

Removing the Bottom Cover

In order to service the MX-22 Series Consoles it is necessary to first remove the mixer's bottom cover. The following steps give the procedure for safely removing the cover.

1. *Disconnect the AC line cord and all other cables connected to the mixer.*

2. *Remove the 3 large screws at the bottom of each wooden end panel.*

3. *Stand the mixer up on its right end and remove the eight screws securing the bottom cover. Lift off the cover and note the locations of the foam pads so that you orient the cover correctly when you replace it.*

4. *When removing the bottom panel you will have to disconnect the reverb signal cables. You should mark which cables route signal to the input of the reverb tank and which cables return the signal from the tank. The yellow/violet cables should connect to the reverb input & the green/white cables connect to the reverb output.*

Removing a Circuit Module

When removing any of the Circuit Modules, it is important that all ribbon cable connectors be unplugged from the module before the module is loosened from the top panel. Likewise, on reassembly the module should be secured to the top panel before the ribbon cable connectors are plugged into the module. The steps for removing a module are as follows:

1. *Remove the bottom cover as described above.*

2. *Identify the module to be removed and carefully disconnect all ribbon cable connectors from that module. Be careful not to pull the ribbon out of the connector.*

3. *Remove all knobs from the controls on the top panel; then remove the control nuts from each pot being careful not to mar the finish. Use an 11mm or 7/16" nut driver for all control nuts.*

4. *The module should now be free to be removed. Pull the module behind the top panel and then bring it down under the master buss cable and out of the mixer.*

Replace the module(s) and cover by reversing the steps above. Inspect the connectors that have been removed and reinstalled to see if any conductors of the ribbon cable have come loose due to handling. In the event that you see any loose connections use a small screwdriver to reseat the wire back into the connector being careful not to use too much force on the module.

BACKUP CIRCUIT MODULES AND ACCESSORIES

The MX-22 series consoles have been carefully designed to employ only five major circuit modules. These modules comprise the major circuitry of the console and should be the only ones needed as appropriate back-ups. This modular construction simplifies not only the manufacturing of the mixer but also its servicing. If a problem occurs, in many cases it will be possible to determine which module has failed and then that module can be returned to CARVIN for repair. This spares the expense and inconvenience of returning the complete mixer. If the returned module is an input channel you will be able to continue operating the mixer while the problem module is being repaired. Failure of either the system master module or power supply module will make the mixer inoperable. For applications where mixer down time would cause significant loss of income, we suggest that spare modules be purchased and kept available in the event of failure. The most important module to back up is the System Master. The next most important module to back up is the Input Channel followed by the Power Supply. Module pricing as of the date of this manual's publication is given below.

MODULE	PRICE	PART NO.
System Master	$120.00	PCB 80-MX22-4
Input Channel	$90.00	PCB 80-MX22-1
Power Supply	$75.00	PCB MXPS-8S
Stereo Power Mod	$175.00	PCB 80-MX22-8
Talkback Module	$50.00	PCB 80-MX22-9
Dual 9 band EQ module.	$115.00	PCB-80-2244-5

ACCESSORIES	PRICE
AN-68 (Anvil Case for MX-622P, MX-822 and MX-822P Consoles)	$199.00
AN-1216 (Anvil Case for MX-1222, MX-1222P & MX-1622 Consoles)	$229.00
AN-24 (Anvil Case for MX-2422 console)	$269.00
G-12 (BNC type Little Lite for all above consoles)	$25.00

(Shipping on all above Anvil Cases $15.00 dollars)
(Shipping on G-12 Little Lites $2.00 dollars)

MX22 TECHNICAL SPECIFICATIONS

Frequency Response

Mic or line inputs to two-track output:	20 Hz-20kHz ± 1 dB

Total Harmonic Distortion

Mic in to two-track out 40 dB gain 0 dBv output, 20-20kHz:	less than .03%
Line in to two-track out 10dB gain +10 dBv output, 20-20kHz:	less than .02%

Equivalent Input Noise

unweighted, 150 ohm source:	-128 dBv

Output Noise

(unweighted/20kHz B/W) (two-track, mono, or monitor outs)	
All faders minimum:	-85 dBv
Master faders nominal	-80 dBv

Maximum Gain

Mic in to two-track out:	70 dB
Line in to two-track out:	35 dB
Effects return to two-track out:	22 dB

Crosstalk

Adjacent channels:	-60 dB at 1kHz -50 dB at 10kHz

Common Mode Rejection Ratio:	-75 dB at 1kHz
Peak Warning Level:	6 dB below clipping (+14 dBv)
Phantom Power:	+48 VDC applied to pins 2 and 3 of XLR

Channel Equalizer

Type:	3 band w/sweep mid frequency
Hi band:	±15 dB @ 10kHz
Mid band:	±15 dB, 200 Hz to 4kHz sweep
Low band:	±15 dB @ 80 Hz

Graphic Equalizers (2)

Type:	9 band at octave intervals
Max boost/cut:	±15 dB
Center frequencies:	63, 125, 250, 500, 1k, 2k, 4k, 8k, 16k

Mic Input

Connector:	3-pin XLR type
Input impedance:	3k ohms (balanced)
Source impedance:	nominal "low impedance" (50 ohms to 1k ohms)
Nominal input range:	-70 to -10 dBv (.3mV to 300mV)
Maximum input level:	+10 dBv (3.3V)

Line Input

Connector:	¼" phone jack
Input impedance:	22k ohms (unbalanced)
Nominal input range:	-20 dBv to +10 dBv (100mV to 3V)
Maximum input level:	+30 dBv (30V)

Metering	High quality dynamic VU meters are factory calibrated for +4 dBv at the balanced outputs. Meters may be recalibrated for -10 dBv, or +8 dBv at 0 VU.

Mono and Two-Track Outputs

Balanced output connectors:	3-pin XLR
Unbalanced output connectors:	¼ inch phone jack
0 VU Nominal output level:	+4 dBv balanced -2 dBv unbalanced
Maximum output level: (10k ohm load)	+20 dBv unbalanced +26 dBv balanced

Monitor 1, Monitor 2, and Effects Outputs

Connector:	¼ inch phone (unbalanced)
Nominal output level:	+4 dBv
Maximum output level:	+20 dBv (10k ohm load)

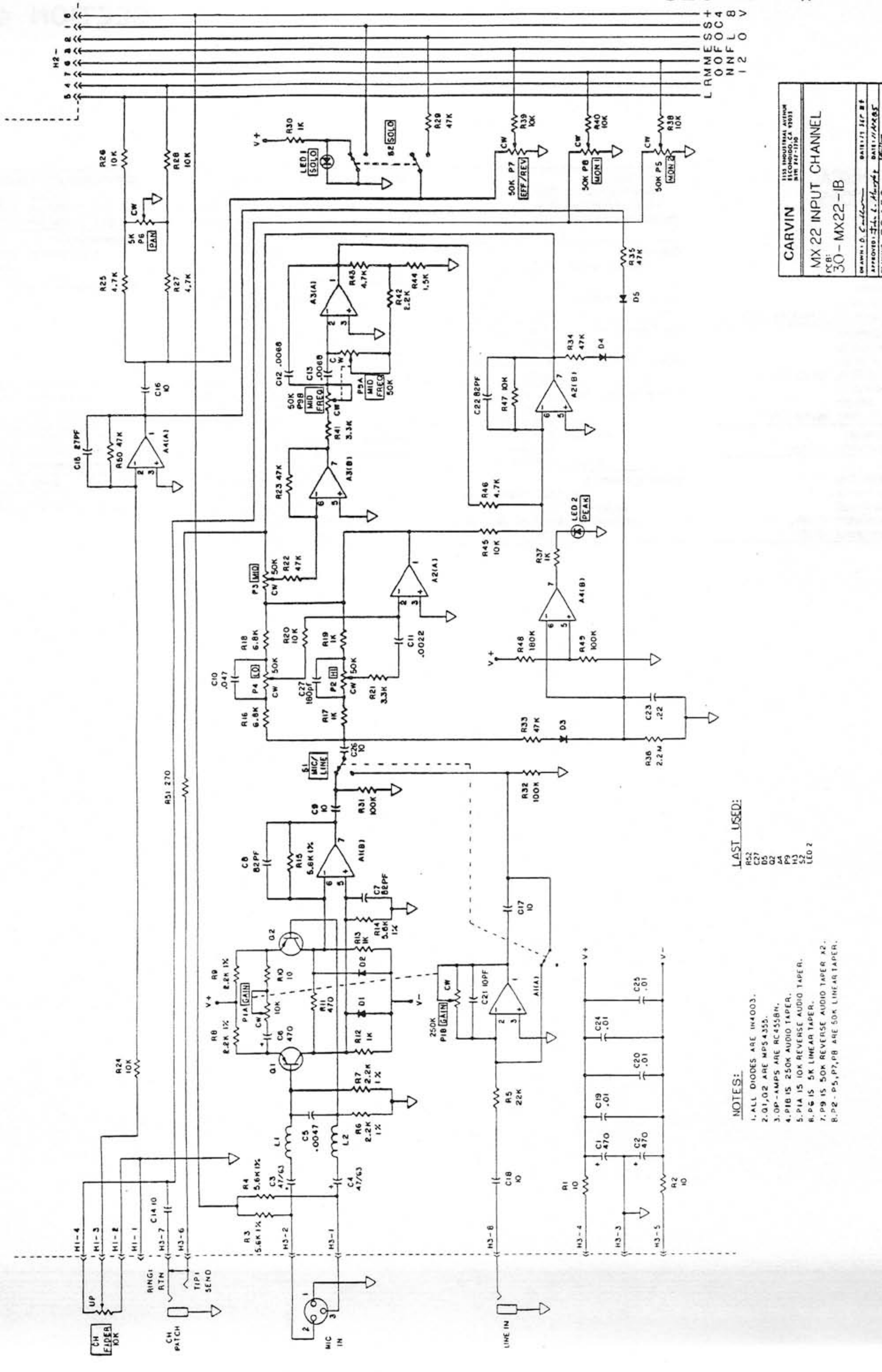
Headphone Output

Load impedance:	¼" stereo phone jack 8 ohms or higher (stereo)

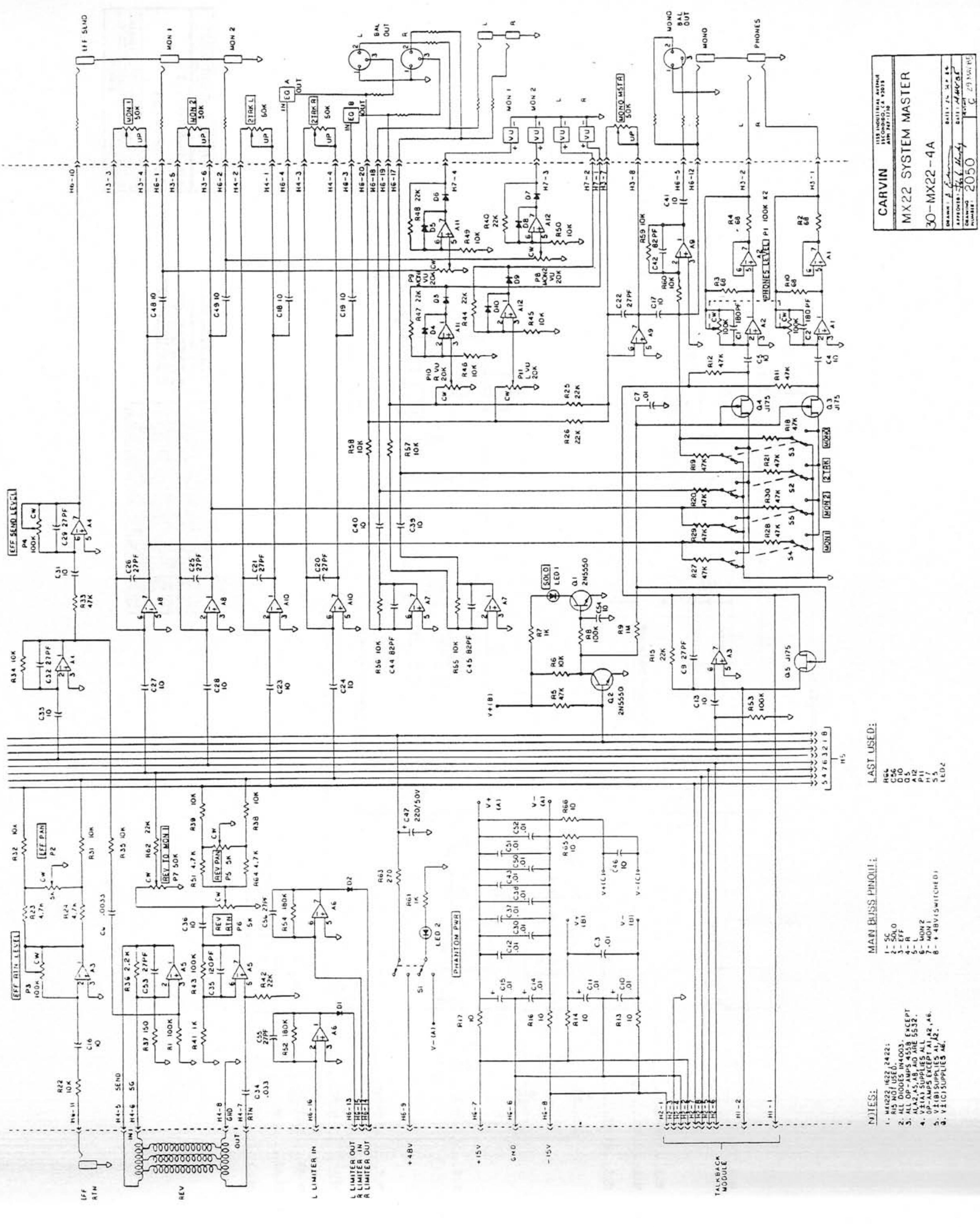
Power Amplifier (All Powered Models)

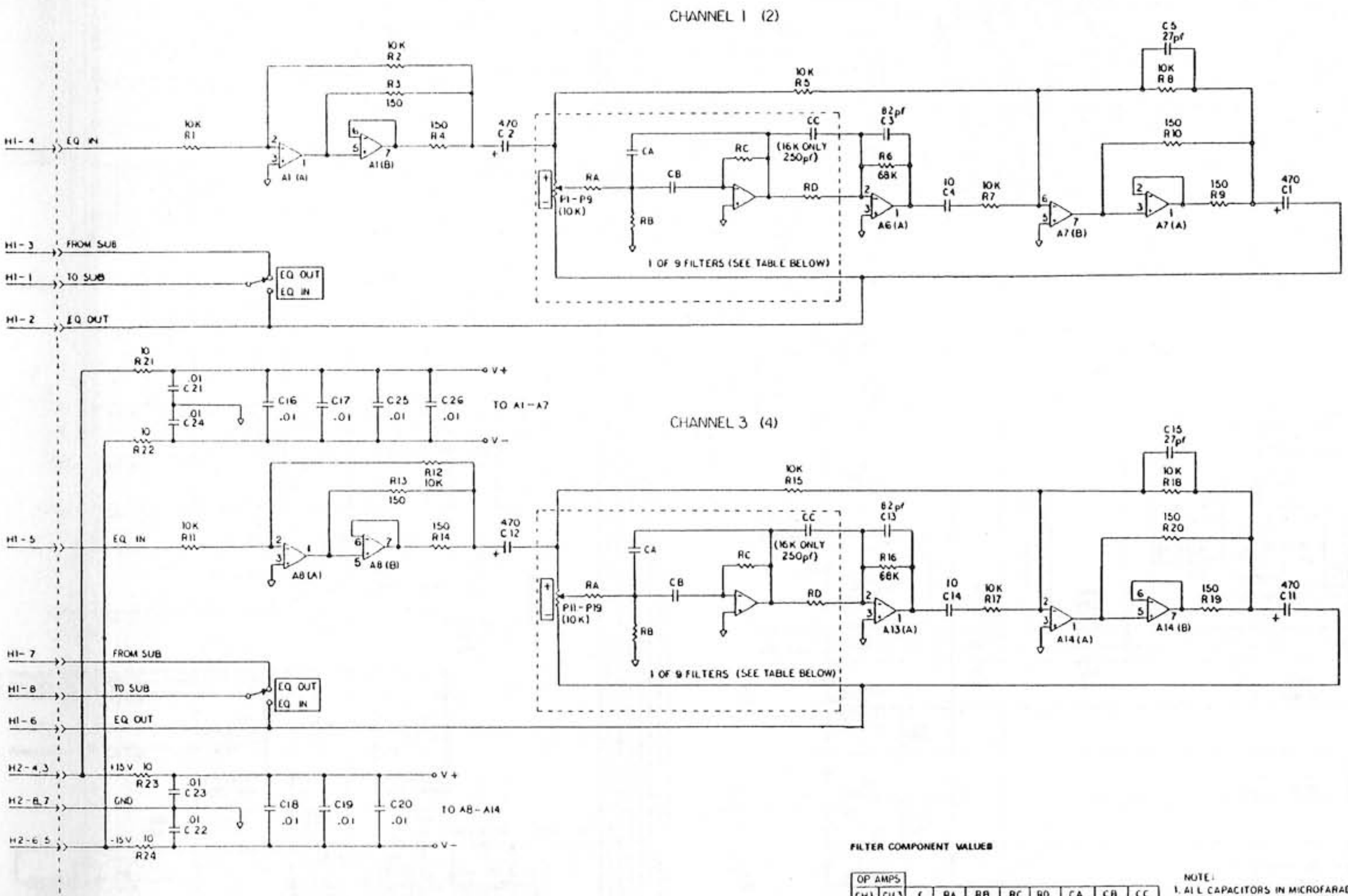
	800 watts @ 2 ohms (400 watts/ch) 600 watts @ 4 ohms (300 watt/ch) 350 watts @ 8 ohms (175 watts/ch)
Total harmonic distortion:	.1% (20 Hz to 20kHz)
Frequency response:	20 Hz to 60kHz (±3dB)
Minimum load impedance:	2 ohms per side

Power Supply:	Fully regulated & protected
Mini-Lamp:	BNC connector (6 VAC)
Power Requirements:	120/240 V AC 50 or 60 Hz
Weight:	See Price Chart
Warranty:	1 year parts and labor


CARVIN
MX 22 INPUT CHANNEL
PCB: 30-MX22-1B
DRAWING NUMBER: 2045C
NOTES:
1. ALL DIODES ARE IN4003.
2. Q1,Q2 ARE MPS4355.
3. OP-AMPS ARE RC4558N.
4. P1B IS 250K AUDIO TAPER.
5. P1A IS 10K REVERSE AUDIO TAPER.
6. P6 IS 5K LINEAR TAPER.
7. P9 IS 50K REVERSE AUDIO TAPER X2.
8. P2-P5,P7,P8 ARE 50K LINEAR TAPER.
LAST USED:
R52
C27
D5
Q2
A4
P9
H3
S2
LED 2
MIC IN
LINE IN
CH PATCH
CH FADER 10K

SECTION #9-5


CARVIN
MX22 SYSTEM MASTER
30-MX22-4A
2050
LAST USED:
MAIN BUSS PINOUT:
NOTES:
TALKBACK MODULE



FILTER COMPONENT VALUES

OP AMPS									
CH1	CH3	f	RA	RB	RC	RD	CA	CB	CC
A2	A9	63	7.5K	7.5K	33K	4.7K	.22	.22	—
A3	A10	125	3.9K	3.9K	18K	5.6K	.22	.22	—
A3	A10	250	20K	20K	91K	5.6K	.022	.022	—
A2	A9	500	10K	10K	43K	5.6K	.022	.022	—
A4	A11	1K	4.7K	4.7K	22K	5.6K	.022	.022	—
A4	A12	2K	24K	24K	110K	5.6K	.0022	.0022	—
A5	A12	4K	12K	12K	56K	5.6K	.0022	.0022	—
A6	A11	8K	6.2K	6.2K	27K	4.7K	.0022	.0022	—
A5	A13	16K	3K	3K	15K	4.7K	.0022	.0022	250pf

NOTE:

1. ALL CAPACITORS IN MICROFARADS UNLESS OTHERWISE INDICATED
2. ALL RESISTORS IN OHMS
3. ALL RESISTORS ARE .25W
4. A1, A6, A7, A8, A13, A14 ARE 5532 TYPES. ALL OTHER OP AMPS ARE 4558 TYPES.

CARVIN	1155 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 (619) 747-1710
MX GRAPHIC EQ	
30-2244-5A	
DRAWN: Sharon Nissing	DATE: 30 Jan 80
APPROVED: John L. Murphy	DATE: 27FEB84
DRAWING NUMBER: 2025C	2 NOV 84

CARVIN
1155 INDUSTRIAL AVENUE
ESCONDIDO, CA 92025
747-1710

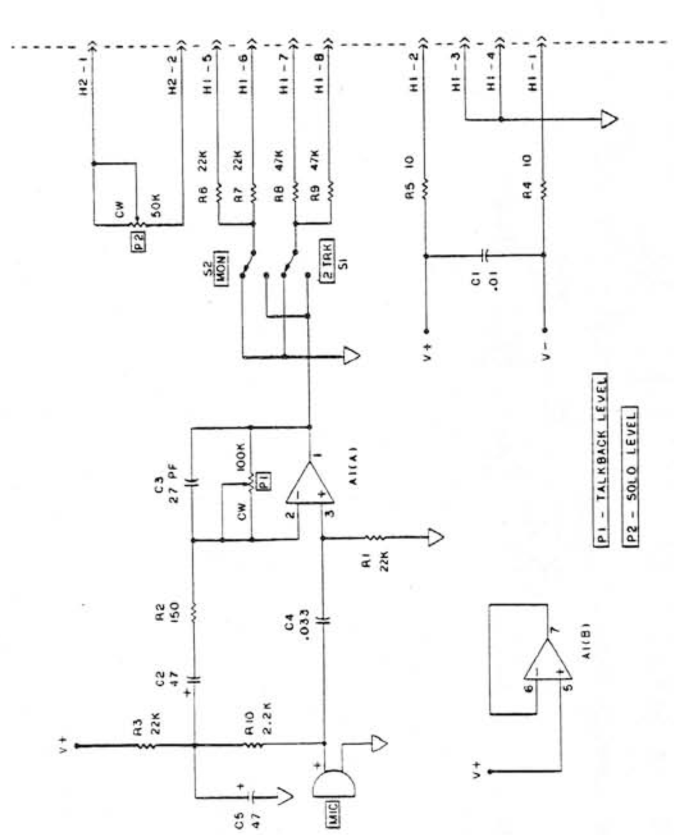
TALKBACK MODULE

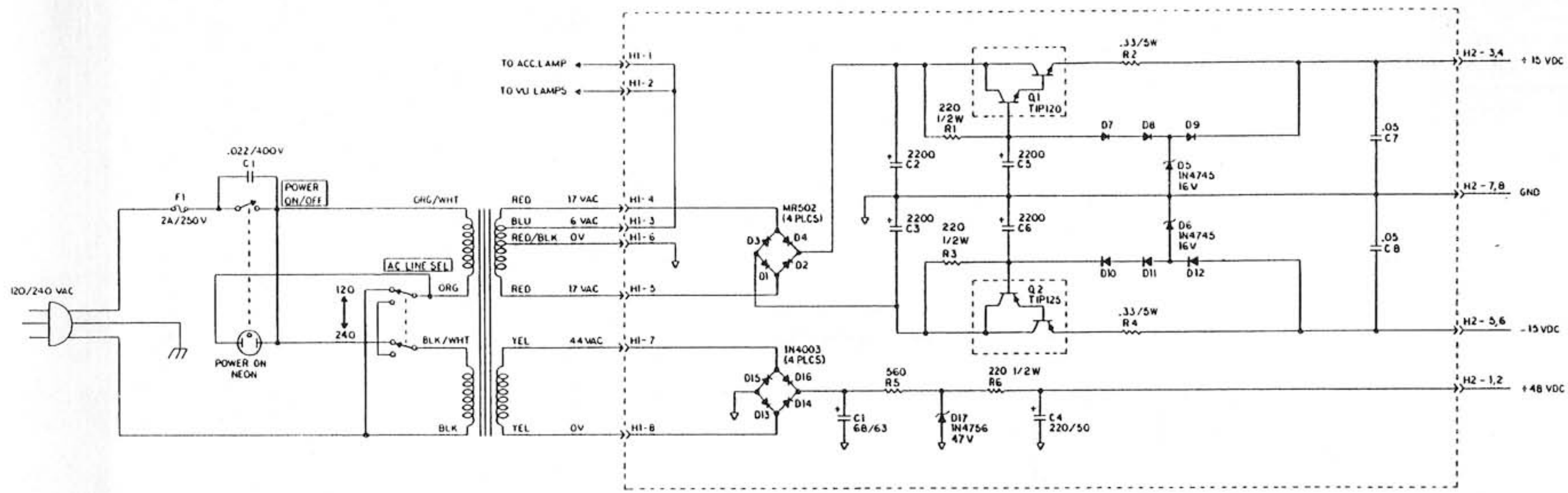
30-MX22-9A

DRAWN: C. C. [illegible] DATE: 18 SEP 84

APPROVED: [illegible] DATE: 13 MAY 85

DRAWING NUMBER 20478 REV B



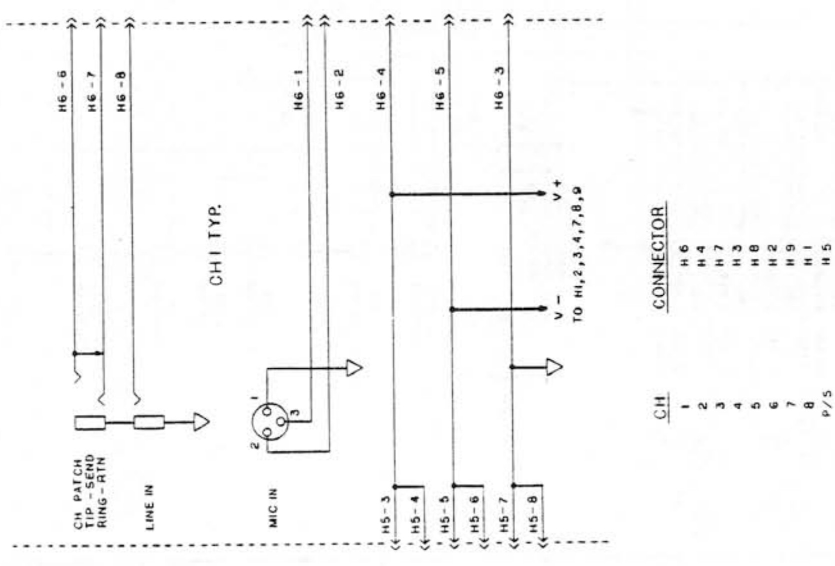


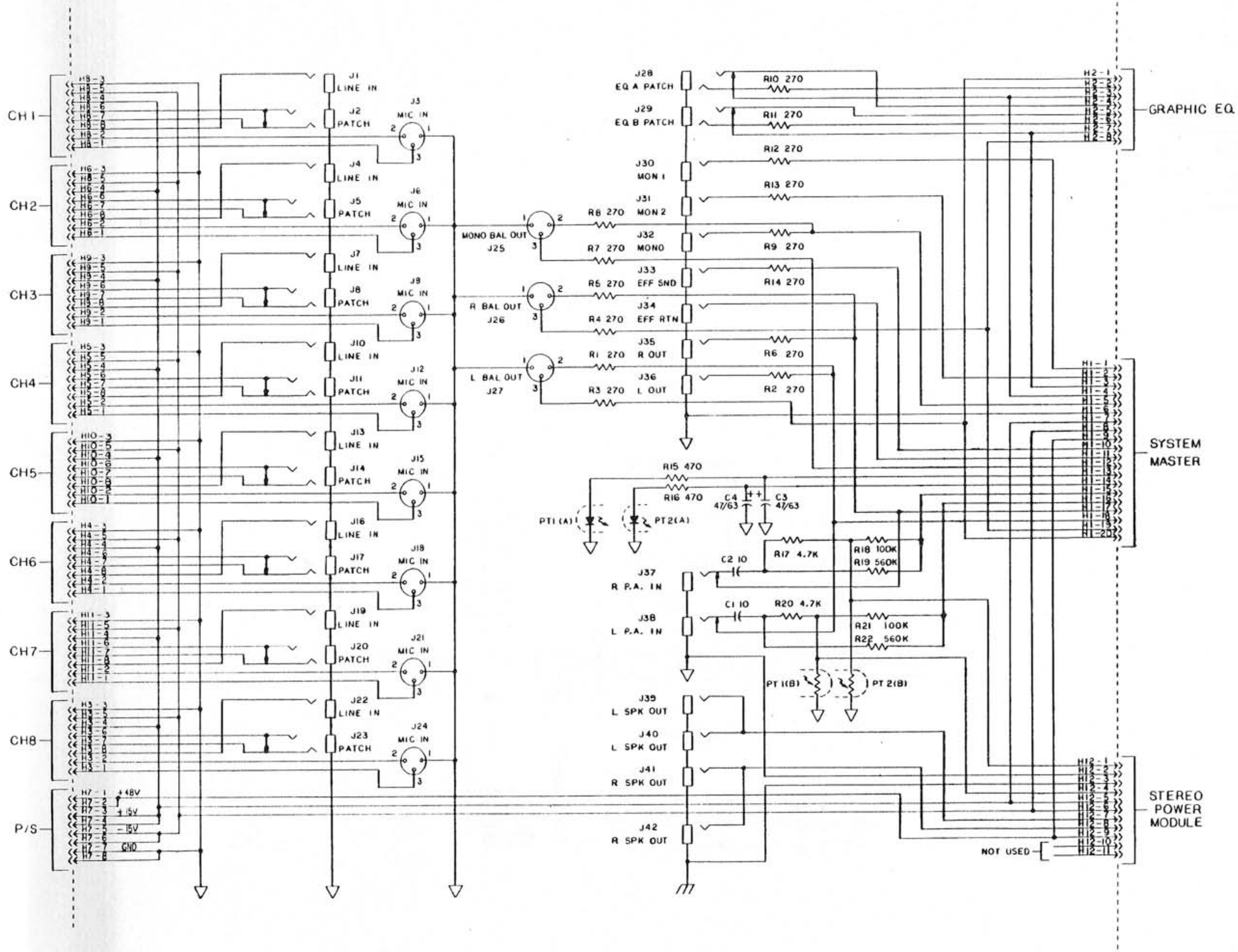
NOTE: UNLESS OTHERWISE INDICATED

1. ALL CAPACITORS IN MICROFARADS
2. DIODES 7, 8, 9, 10, 11, 12 ARE 1N4003
3. ALL RESISTORS ARE 1/4W

CARVIN	1155 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 (619) 747-1710
DC POWER SUPPLY MODULE 30-MXPS-8A	
DRAWN: Sharon Maup	DATE: 13 MAR 84
APPROVED: John L. Murphy	DATE: 16 MAR 84
DRAWING NUMBER: 2035C	2 NOV 84

CARVIN
1155 INDUSTRIAL AVENUE
ESCONDIDO, CA 92025
MX22 INPUT CONNECT
30- MX22 - 6B
DRAWN: B. Cullen DATE: 18 SEP 84
APPROVED: DATE:
DRAWING NUMBER: 2048





NOTES:

1. PT1 AND PT2 ARE VTL5C2.
2. D.C. VOLTAGE IS SUPPLIED BY H12 AND DISTRIBUTED BY H7 IN POWERED UNITS.
3. D.C. VOLTAGE IS SUPPLIED BY H7 IN NON-POWERED UNITS.
4. CH PATCH:
   TIP - SEND
   RING - RTN
5. EQ. PATCH:
   TIP - IN
   RING - OUT
6. THERE ARE FIVE(5) VARIATIONS OF THIS PCB AS FOLLOWS-

C. MX622P

1. DELETE: J19 - J24
   H3
   R10 - R11
   H11

D. MX822P, MX1622P

1. AS SHOWN

A. MX822, MX1622, MX2422

1. DELETE: J37 - J42
   PT1 - PT2
   R15 - R22
   C1 - C4
   H12
2. MX1622: CH1 - 8 BECOME CH9 - 16
3. MX2422: CH1 - 8 BECOME CH17 - 24

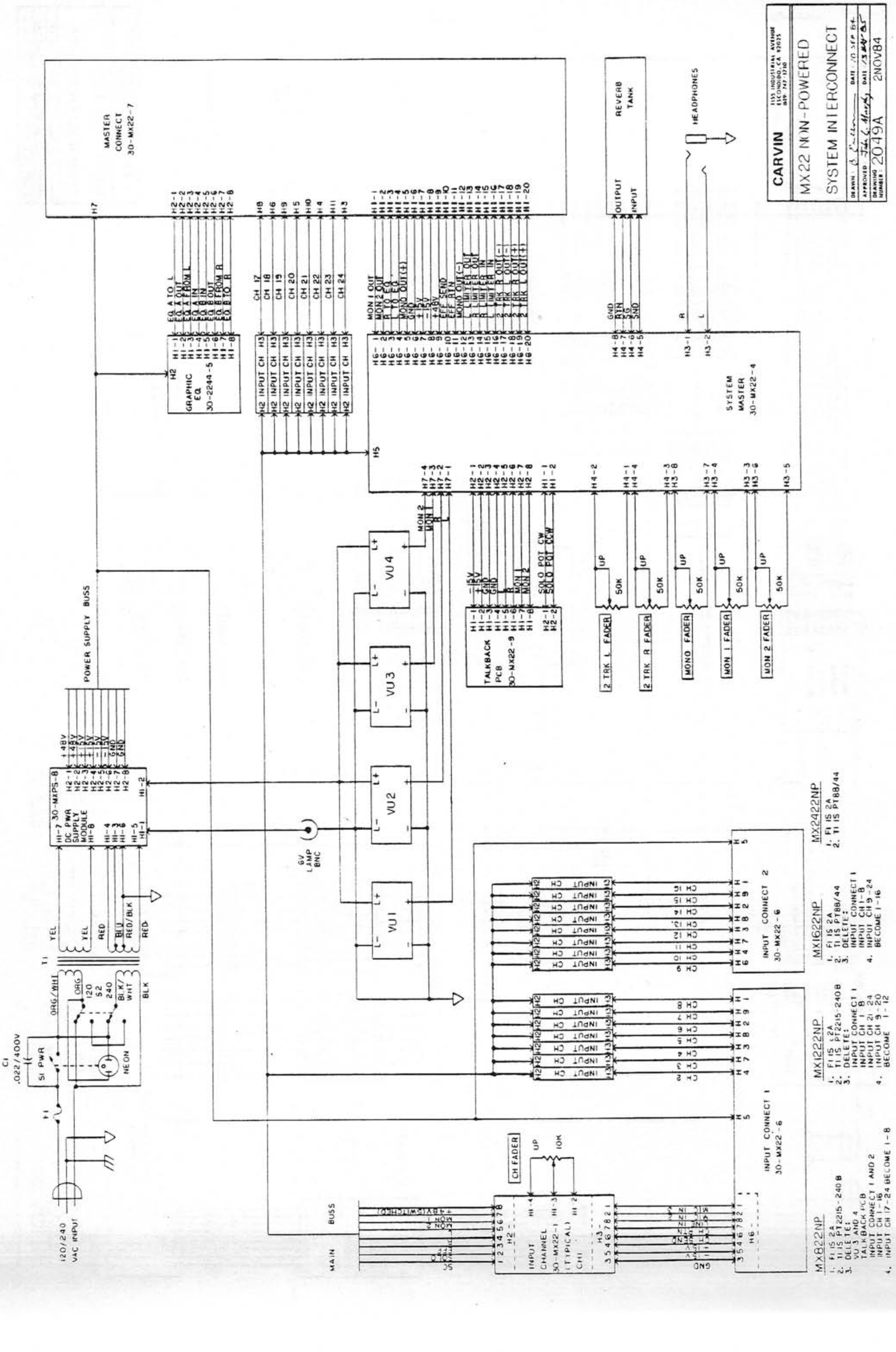
B. MX1222

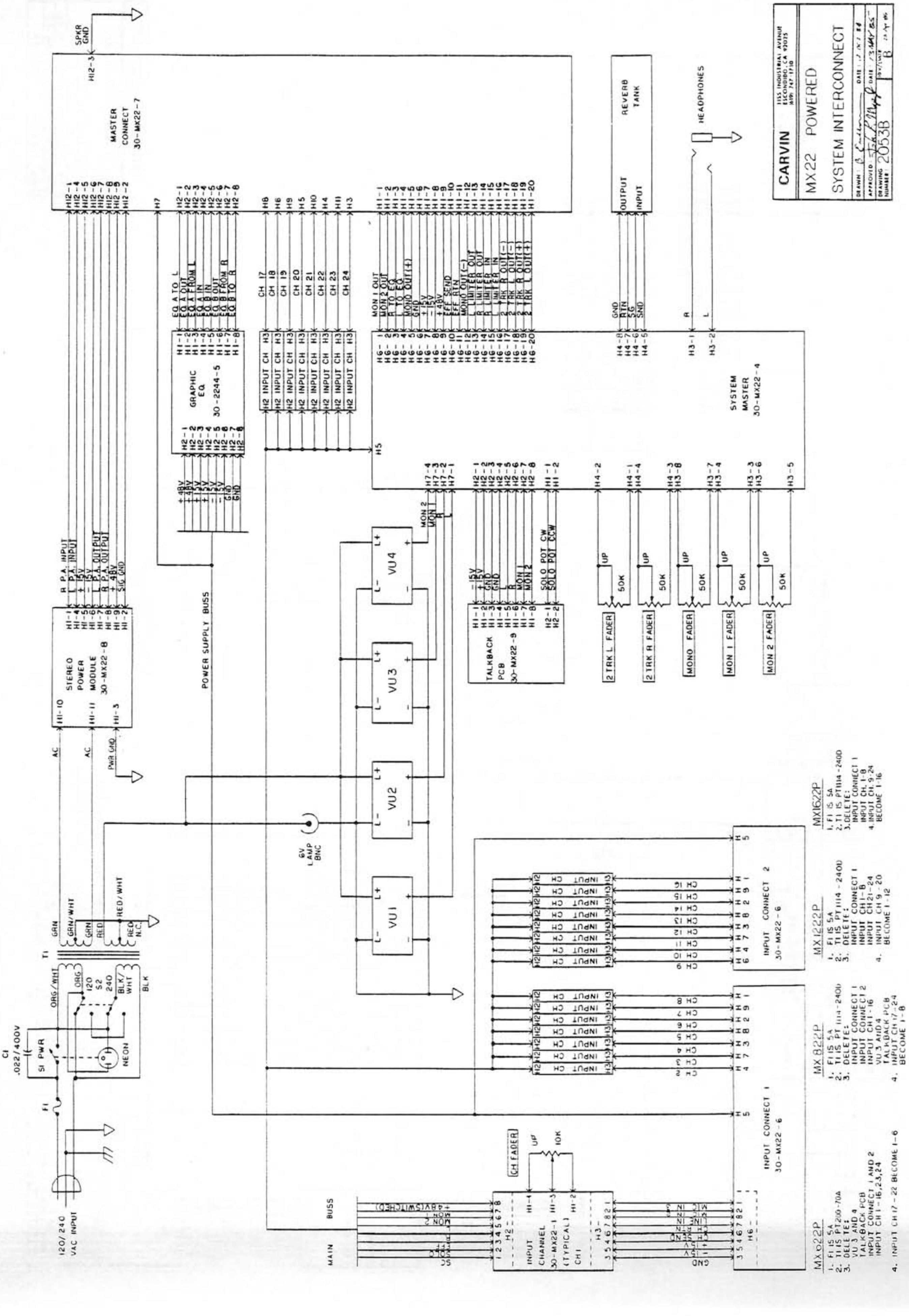
1. DELETE: J13 - J24
   J37 - J42
   PT1 - PT2
   R15 - R22
   C1 - C4
   H3 - H4
   H10 - H12
2. CH1 - 4 BECOME CH9 - 12

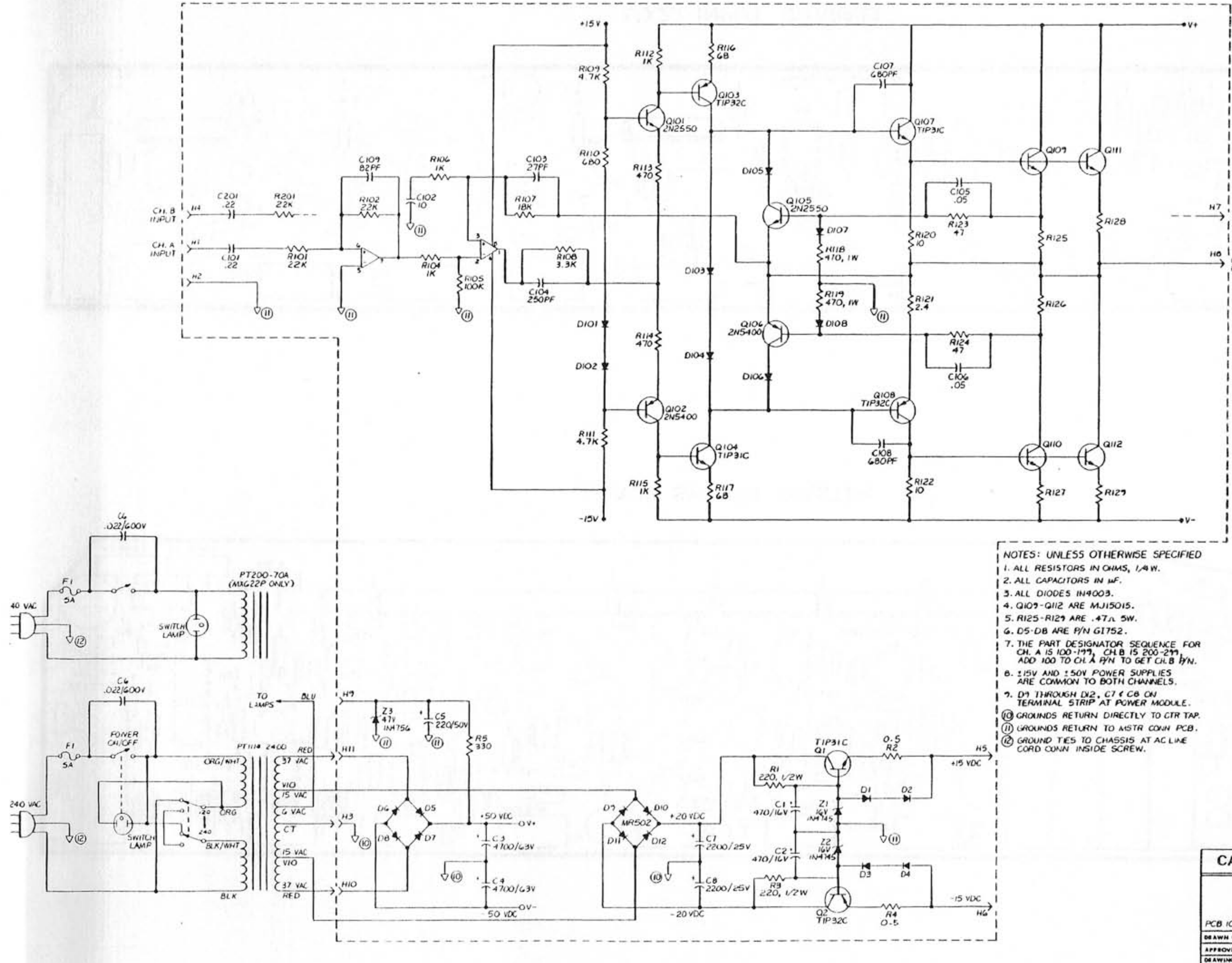
E. MX1222P

1. DELETE: J13 - J24
   H3 - H4
   H10 - H11
2. CH1 - 4 BECOME CH9 - 12

CARVIN	1155 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 (619) 747-1710
MX22 MASTER CONNECT	
30- MX22 - 7A	
DRAWN: B. Ellis	DATE: 26 SEP 84
APPROVED: [illegible]	DATE: 14 MAR 85
DRAWING NUMBER: 20518	REV: B


CARVIN
MX22 NON-POWERED
SYSTEM INTERCONNECT
2049A
2NOV84
MASTER CONNECT 30-MX22-7
REVERB TANK
HEADPHONES
GRAPHIC EQ 30-2244-5
SYSTEM MASTER 30-MX22-4
TALKBACK PCB 30-MX22-9
DC PWR SUPPLY MODULE 30-MXPS-8
POWER SUPPLY BUSS
VU 1
VU 2
VU 3
VU 4
2 TRK L FADER
2 TRK R FADER
MONO FADER
MON 1 FADER
MON 2 FADER
INPUT CONNECT 1 30-MX22-6
INPUT CONNECT 2 30-MX22-6
INPUT CHANNEL 30-MX22-1 (TYPICAL) CH1
CH FADER
120/240 VAC INPUT
MX822NP
MX1222NP
MX1622NP
MX2422NP


CARVIN
MX22 POWERED
SYSTEM INTERCONNECT
2053B
MASTER CONNECT 30-MX22-7
REVERB TANK
HEADPHONES
SYSTEM MASTER 30-MX22-4
GRAPHIC EQ 30-2244-5
STEREO POWER MODULE 30-MX22-8
POWER SUPPLY BUSS
TALKBACK PCB 30-MX22-9
VU1
VU2
VU3
VU4
6V LAMP BNC
2 TRK L FADER
2 TRK R FADER
MONO FADER
MON 1 FADER
MON 2 FADER
INPUT CONNECT 1 30-MX22-6
INPUT CONNECT 2 30-MX22-6
INPUT CHANNEL 30-MX22-1 (TYPICAL) CH1
CH FADER
120/240 VAC INPUT
S1 PWR
NEON
MX622P
MX822P
MX1222P
MX1622P



NOTES: UNLESS OTHERWISE SPECIFIED

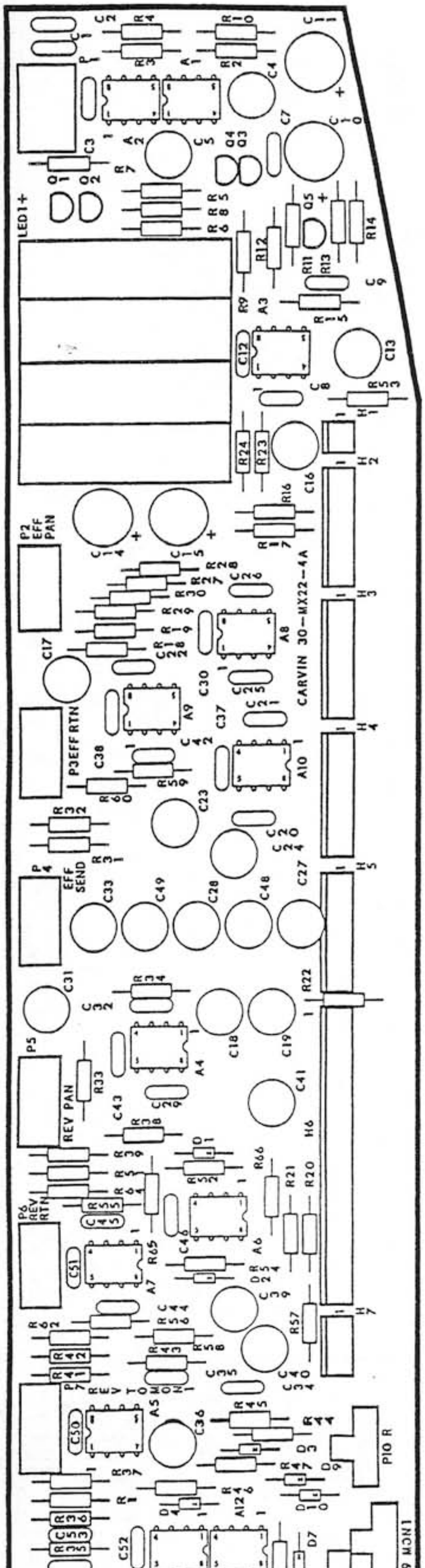
1. ALL RESISTORS IN OHMS, 1/4 W.
2. ALL CAPACITORS IN µF.
3. ALL DIODES 1N4003.
4. Q109-Q112 ARE MJ15015.
5. R125-R129 ARE .47Ω 5W.
6. D5-D8 ARE P/N G1752.
7. THE PART DESIGNATOR SEQUENCE FOR CH. A IS 100-199, CH. B IS 200-299. ADD 100 TO CH. A P/N TO GET CH. B P/N.
8. ±15V AND ±50V POWER SUPPLIES ARE COMMON TO BOTH CHANNELS.
9. D9 THROUGH D12, C7 & C8 ON TERMINAL STRIP AT POWER MODULE.

(10) GROUNDS RETURN DIRECTLY TO CTR TAP.
(11) GROUNDS RETURN TO MSTR CONN PCB.
(12) GROUND TIES TO CHASSIS AT AC LINE CORD CONN INSIDE SCREW.

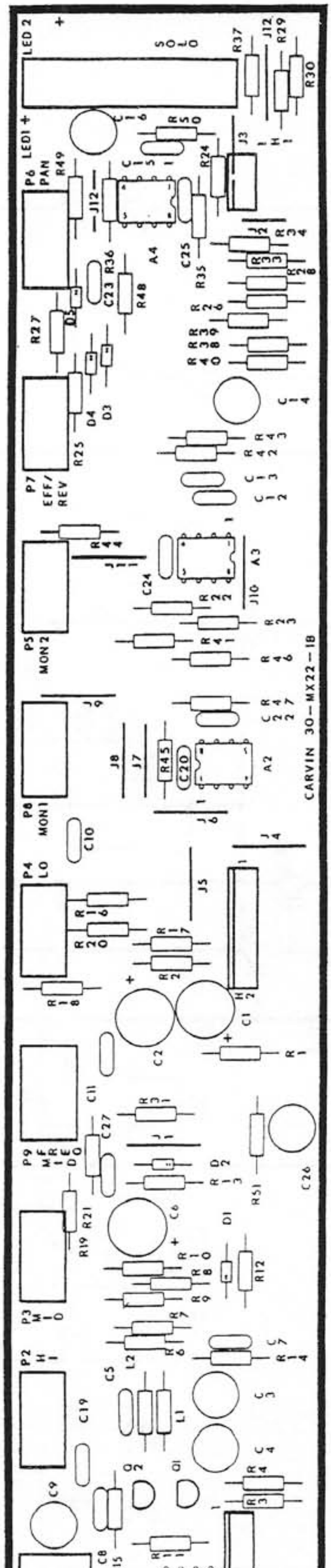
LAST USED (POWER SUPPLY)
R5
C8
D12
Q2
Z3

LAST USED (STEREO POWER MOD)
R129
C109
D108
Q112

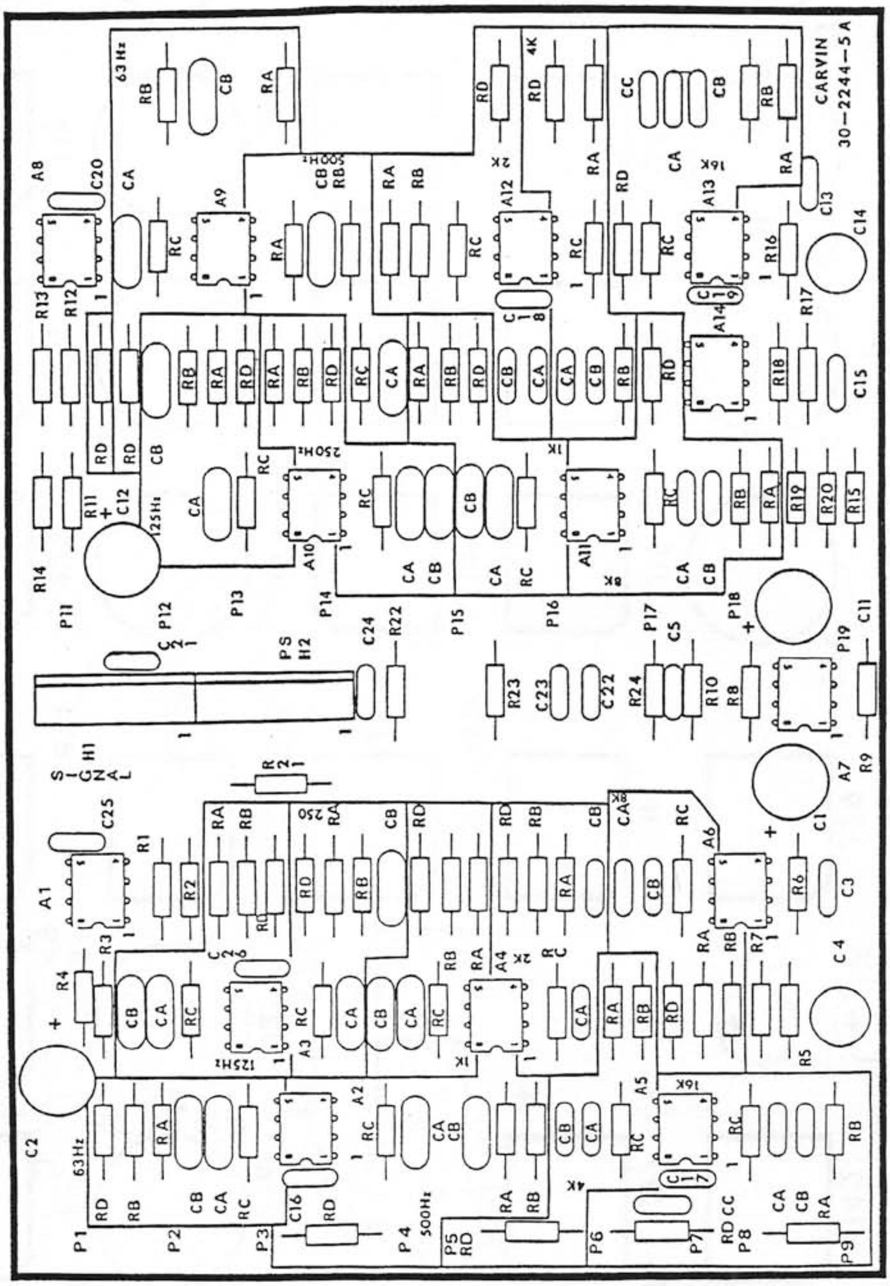
CARVIN	1551 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 619 747-1710
MX22 STEREO POWER MODULE	
PCB 1001D (50-MX22-B)	
DRAWN: S. W...	DATE: 16 MAY 85
APPROVED: John L. ...	DATE: 20 MAY 85
DRAWING NUMBER: 2054	REV/DATE A 20 MAY 86



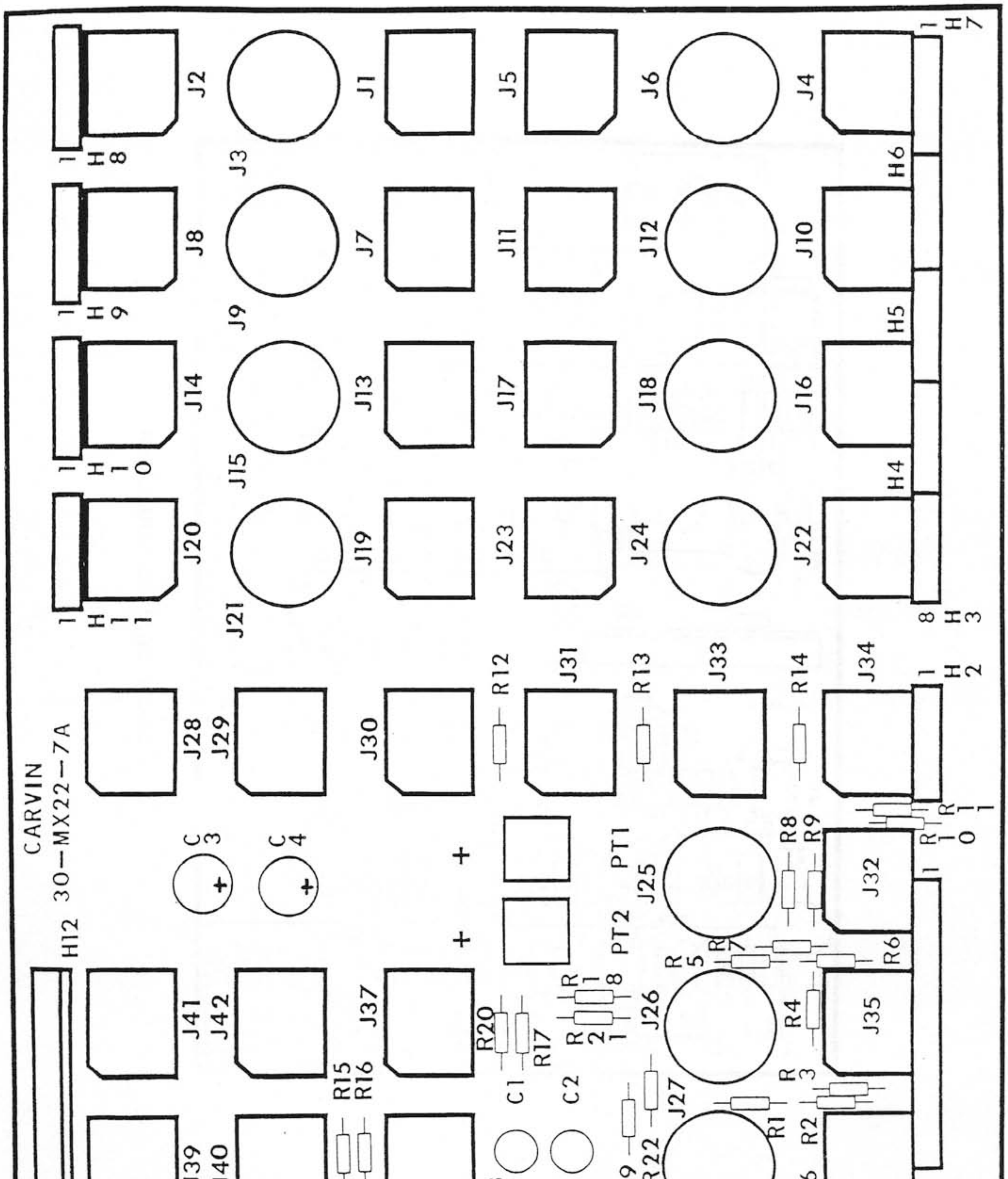
MX22 SYSTEM MASTER



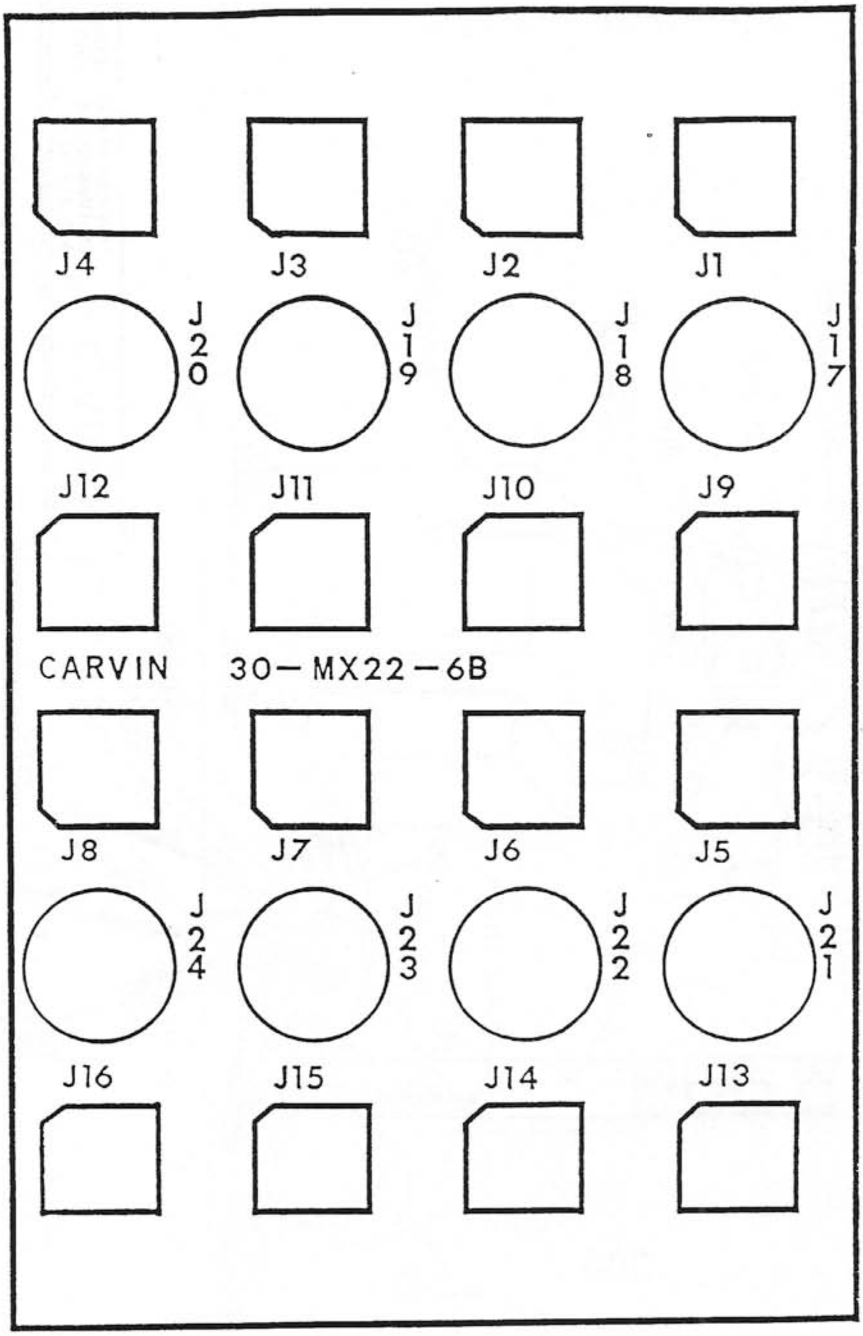
MX22 INPUT CHANNEL



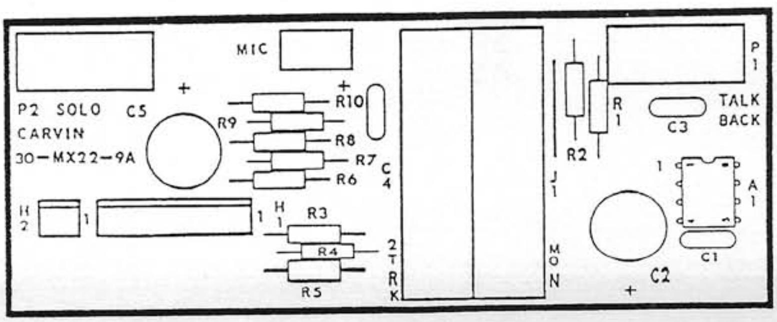
MX22 GRAPHIC EQUALIZER



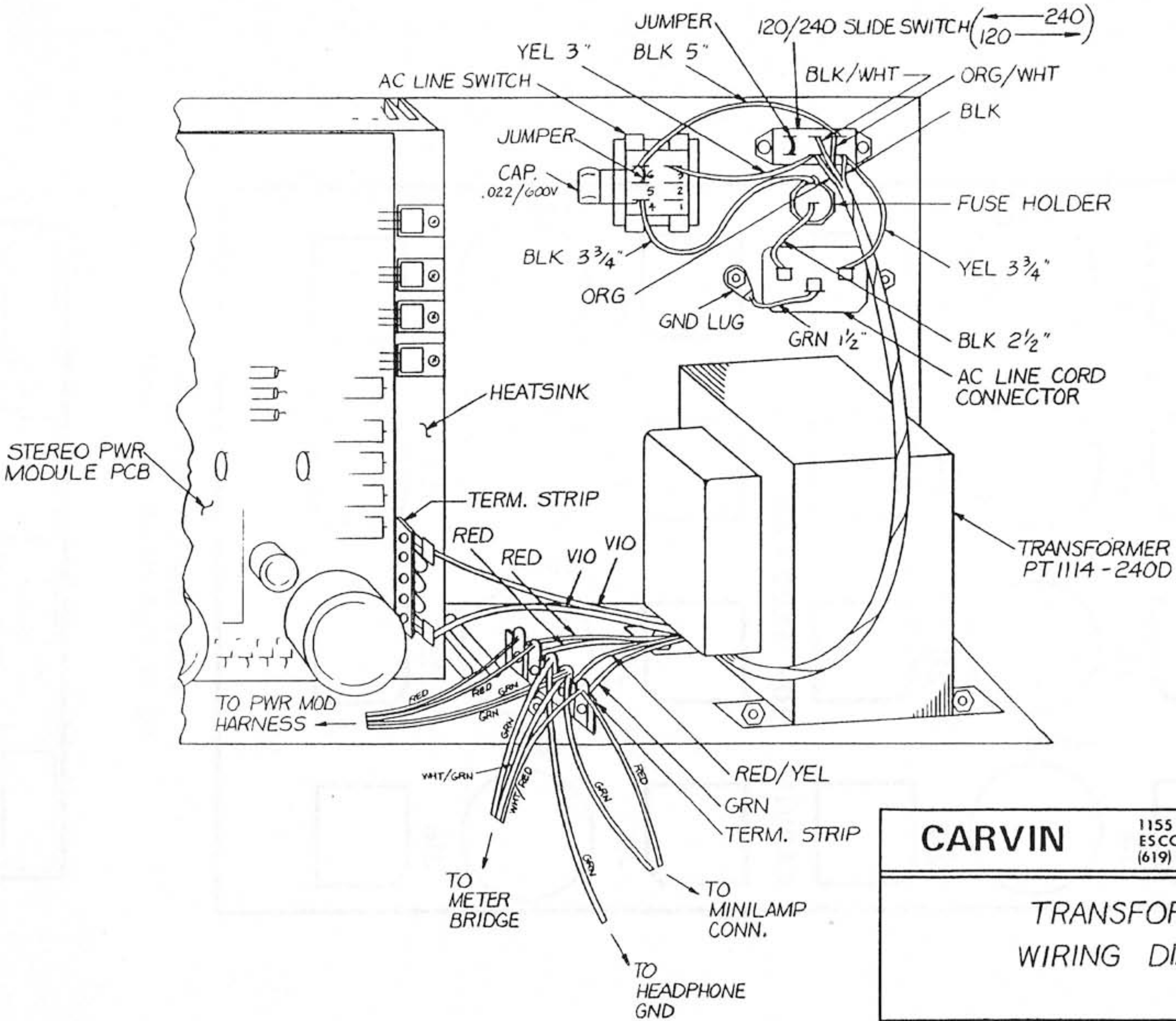
MX22 MASTER CONNECT



MX22 INPUT CONNECT
(12, 16 & 24 CH MODELS ONLY)

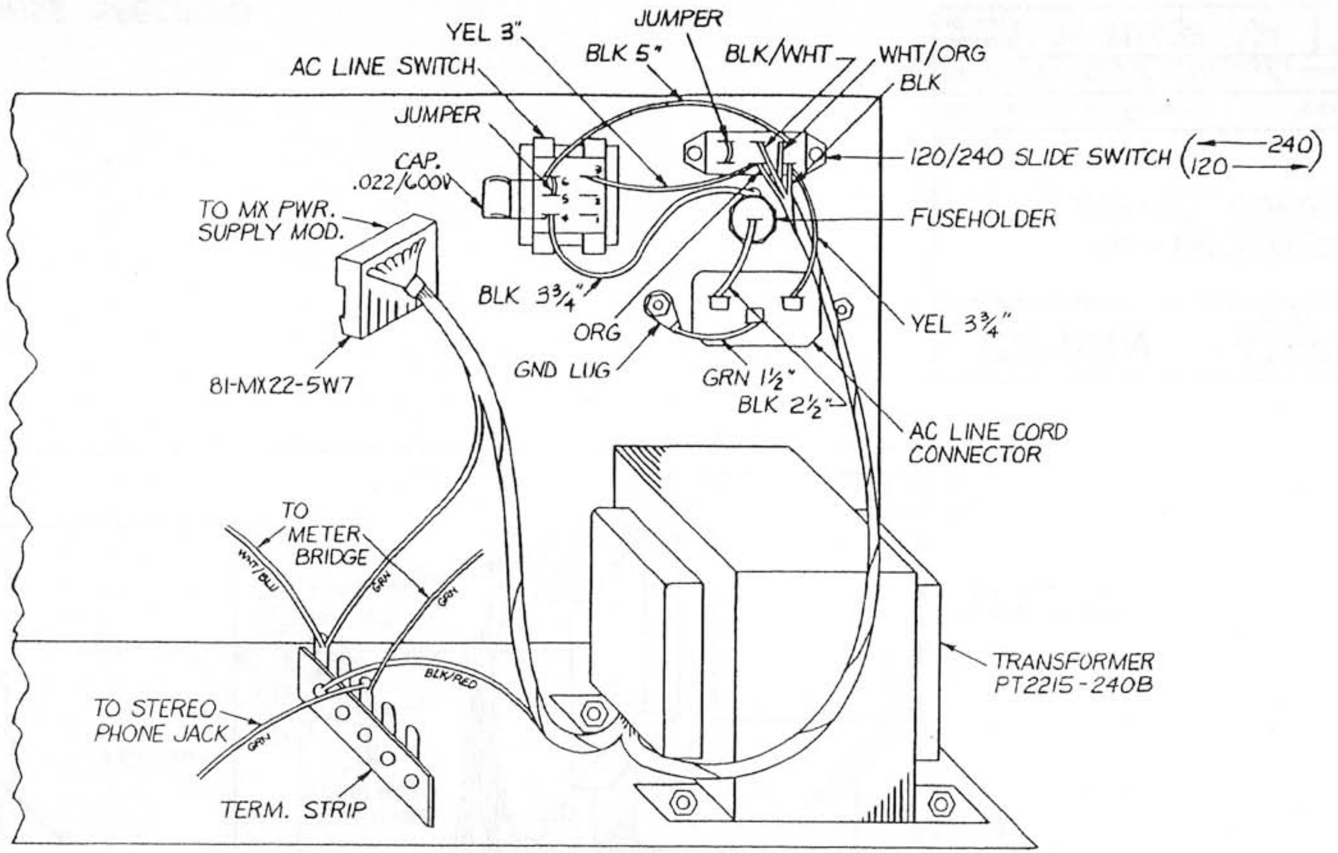


MX22 TALKBACK BOARD
(12, 16 & 24 CH MODELS ONLY)



CARVIN	1155 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 (619) 747-1710		
TRANSFORMER WIRING DIAGRAM (MX22)			
DRAWN : S. Mc Kenna		DATE : 28 Mar 85	
APPROVED : John L. Murphy		DATE : 28MAR85	
DRAWING NUMBER :	85-MX22-1114XF		

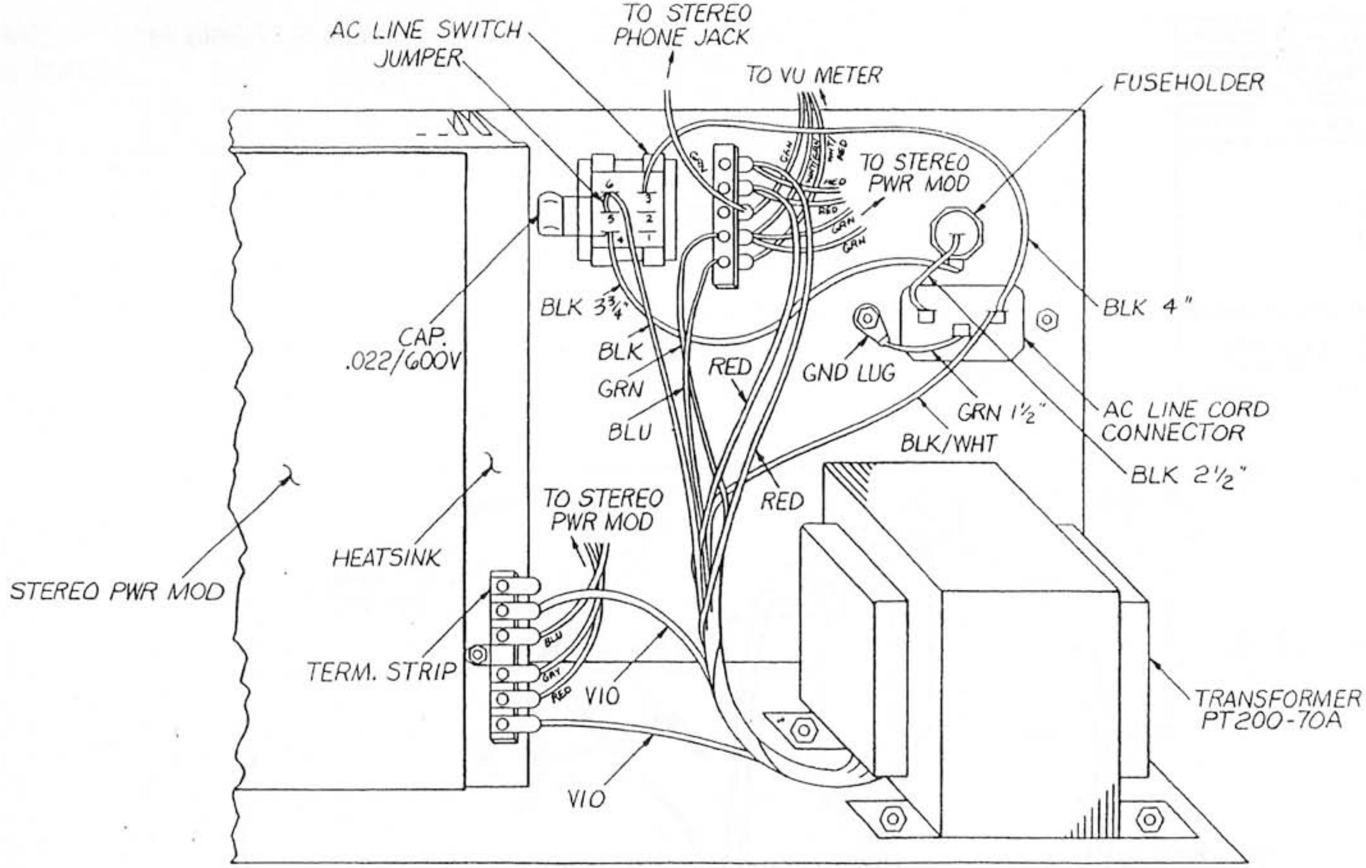
1. DO NOT SCALE.
NOTES: UNLESS OTHERWISE SPECIFIED



CARVIN	1155 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 (619) 747-1710
TRANSFORMER WIRING DIAGRAM (MX22)	
DRAWN: S. McKenna	DATE: 22 Apr 85
APPROVED: John L. Murphy	DATE: 23 APR 85
DRAWING NUMBER: 85-MX22-2215XF	REV/DATE

NOTES: UNLESS OTHERWISE SPECIFIED

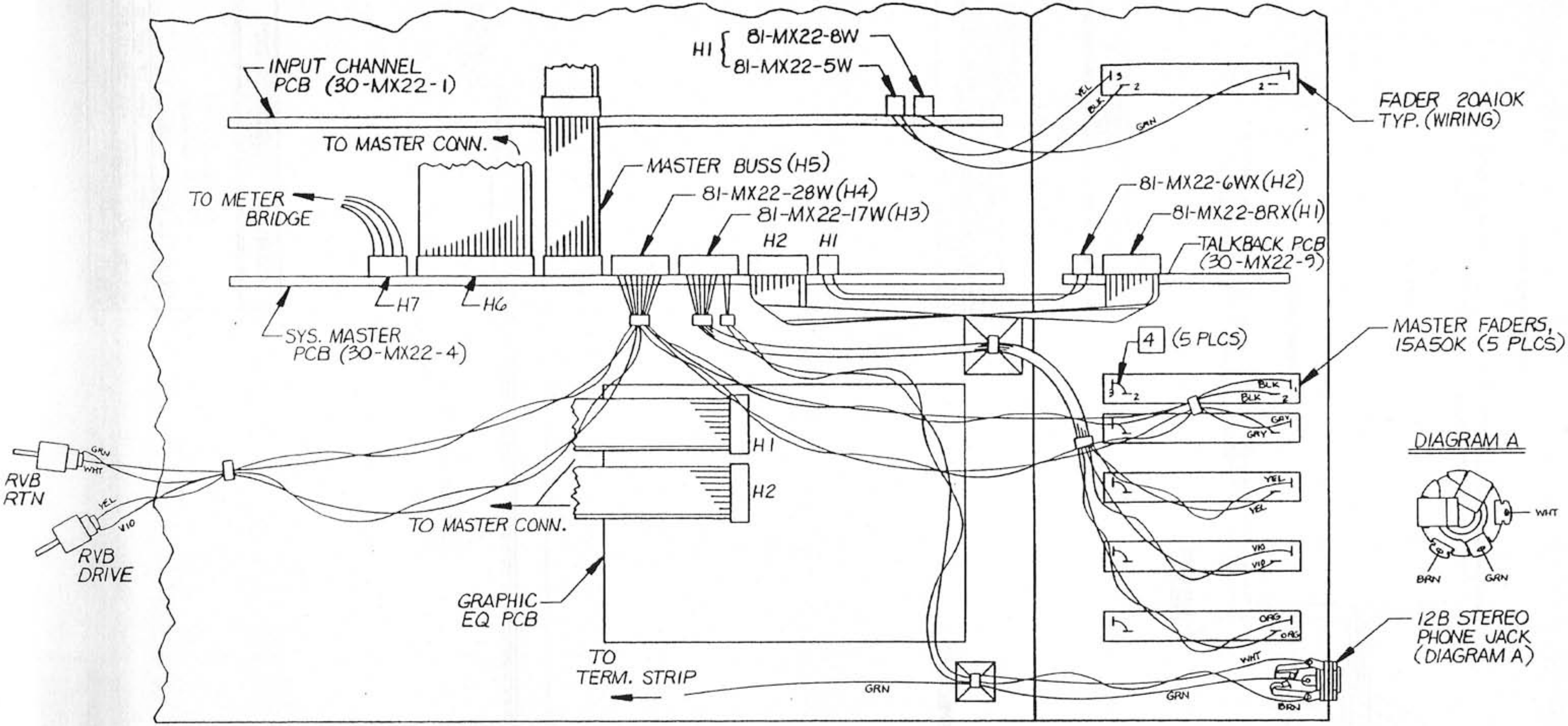
1. DO NOT SCALE.



CARVIN	1155 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 (619) 747-1710
TRANSFORMER WIRING DIAGRAM (MX622P)	
DRAWN: S. McKenna	DATE: 22 May 85
APPROVED: John L. Murphy	DATE: 23 MAY 85
DRAWING NUMBER: 85-MX622-70A	REV/DATE

NOTES: UNLESS OTHERWISE SPECIFIED

1. DO NOT SCALE.



CARVIN	1155 INDUSTRIAL AVENUE ESCONDIDO, CA 92025 (619) 747-1710
CHANNEL FADER WIRING DIAGRAM (MX22)	
DRAWN: S. McKenna	DATE: 26 Mar 85
APPROVED: John L. Murphy	DATE: 28 MAR 85
DRAWING NUMBER: 85-MX22-FDRS	

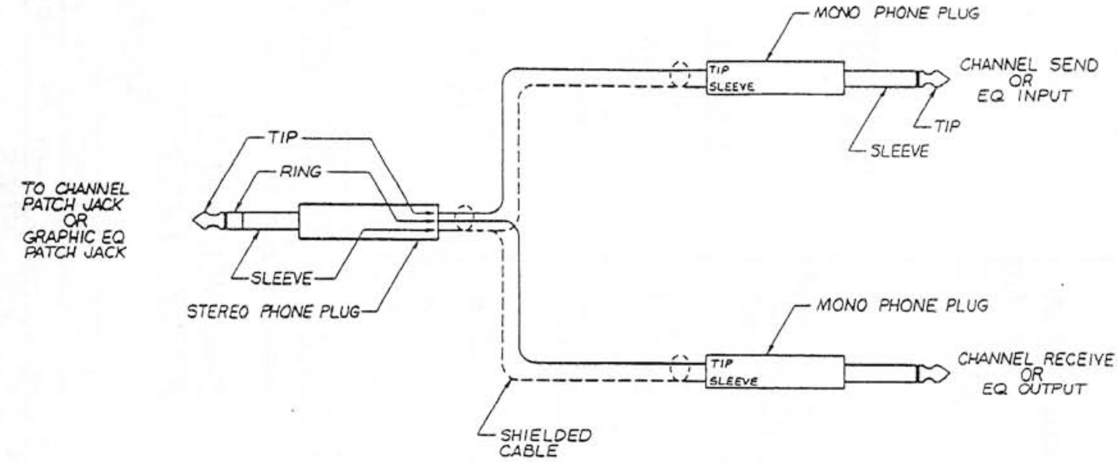
4 TERMINALS 2 & 3 AT THE MASTER FADERS ARE JUMPERED.

3. FOR MX822 AND MX822P DELETE TALKBACK PCB & HARNESSES.

2. FOR MX622P DELETE TALKBACK PCB & HARNESSES AND GRAPHIC EQ PCB & HARNESSES.

1. DO NOT SCALE.

NOTES: UNLESS OTHERWISE SPECIFIED

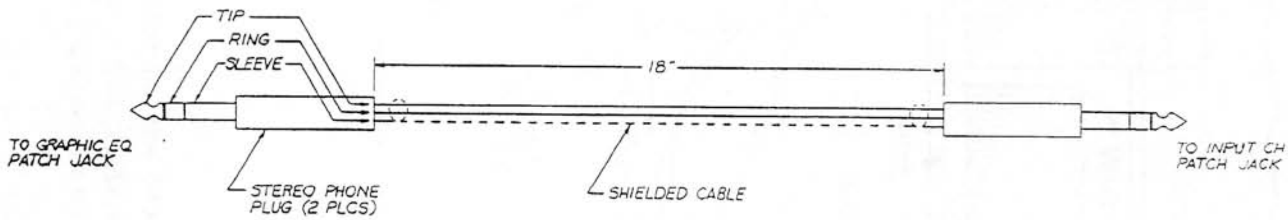


CARVIN
1155 INDUSTRIAL AVENUE
ESCONDIDO, CA 92025
(619) 747-1710

MX22 SERIES
PATCH CABLE
(SEND/RETURN)

DRAWN: S. McKenna DATE: 23 MAY 83
APPROVED: John L. Murphy DATE: 24 MAY 85
DRAWING NUMBER: 83-MX22-1 REV/DATE

1. DO NOT SCALE
NOTES: UNLESS OTHERWISE SPECIFIED

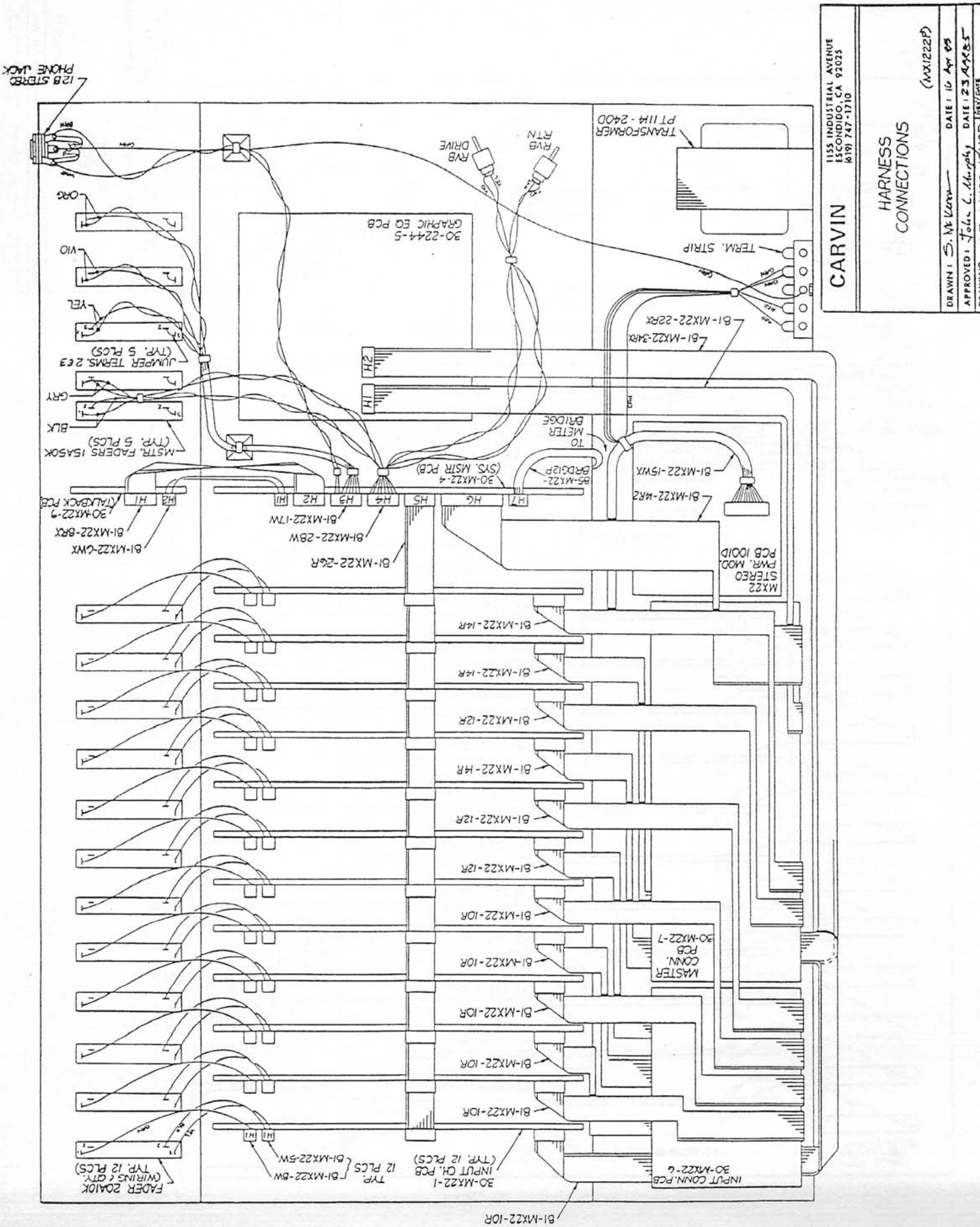


CARVIN
1155 INDUSTRIAL AVENUE
ESCONDIDO, CA 92025
(619) 747-1710

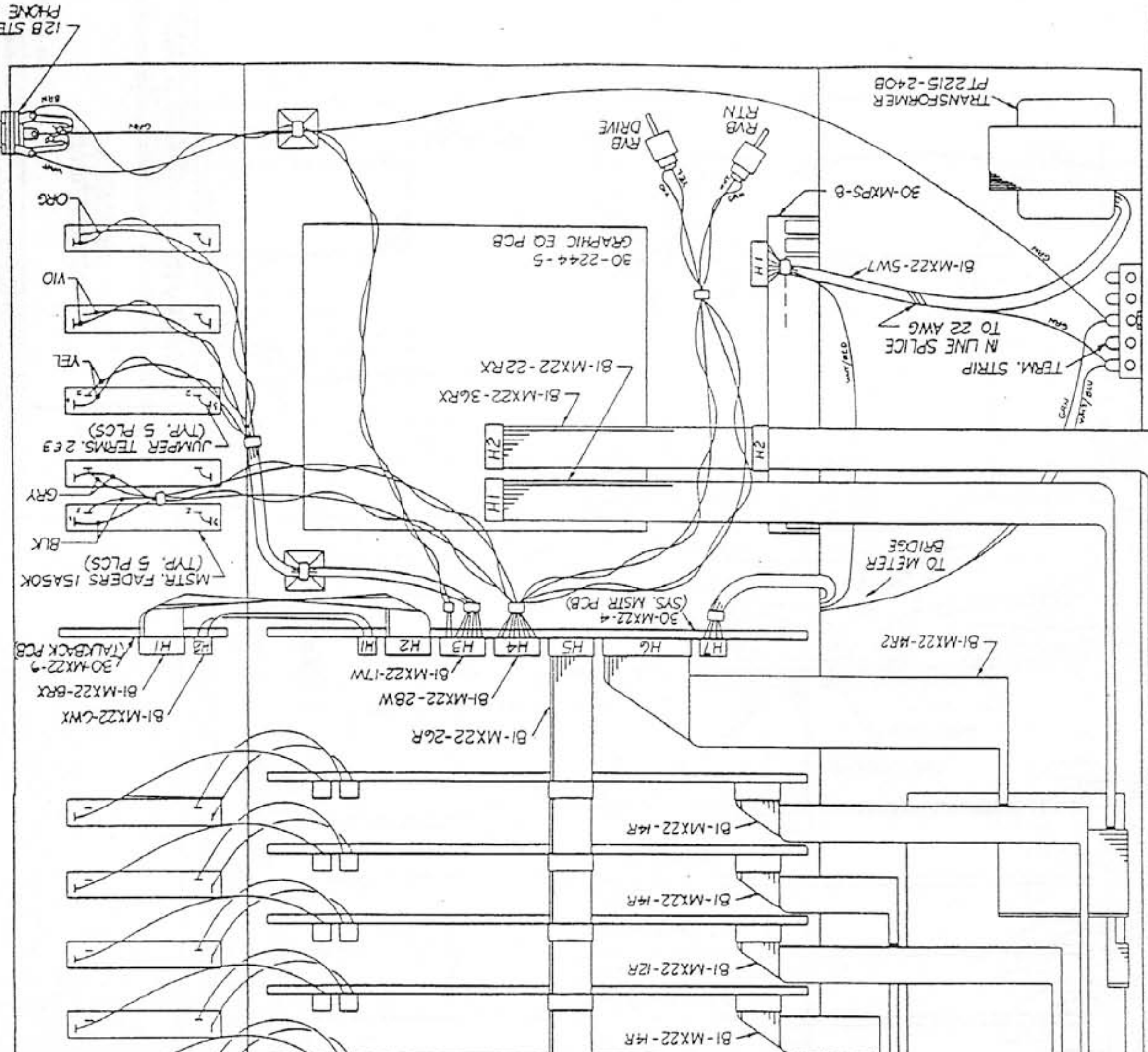
MX22 SERIES
PATCH CABLE
(GRAPHIC EQ PATCH TO INPUT CH)

DRAWN: S. McKenna DATE: 28 MAY 83
APPROVED: John L. Murphy DATE: 28 MAY 85
DRAWING NUMBER: 83-MX22-2 REV/DATE

1. DO NOT SCALE.
NOTES: UNLESS OTHERWISE SPECIFIED



NOTES: UNLESS OTHERWISE SPECIFIED
1. DO NOT SCALE.



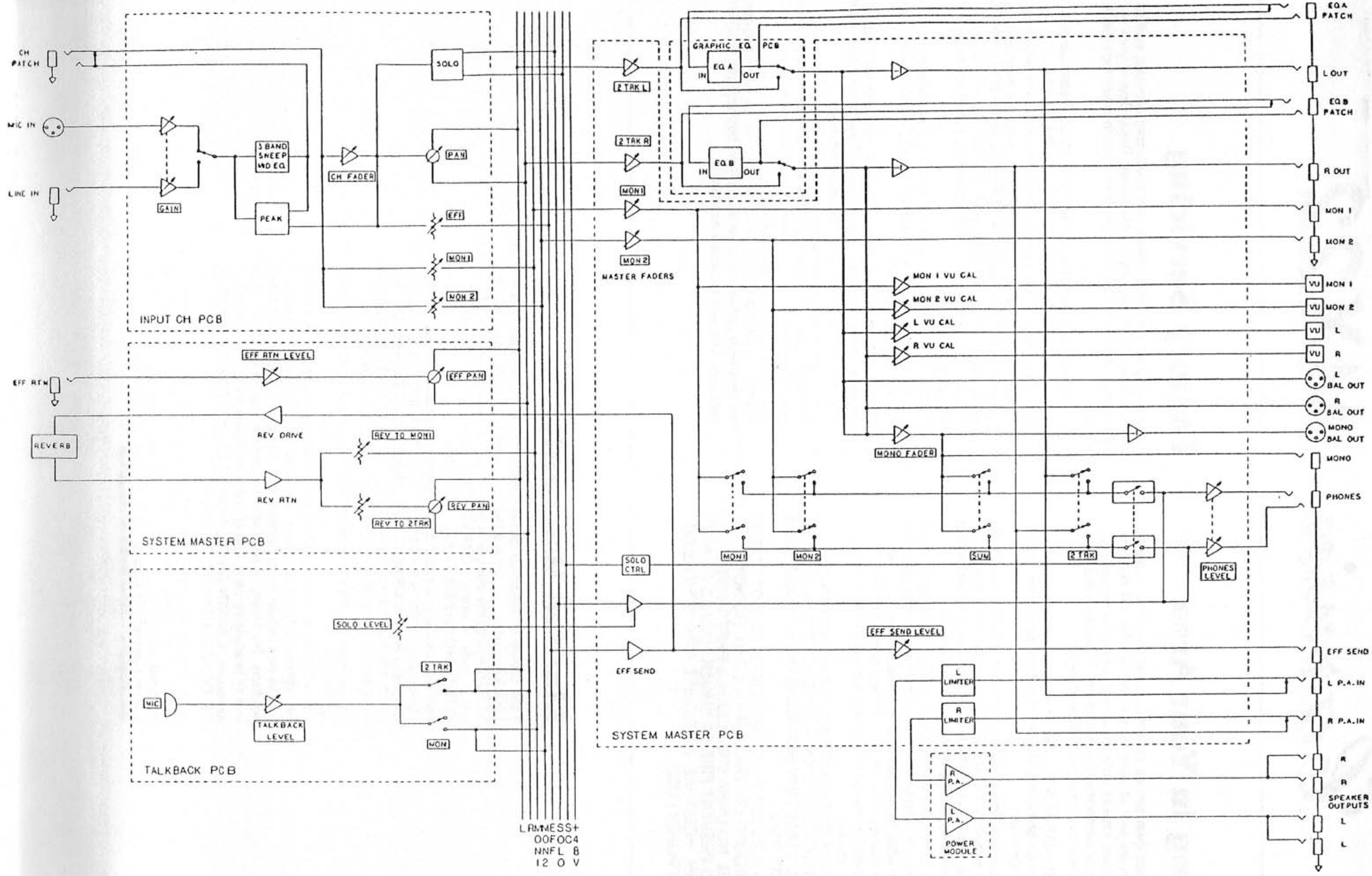
CARVIN
1155 INDUSTRIAL AVENUE
ESCONDIDO, CA 92025
(619) 747-1710

HARNESS CONNECTIONS

(MX1222)

DRAWN: S. McKenna DATE: 16 Apr 85
APPROVED: John L. Murphy DATE: 23 MAR 85
DRAWING NUMBER: 85-MX22-12HC REV/DATE

NOTES: UNLESS OTHERWISE SPECIFIED
1. DO NOT SCALE.



NOTES:

1. MX622 - DELETE TALKBACK.
2. MX822 - DELETE TALKBACK.
3. ALL - POWER MODULE INSTALLED IN POWERED UNITS ONLY (P SUFFIX ON MODEL NO).

CARVIN	
MX22 SERIES	
BLOCK DIAGRAM	
DRAWN:	DATE: 1 OCT 84
APPROVED:	DATE:
DRAWING NUMBER: 2052B	B

Carvin MX22

Servicing in Your Area

You may select your own service center or have your own qualified technician work on the unit at your own expense. This will not void the warranty for future repairs unless damage was done because of improper servicing or component replacement. If damage was done, a normal fee for parts and servicing will be charged.

Under the 1 YEAR WARRANTY, Carvin will ship parts pre-paid to you or your technician providing that the defective part(s) are first returned for our inspection.

If you do not have a qualified service person, we ask that you do not involve yourself in servicing the unit. By sending the unit back to us, you may save time, money, and frustration. Also, you will know that your unit was serviced according to factory specifications.

If it is necessary to have your unit serviced locally, we strongly recommend that you have your technician call us before servicing your unit. We find that those who do this are able to make necessary repairs faster, and for less money. We are glad to help in this manner because we have pride in our products and we want them to work properly for many years.

REMINDER: CARVIN DOES NOT PAY FOR SERVICING OR PARTS OTHER THAN OUR OWN — NO EXCEPTIONS. IF YOU ELECT TO HAVE YOUR OWN SERVICING DONE, THESE BILLS MUST BE PAID BY YOU.

CAUTION — TO PREVENT ELECTRIC SHOCK DO NOT DEFEAT THE SAFETY GROUND ON THE POWER CORD. DO NOT REMOVE COVER. NO USER-SERVICEABLE PARTS INSIDE. WARNING — TO PREVENT FIRE OR SHOCK HAZARD DO NOT EXPOSE TO RAIN, MOISTURE, EXPLOSIVE ATMOSPHERE OR INSTALL AN IMPROPER FUSE!

Factory Servicing

We highly recommend utilizing our specialized servicing staff to bring your unit up to factory specifications. Regardless of your warranty status, please follow these guidelines when returning units for service:

1. Enclose a full description of the malfunction. Please use the "Service Authorization Form" included with this manual.
2. Include a copy of the original invoice to verify your warranty.
3. Return the product in its original carton with the original packing material. NEITHER CARVIN NOR THE SHIPPING COMPANY WILL ASSUME LIABILITY FOR IMPROPERLY PACKED UNITS. Ship the unit by UPS if possible. You must pre-pay the shipping cost.
4. Please allow 5 working days for servicing plus shipping time to and from destination. All repairs in by MONDAY will be ready by the following MONDAY.
5. Carvin will pre-pay the shipping back to you providing the unit is covered under warranty. If you wish to have it sent back by AIR, you will be required to pay the difference COD.
6. If your unit is out of warranty, you will be charged a modest fee (generally lower than typical repair shops). You must pay shipping charges both ways. These charges will be collected COD.
7. If in doubt about the malfunction, please call a Carvin salesman toll-free at 800-854-2235 (in Calif. 800-542-6070). Occasionally we receive merchandise that works fine, but because of an oversight, the unit was returned needlessly.

Limited Warranty

Your Carvin Professional Series Product is protected against failure for 1 YEAR. Carvin will service the unit, supply all parts, and pay the RETURN shipping charges at no charge to the customer providing the unit is under warranty. CARVIN WILL NOT PAY FOR PARTS OR SERVICING OTHER THAN OUR OWN.

This warranty is extended to the original purchaser only and is not transferable. THIS WARRANTY DOES NOT INCLUDE FAILURES CAUSED BY INCORRECT USE, INADEQUATE CARE OF THE UNIT, OR NATURAL DISASTERS. A COPY OF THE ORIGINAL INVOICE IS REQUIRED TO VERIFY YOUR WARRANTY.

Carvin takes no responsibility for any horn driver or speaker damaged by this unit.

This warranty is in lieu of all other warranties, expressed or implied. No representative or person is authorized to represent or assume for Carvin any liability in connection with the sale or servicing of Carvin products. No liability is assumed for damage due to accident, abuse, lack of reasonable care, loss of parts, or failure to follow Carvin's directions. CARVIN SHALL NOT BE LIABLE FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES.

In the interest of creating new products and improving existing ones, Carvin is continually researching the latest state of the art audio design methods, and modern packaging and production techniques. Thus, Carvin reserves the right to make changes in its products and specifications without notice or obligation.

12340 World Trade Drive., San Diego, CA 92128
800-854-2235