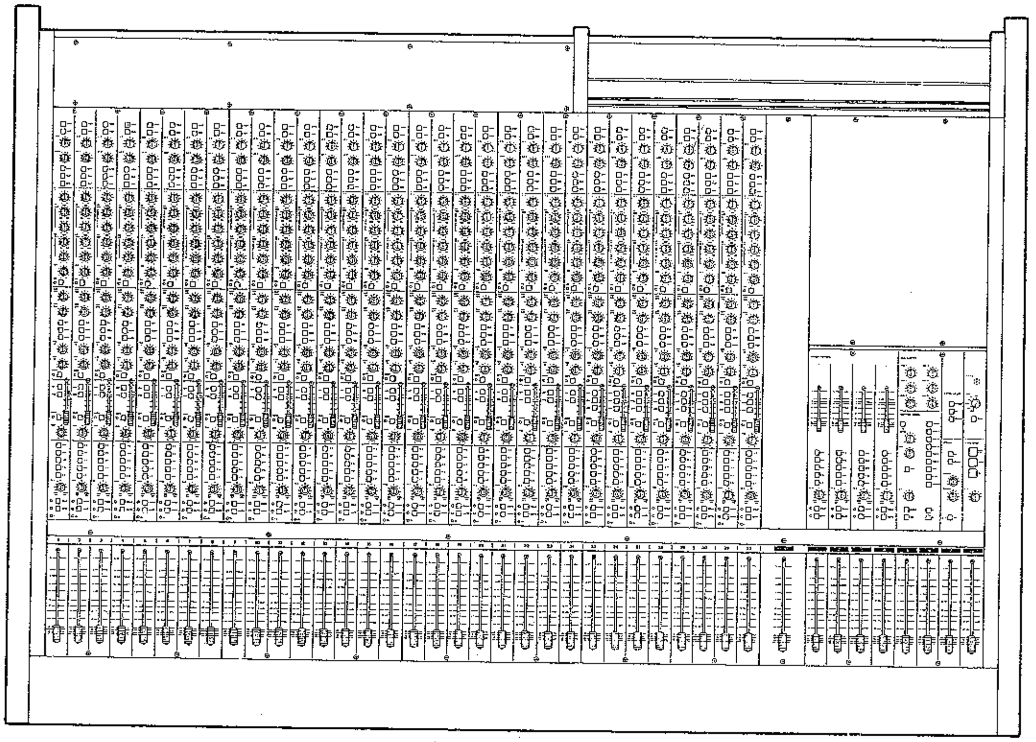
TASCAM
TEAC Professional Division

M-3500 Series

Mixing Console



OPERATION/MAINTENANCE

5700119900

TABLE OF CONTENTS

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER-SERVICEABLE PARTS INSIDE REFER SERVICING TO QUALIFIED SERVICE PERSONNEL

The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance

This appliance has a serial number located on the rear panel Please record the model number and serial number and retain them for your records.
Model number____________
Serial number____________

WARNING: TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

SAFETY INSTRUCTIONS

Before anything else, please remember that the M-3500 is an electrical device and should be treated with respect Read the following precautions. Don't take chances with your life or the lives of those you work with

CAUTION:

- **Read all of these instructions.**
- **Save these instructions for later use.**
- **Follow all warnings and instructions marked on the audio equipment.**

1. **Read Instructions** – All the safety and operating instructions should be read before the appliance is operated
2. **Retain Instructions** – The safety and operating instructions should be retained for future reference.
3. **Heed Warnings** – All warnings on the appliance and in the operating instructions should be adhered to.
4. **Follow Instructions** – All operating and use instructions should be followed.
5. **Water and Moisture** – The appliance should not be used near water – for example, near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool, etc.
6. **Carts and Stands** – The appliance should be used only with a cart or stand that is recommended by the manufacturer

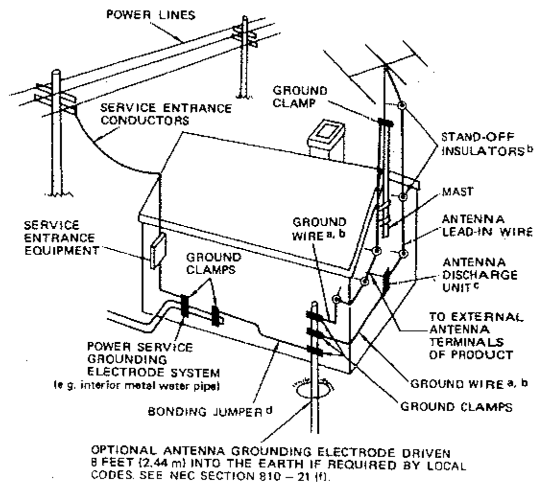
6A. An appliance and cart combination should be moved with care Quick stops, excessive force, and uneven surfaces may cause the appliance and cart combination to overturn

7. **Ventilation** – The appliance should be situated so that its location or position does not interfere with its proper ventilation. For example, the appliance should not be situated on a bed, sofa, rug, or similar surface that may block the ventilation openings; or, placed in a built-in installation, such as a bookcase or cabinet that may impede the flow of air through the ventilation openings

   Heat – The appliance should be situated away from heat sources such as radiators, heat registers, stoves or other appliances (including amplifiers) that produce heat

9. **Power Sources** – The appliance should be connected to a power supply only of the type described in the operating instructions or as marked on the appliance.
10. **Grounding or Polarization** – The precautions that should be taken so that the grounding or polarization means of an appliance is not defeated
11. **Power-Cord Protection** – Power-supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords at plugs, convenience receptacles, and the point where they exit from the appliance.
12. **Cleaning** – The appliance should be cleaned only as recommended by the manufacturer.
13. **Power Lines** – Any devices connected to the audio system must be located away from power lines.
14. **Outdoor Antenna Grounding** – If an outside antenna is connected to the receiver be sure the antenna system is grounded so as to provide some protection against voltage surges and built up static charges. Section 810 of the National Electrical Code, ANSI/NFPA No 70 – 1984, provides information with respect to proper grounding of the mast and supporting structure, grounding of the lead-in wire to an antenna discharge unit, size of grounding conductors location of antenna-discharge unit, connection to grounding electrodes, and requirements for the grounding electrode. See Figure below

EXAMPLE OF ANTENNA GROUNDING ACCORDING TO NATIONAL ELECTRICAL CODE INSTRUCTIONS CONTAINED IN ARTICLE 810 – RADIO AND TELEVISION EQUIPMENT"



a Use No 10 AWG (5 3 mm²) copper, No 8 AWG (8.4 mm²) aluminum, No. 17 AWG (1.0 mm²) copper-clad steel or bronze wire or larger as a ground wire.

b Secure antenna lead-in and ground wires to house with stand-off insulators spaced from 4 feet (1 22 m) to 6 feet (1.83 m) apart

c Mount antenna discharge unit as close as possible to where lead-in enters house

d Use jumper wire not smaller than No. 6 AWG (13.3 mm²) copper, or the equivalent, when a separate antenna-grounding electrode is used See NEC Section 810-21(j)

15. **Nonuse Periods** – The power cord of the appliance should be unplugged from the outlet when left unused for a long period of time
16. **Object and Liquid Entry** – Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.
17. **Damage Requiring Service** – The appliance should be serviced by qualified service personnel when:
    A. The power-supply cord or the plug has been damaged; or
    B. Objects have fallen, or liquid has been spilled into the appliance; or
    C. The appliance has been exposed to rain; or
    D. The appliance does not appear to operate normally or exhibits a marked change in performance; or
    E. The appliance has been dropped, or the enclosure damaged.
18. **Servicing** – The user should not attempt to service the appliance beyond that described in the operating instructions. All other servicing should be referred to qualified service personnel

Introduction

Thank you for choosing the TASCAM M-3500 mixing console. The M-3500 is the latest in a long line of the most successful consoles in the world. It was designed for maximum flexibility without compromising performance in any way. This flexibility increases its value to you, however it may also make it difficult to understand at first. With study and experience, you will find the M-3500 readily understandable, easy to use, and adaptable to many different situations.

Using the manual: This manual is a reference book. If you read it thoroughly at least once, you will know where to turn when you need answers. Not all of this information will necessarily apply to your studio, but understanding all the options will help improve your sound.

The M-3500 is available in either a 32 or a 24 input configuration. All information contained in this manual is valid for both models unless otherwise specified.

Unfortunately, it is impossible for this manual to cover in detail every possible operation and connection of the M-3500. Every system is a little different, and so the manual covers only the typical applications. Also, this manual is not intended as a complete course in audio engineering. If you need help beyond this manual, turn to your TASCAM dealer. There are also books and magazines that can provide valuable information to help you.

It is easier to understand if you set up the system and can experiment with the console as you read. Don't make the mistake of booking a crucial recording session before you've had free time to really get to know your M-3500.

Use of Capital Letters: In general, we use all upper case type to designate a particular switch, control or jack name or label (like PAD). All upper case type is also used to clearly distinguish an item from other similar ones even if that item is not actually so labeled on the console (like CHANNEL FADER, as distinct from MONITOR FADER or other faders).

Recording is an art as well as a science. A successful recording is often judged primarily on the quality of sound as art, and we obviously cannot guarantee that. Your skill as a technician and your abilities as an artist will be significant factors in the results that you achieve. The M-3500 console will perform properly only if it is connected and operated as stated in this manual. We cannot guarantee your skill in adjustment or your technical comprehension of this manual. However, we want to do everything we can to help you get the most out of the M-3500. So, please, READ THE MANUAL! If you invest the time and are patient, you'll find it pays off in the long run.

NOTE FOR U.K. CUSTOMERS

U.K. Customers Only:

Due to the variety of plugs being used in the U.K., this unit is sold without an AC plug. Please request your dealer to install the correct plug to match the mains power outlet where your unit will be used as per these instructions.

IMPORTANT

The wires in this mains lead are coloured in accordance with the following code:

BLUE: NEUTRAL
BROWN: LIVE

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloures markings identifying the terminals of your plug, proceed as follows:

The wire which is coloured BLUE must be onnected to the terminal which is marked with the letter N or coloured BLACK. The wire which is coloured BROWN must be connected to the terminal which is marked with the letter L or coloured RED.

Bescheinigung des Herstellers/Importeurs

Hiermit wird bescheinigt, daß der/die/das

AUDIO MISCHPULT TASCAM M-3500

(Gerät Typ Bezeichnung)

in Übereinstimmung mit den Bestimmungen der

AMTSBLATT 163/1984, VFG 1045/1984

(Amtsblattverfügung)

funk-entstört ist

Der Deutschen Bundespost wurde das Inverkehrbringen dieses Gerätes angezeigt und die Berechtigung zur Überprüfung der Serie auf Einhaltung der Bestimmungen eingeräumt

TEAC CORPORATION

Name des Herstellers/Importeurs

This product is manufactured to comply with the radio interference of EEC directive "82/499/EEC."

Three Types of Controls

All mixers share some fundamental principles that are not hard to understand. No matter how many there are, or what company made them, each control on a mixer performs one or more of these three basic functions:

1. **Input selection**—where is the signal coming from?
2. **Output assignment**—where is the signal going to?
3. **Gain control**—how loud is the signal?

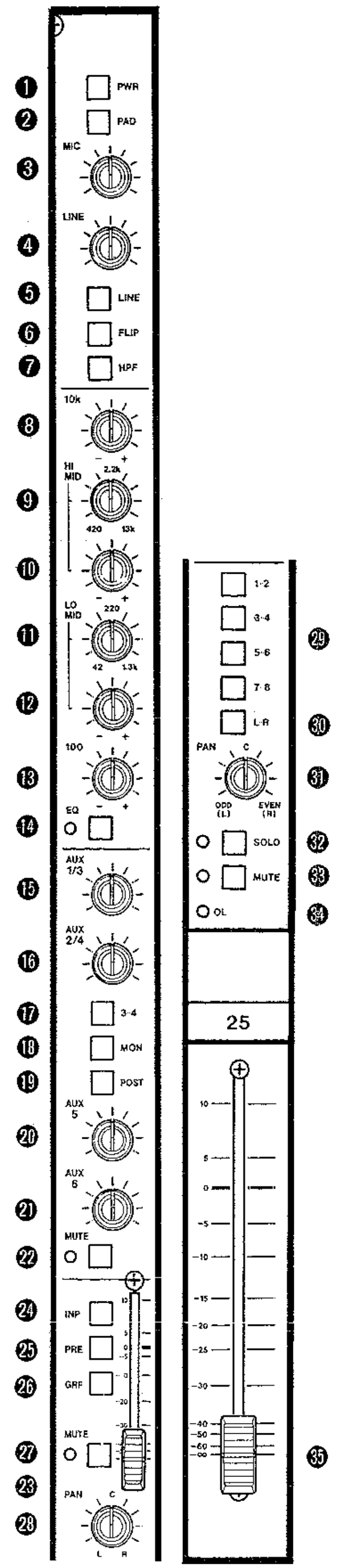
Keep these three types of controls (where from, where to, and how much) in your mind as you go through each control. For example, LINE is an input select switch—when it is pressed, signal comes from the line jack instead of the mic jack for that channel. The "1-2" switch is an output assign type—it sends the output of the channel to group outputs 1 and 2.

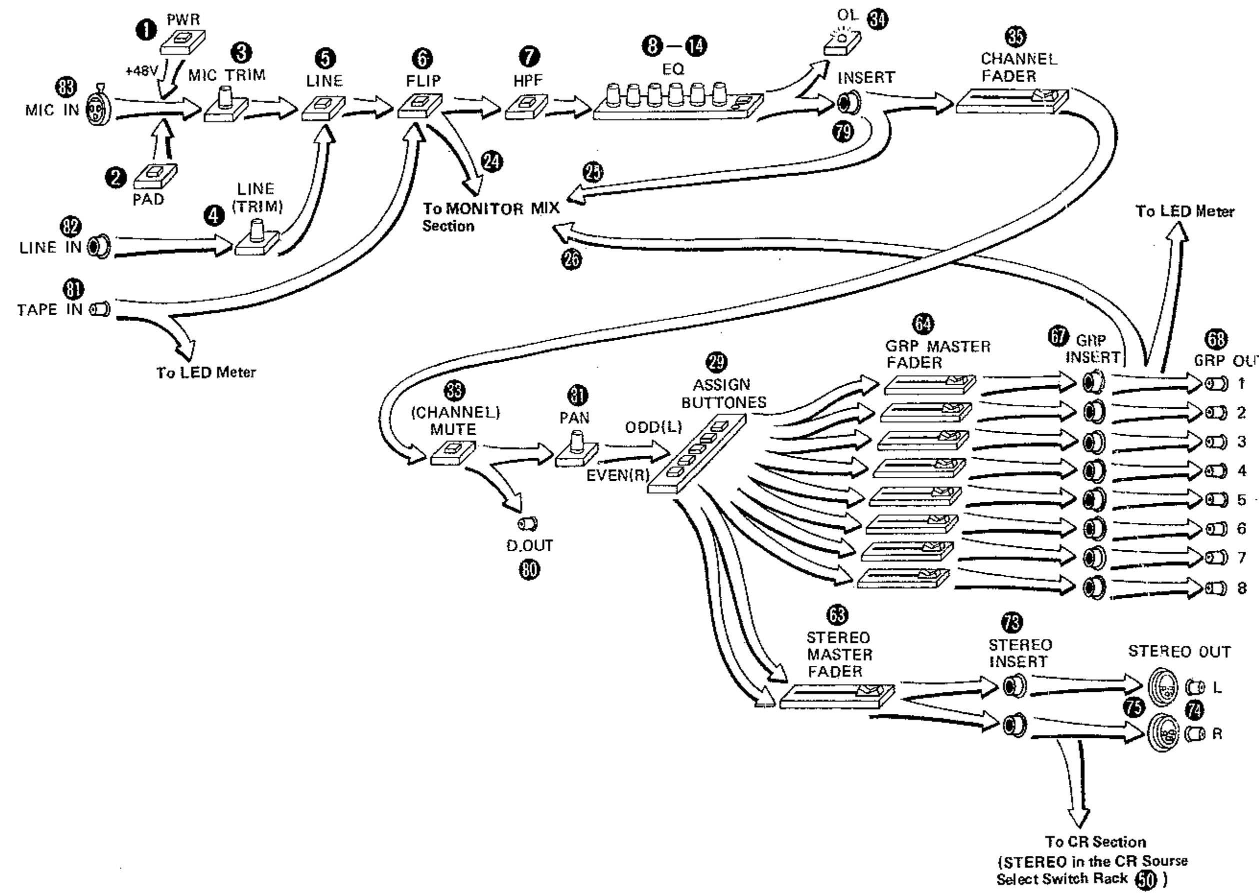
Signal Path

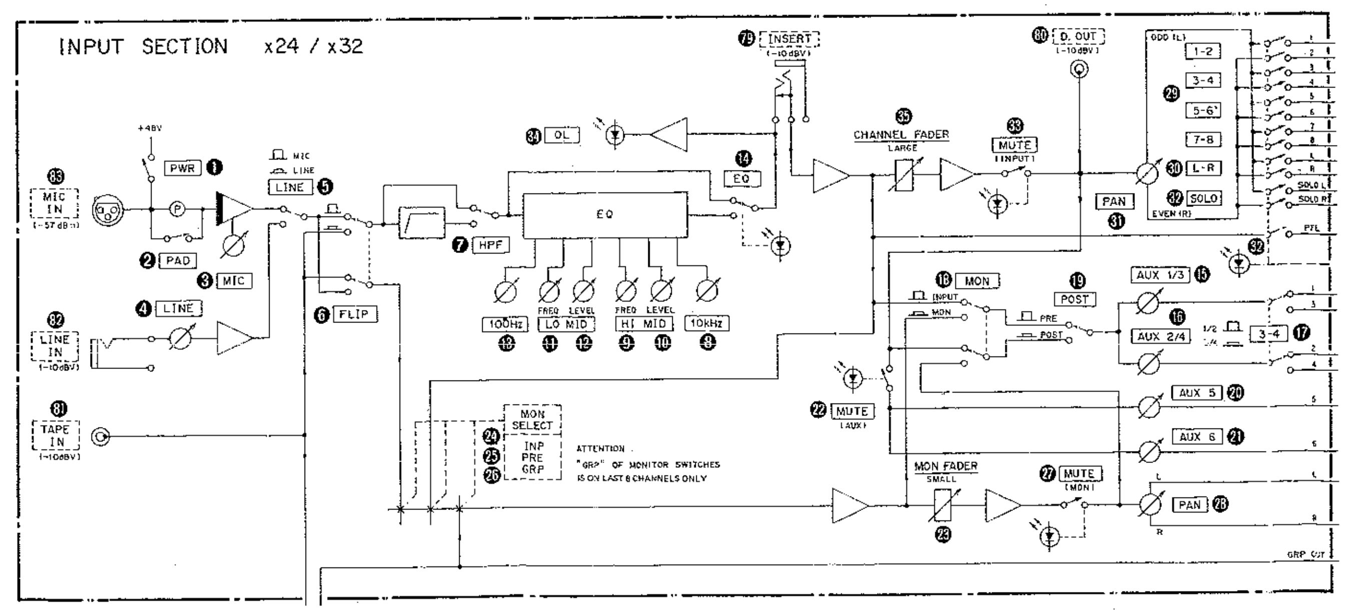
The path of signal flow through the M-3500 isn't obvious unless you read the block diagram, which is a "road map" of the actual internal signal paths. Every switch, knob, and jack of the mixer has its place on the block diagram. These three illustrations will help you to learn how to read the block, if you don't already know. First is the illustration of the top panel controls, with a number for each. The second is a pictogram, showing the controls laid out according to how they are wired internally. Note the numbers matching controls on one drawing to another. Finally is a section of the block diagram. Each symbol on the block stands for a jack or control you can see, or an amplifier stage (symbolized by the triangles). By following the signal flow through the diagram, you will be able to answer questions such as "is the level at the direct out jack controlled by the channel fader?" for yourself.

Keep in mind that each symbol in these drawings represents either a "where" or "how much" point. Using the numbers on these illustrations, you can see how a control knob on the first drawing relates to a function on the second, and a symbol on the third. Studying these relationships will help you learn to read the block diagram, on the next page. This is a valuable skill which will aid you in getting the most performance from your console. Experienced engineers can operate a sophisticated mixing console guided only by the block diagram.

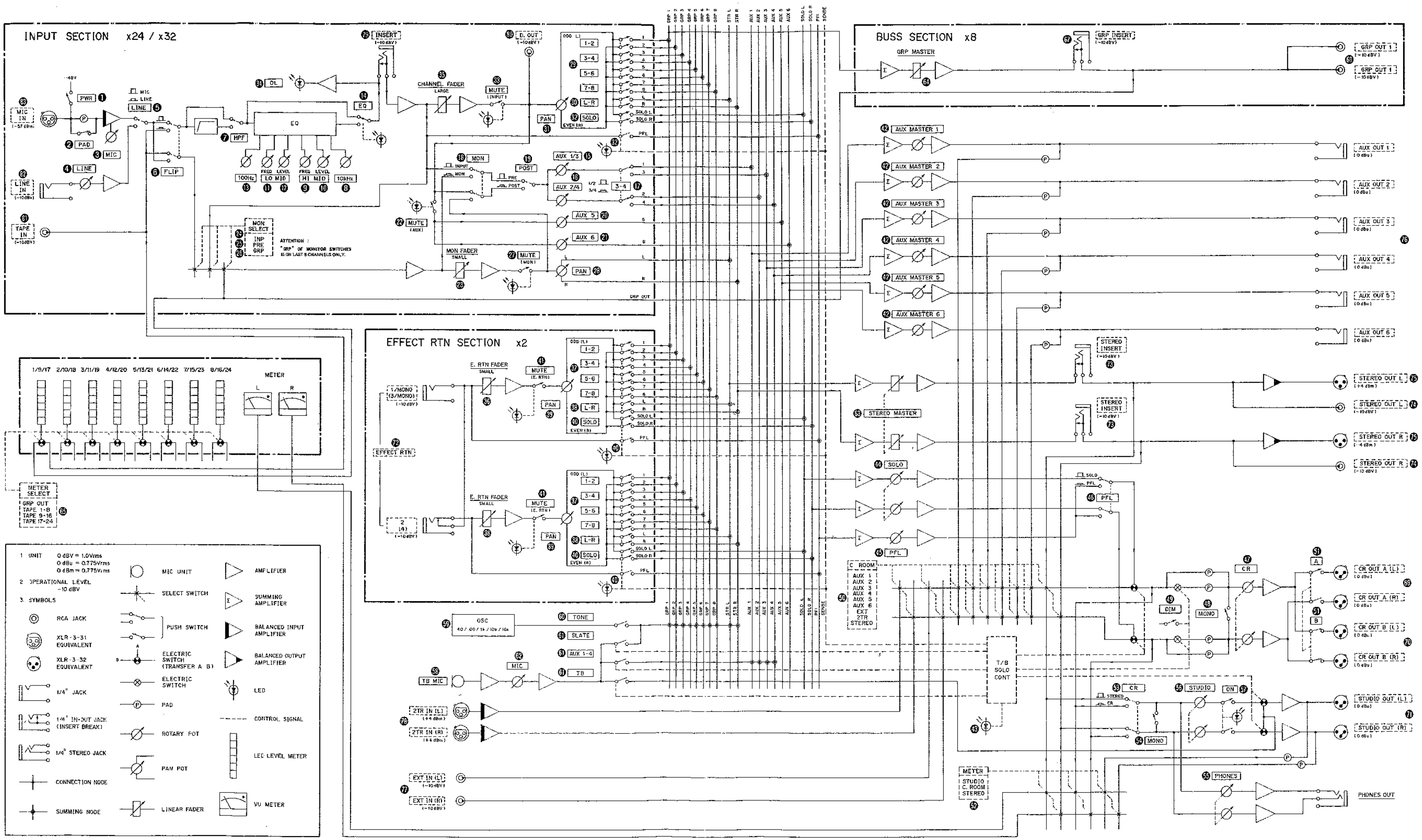
Not every channel is shown in the block—only one of each type. Running vertically down the block are lines called busses, which are connected to several different points. A buss is basically a wire running across the entire console, collecting signals from the different channels, like a river gathering strength from small streams flowing into it.



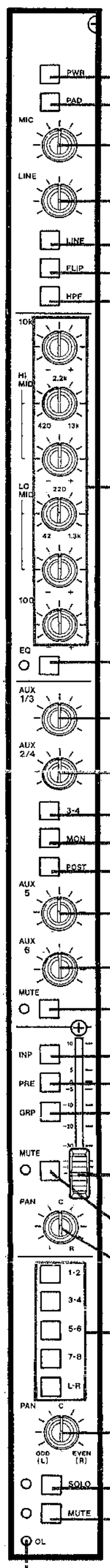




Signal Routing/Block Diagram



M-3500 Brief Guide



Input selection and adjustment

PWR: Applies 48 v. phantom power to the MIC (XLR) jack.

PAD: Puts a 30 dB pad on the MIC (XLR) input.

MIC: Sets the mic input level (trim). Input range 42 dB (-67 to -25 dBm).
At minimum gain, with the PAD switch on, a balanced line-level signal of +4 dBm can be connected, with an additional 25 dB of headroom available.

LINE (pot): Sets the Line input level (trim). The unbalanced line input can accommodate signals from nominal -20 to +20 dBV, again with 25 dB of headroom.

LINE (switch): Press to select LINE (instead of MIC) as the input SOURCE.

FLIP: This "flips" the source of the monitor INPut (which is usually tape) with the source of the main channel (which can be MIC or LINE). At mixdown, this allows you to use the monitor section (short faders) for MIDI controlled inputs or effects returns, doubling the total inputs to the stereo mix.

Equalization

HPF: High Pass Filter (80 Hz, 12 dB/octave slope), before the EQ section. It's available even when the EQ is switched off.

Equalizer: 4 band with 2 midrange frequency sweeps
10k: Shelving type treble control with shelving point at 10 kHz, can affect frequencies down to 5 kHz.
Hi Mid: Sweep frequency range is 420 Hz to 13 kHz.
Low Mid: Sweep frequency range is 42 Hz to 1.3 kHz.
100: Shelving type bass control with shelving point at 100 Hz, can affect frequencies up to 200 Hz.

EQ: Totally removes the equalizer from the signal path when off.

Auxiliary sends

AUXILIARIES 1-4: These can be used as:
- Pre fader sends off the channel OR monitor, for studio headphone mixes
- Post channel fader sends, for effects at mixdown
- Post monitor fader, for effects in the monitor

AUX 1/3: Level control for auxiliary sends 1 or 3 (set by switch below)

AUX 2/4: Level control for auxiliary sends 2 or 4 (set by switch below)

The three switches affect Auxiliaries 1-4 only:

3-4: Changes the destination of the AUX sends above it (from 1/2 to 3/4).

MON: Normally, the source of AUX 1-4 is the channel. Pressing MON selects the MONITOR (short fader) as the source instead. This allows you to put effects on your monitor mix, or to develop different headphone (cue) mixes based on the monitor source.

POST: This selects POST fader (either channel or monitor depending on the MON switch) as the source for the AUX controls above. When it is up, the source is pre-fader.

AUX 5: A post send from the main channel.

AUX 6: Another post send from the main channel.

MUTE (Aux): Cuts off signal to AUX 5 and 6 only.

In-line monitor section

Monitor fader: This short fader sets the level feeding the stereo mix. During recording it's used to develop a monitor mix (from the multitrack, the group outs, or from the channels). At mixdown it can also be used to bring additional inputs (MIDI instruments, effects returns) into the mix.

INP: (Input) When pressed, the monitor will get signal from an input, as set by the FLIP and LINE switches at the top of the channel:
Flip OFF: TAPE return input
Flip ON, Line OFF: MIC input
Flip ON, Line ON: LINE input

PRE: When pressed, the monitor gets its signal from the main channel (PRE-fader, post-EQ).

GRP (last 8 channels only): Press to select the corresponding GROUP output of the main mix as the monitor source. Useful for submixing as well as monitoring.

MUTE (Monitor): Cuts off signal from the monitor fader.

PAN (Monitor): Sets the pan position of the monitor in the stereo (L-R) mix.

Channel output section

ASSIGNMENT SWITCHES: Send the output of the channel to any of the eight output groups, or directly to the stereo (L-R) mix.

PAN (Main): Sets the pan position of the channel in the stereo mix, and between odd/even groups (1-2, etc.).

SOLO: Stereo solo-in-place. When on, cuts all other signals to the Control room section, without affecting the mix to the eight output groups or the stereo or auxiliary mixes. (When the PFL switch in the Control Room section is down, SOLO acts as a mono pre-fade listen.)

MUTE (Main): Cuts off signal to the output groups and stereo mix, the direct out jack, and any POST auxiliary sends.

OL. Overload indicator. Goes on 3 dB before the electronics of the channel will distort, so you can reset your trim levels properly. (Clipping is 25-28 dB over nominal level.)

Channel fader (not shown): Sets the main channel output level.

MASTER SECTION

EFFECT RETURN: The four dedicated effects returns can be soloed, muted, and assigned to any output groups or the stereo mix as required, just like a channel. You can print the returns to tape, or just hear them in the monitors.

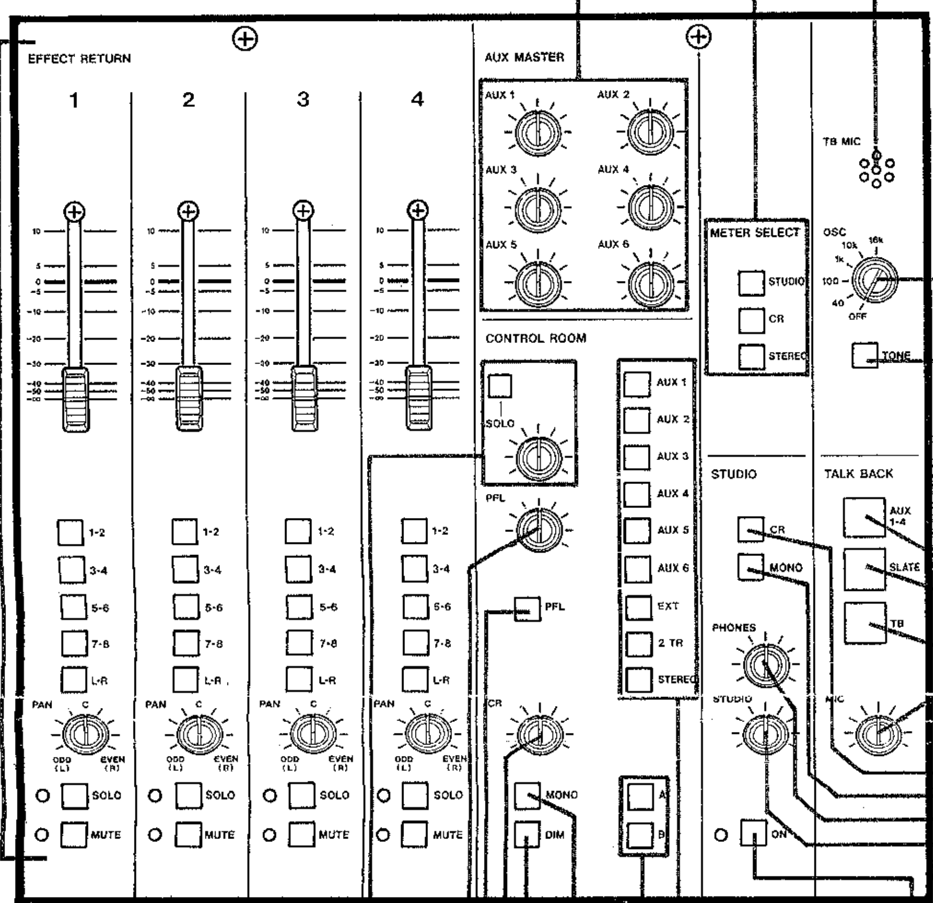
AUX MASTERS: These are the six master controls for the auxiliary outputs.

METER SELECT: The stereo VU meters at the head of the console can follow the **Stereo, Control Room,** or **Studio** mixes. In the Stereo position, Solo/PFL brings an individual channel up on the meter so that trims can be set easily.

MIC: This is a built-in talkback mic, which can communicate with different output feeds (see below).

OSC: Oscillator, frequencies 40, 100, 1k, 10k, and 16k Hz.

TONE: This applies the oscillator to all eight output groups and stereo sends.



Talkback Section

These control where the talkback mic is assigned:

Aux 1-4: For speaking to performers wearing headphones.

SLATE sends the mic to all 8 output groups and stereo sends for "slating" the tape.

TB assigns the mic to the STUDIO outputs.

MIC: This adjusts the volume of the talkback mic, wherever it is assigned.

STUDIO

A separate set of stereo outputs on the M-3500 is designed to be sent to a studio amplifier, for communication and playback.

CR: The studio outputs are usually fed with the Stereo mix. But when CR is pressed, the studio mix will be the same source as the control room.

MONO: This makes the studio mix monophonic.

PHONES: This is the volume control for the built-in headphone amp, which has the same mix as the STUDIO output (except for talkback).

STUDIO: This sets the level being sent to the studio outputs.

ON: This turns the studio outputs on and off. (The TALK-BACK switch can still address the studio, even if it's off).

CONTROL ROOM

SOLO: This control sets the volume in the control room when any SOLO switch is pressed in a channel (if the PFL swtich is up). "Solo" is stereo in-place, post-fader. The indicator will light if any SOLO switch is down.

PFL (level): Pre-Fade-Listen. This control sets the volume of the PFL feeding the control room speakers when a SOLO key is switched, if the PFL switch is down. PFL is similar to SOLO, but gets its signal from pre-fader so you can hear sources in the control room before bringing them into a mix, and is mono instead of stereo.

PFL (switch): Toggles the solo system between SOLO (Up) and PFL (Down) modes.

CR (Control Room): This is the master volume control for the feed to the control room monitor amp and loudspeakers.

MONO: This makes the Control room mix monophonic.

DIM: This lowers the control room level by 30 dB.

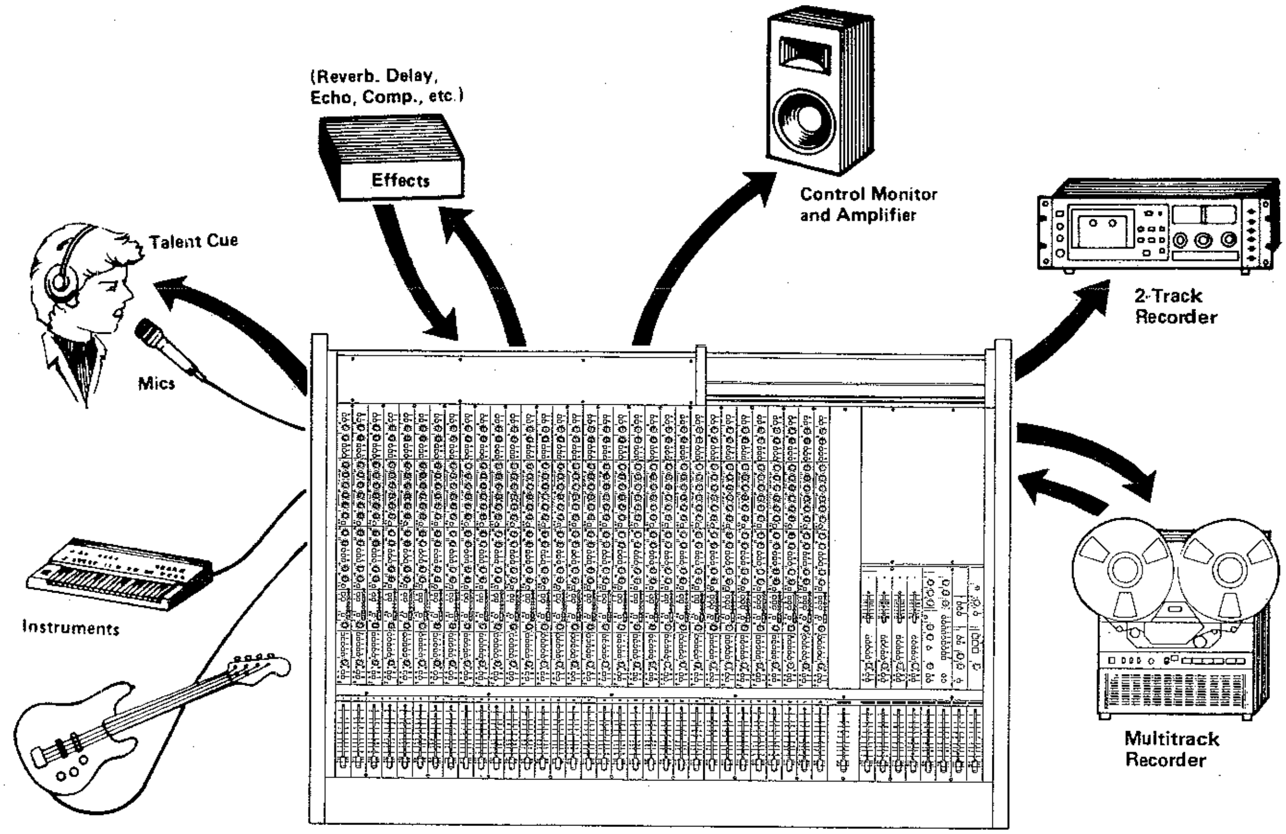
A and B: These switches can turn on the feeds to two separate control room amplifiers, typically one for large monitors (A) and another powering small near-field monitors (B). Only one can be selected at a time.

CONTROL ROOM SOURCE SWITCHES: These select the source of the contro room mix, between the **STEREO** mix (normal monitoring position), **2TR** (2-track playback), **EXT** (External stereo source 2-track playback), or **AUX 1-6** (the six auxiliary mix outputs of the M-3500, so you can monitor the headphone or effects send feeds).

The Recording System and its SUBSYSTEMS

There are six elements to a complete recording system:

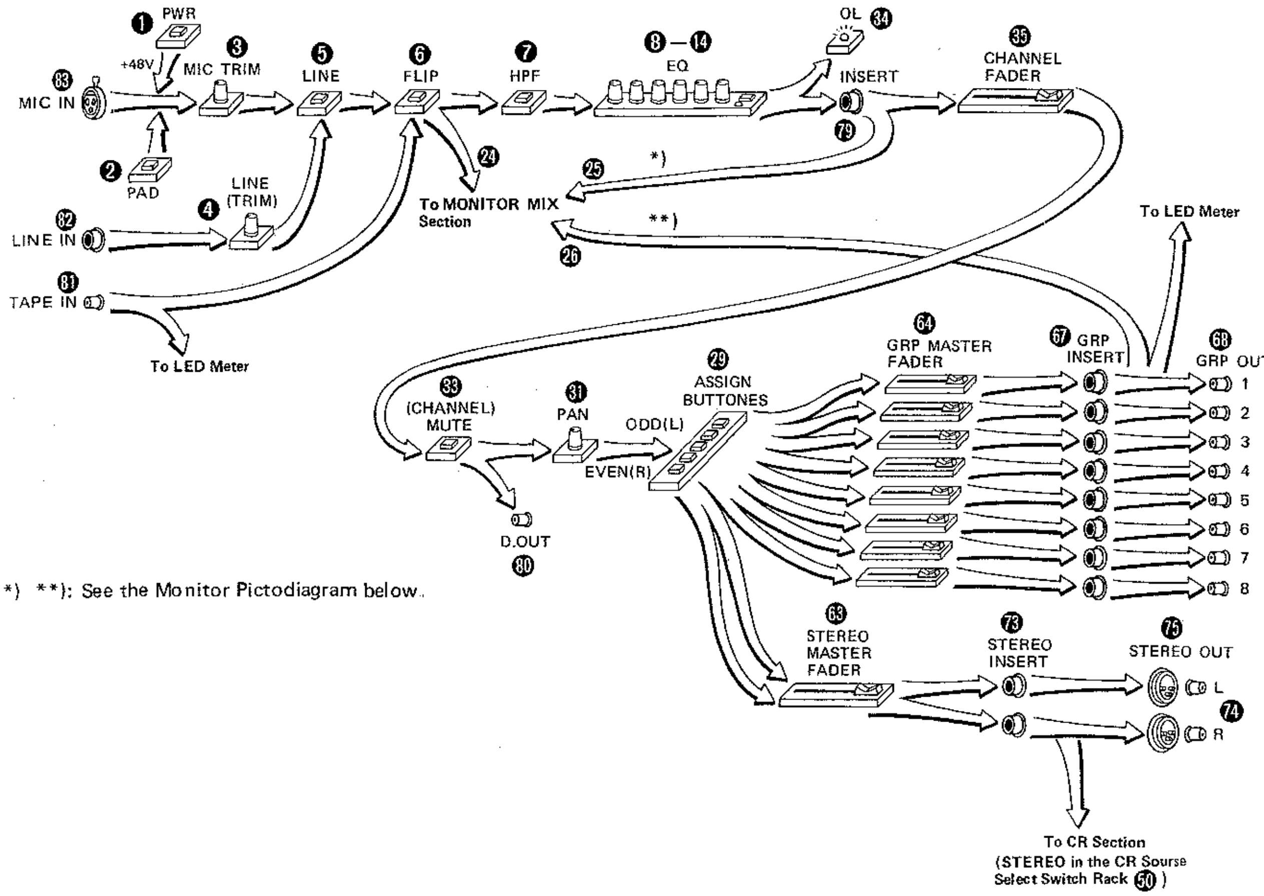
Multitrack recorder
Mixer
2 track (mixdown) recorder
Input devices (microphones, synthesizers)
Output devices (headphones or amp and speakers)
Effects processors



Note that the M-3500 is acting as the traffic control center for all the other system elements. It takes multiple inputs, processes them for level and tone, and sends or assigns them to multiple outputs. The mixer is easier to understand if you look at its relationship with the other elements one at a time.

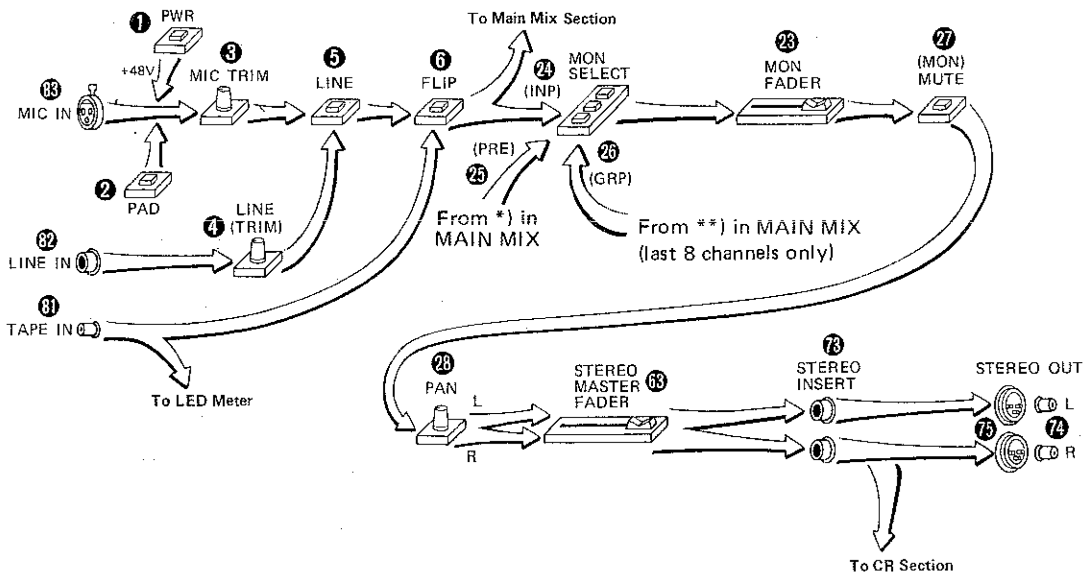
The M-3500 is actually a collection of different sub-systems. Each subsystem is a "mixer" in its own right, with its own task to perform. Each subsystem has its own inputs, processing, and outputs.

MAIN MIX: The primary system is the main mix. It receives signals from the input jacks, routes them through equalizers and faders, and sends them to any of the eight group outputs selected. Each group has its own master level control and is usually connected to a track of a multitrack recorder. The Main Mix can also send signals to the L-R (left-right) Stereo mix, for final mixdown or monitoring in the control room.



*) **): See the Monitor Pictodiagram below.

MONITOR MIX: Just above the main (large) fader, pan, and assignment switches is the Monitor section, with its own (short) fader, mute, and pan controls. Usually, the monitor is "listening to" the outputs of the multitrack recorder (or tape returns), but it can also receive signals from the groups or from individual channels. The monitor is necessary for multitrack recording. It allows you to hear various signals (live, prerecorded or in combination) during the recording process so you can make critical artistic decisions The



monitor section always feeds the L-R Stereo mix, and doesn't affect the levels being sent to the group outputs by the Main mix.

At mixdown, however, the monitor section can perform an additional function. The FLIP switch allows you to route supplementary inputs (such as effects returns or MIDI controlled instruments) into the stereo mix using the monitor section, doubling the number of sources if you wish. While this monitor path does not have EQ it does have level, pan, mute, and effects send capability.

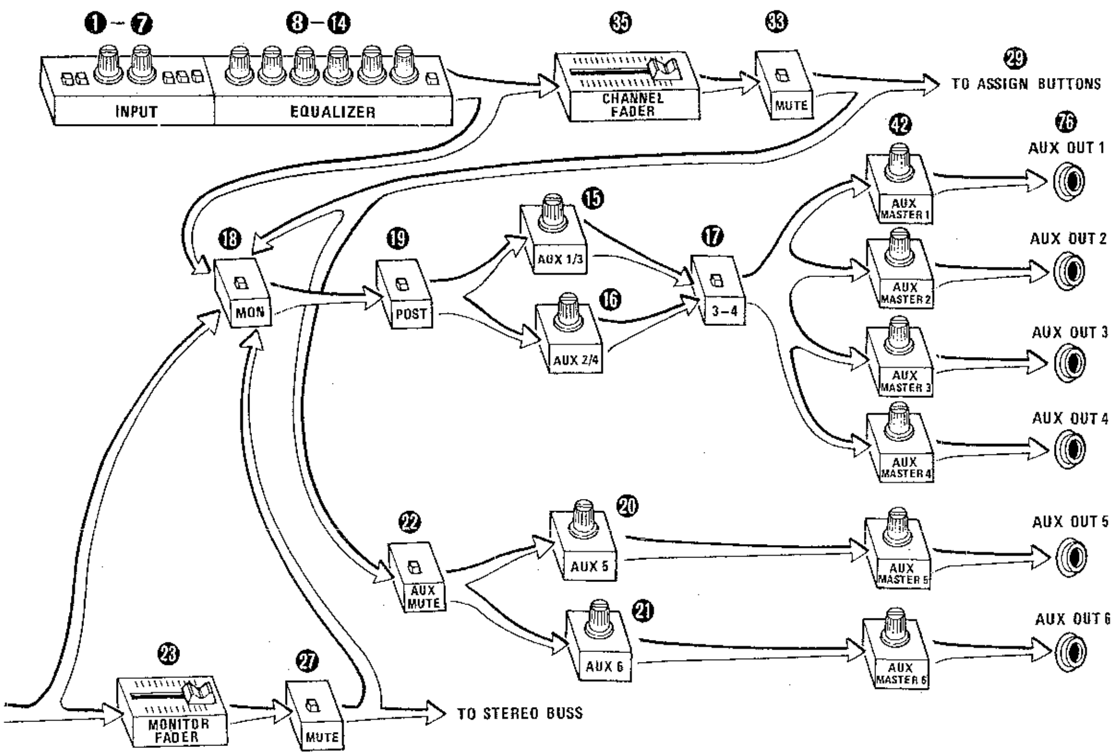
AUXILIARY MIXES: An "auxiliary" is a little less independent because it gets its signals from antoher system (in the M-3500, either the Main or Monitor path). It is even less independent if its source comes from POST fader—when the fader goes down, the AUX send goes down too. In the M-3500, the aux systems are usually intended to feed effects sends or headphone (also called "cue" or "foldback") mixes. What they'll be used for in your studio is entirely up to you. Think of auxiliaries as additional mixers, performing mixing duties separetely from the main mix.

The upper two knobs of the AUX system are the most versatile. You can control both where the signal comes from (pre or post, main or monitor) and where it goes to (Aux 1 and 2, or 3 and 4). When all the switches are UP, the two controls are getting their signal from the main signal path, before (PRE) the fader but after the EQ. This allows it to be used as an independent headphone feed whose source is the channel (similar to "monitor" or "foldback" controls on a PA board). When the POST switch is pressed, the top two aux pots get signal from after the main fader, making them usable as effects sends from the channel. So "POST" is an input select switch ("where from") for the top two auxiliaries.

The MON switch is also an input select for the first two auxiliary knobs. When it is pressed, the signal will come from the Monitor signal path. When MON and POST are both pressed, the two controls above can act as effects sends from the monitor section. If POST is up, they can act as independent cue feeds from the monitor

The 3-4 switch is an output assign switch. When pressed, it sends the output of the two controls above it to AUX busses 3 and 4, instead of their usual 1 and 2. This is used when you have special effects that aren't needed across the whole console. For example, some drum channels can access a gated reverb attached to Aux 1. The vocal channels can still be processed through a plate reverb connected to Aux 3.

Auxiliaries 5 and 6 are comparatively simple. They are always POST MAIN and will usually be used for effects sends The MUTE switch beneath them applies to Aux 5 and 6 only.



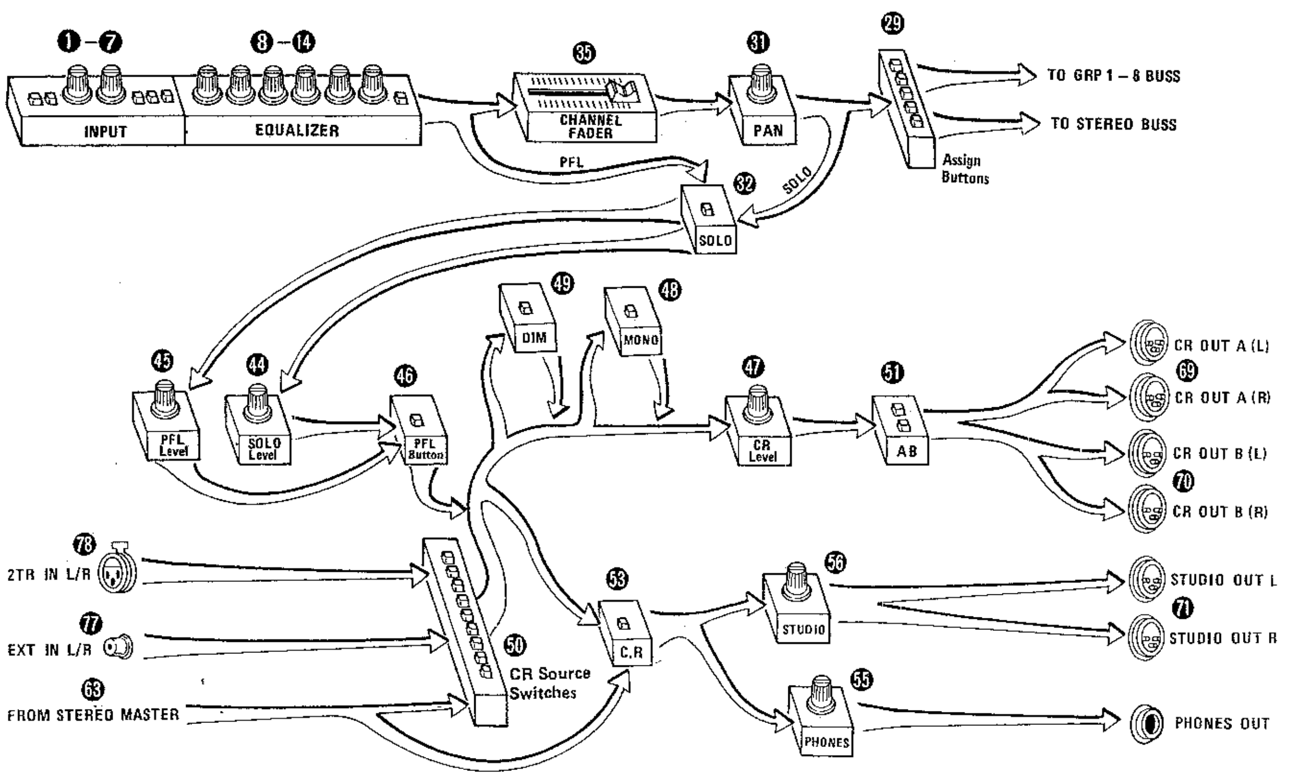
CONTROL ROOM SUBSYSTEM: With all these subsystems going to different places all at once, how can the engineer focus on one at a time? This is the task of the Control Room and SOLO section. The Control Room stereo outputs are intended to be connected to an amplifier and monitor speakers in the control room facing the console. Nine input select switches allow you to hear any of the AUX outputs, external stereo recorders, or the Stereo output of the M-3500 (which is the usual position, so you can hear the short-fader monitor section in the control room).

Regardless of which source is selected by the Control Room switch, pressing any channel SOLO switch automatically cuts off that source and sends the Solo (or PFL) mix to the control room monitors. The Solo system is used to adjust trim, level and tone in an individual channel or group of channels. It allows you to immediately hear only a few signals, and just as quickly return to hearing the entire mix.

There are two types of solo: Stereo Solo in place, and PFL (Pre-Fade-Listen). You select the type you want on the control room PFL switch. Solo takes its signal from after the channel pan pot so you can hear and adjust the level and position of signals in the stereo mix. PFL gets its signal from pre-fader, so you can hear a signal before bringing it up into the mix. SOLO and PFL each have their own master controls.

There are two sets of CR outputs, A and B. This allows you to have two sets of amplifiers and speakers (typically large and small monitors). You can DIM the control room monitors to speak to someone for a moment, or put them in MONO to check for compatibility.

There is also a Studio output and control, intended for a set of speakers in the studio for playback. Usually, this receives the STEREO output, unless the CR switch is pressed so the studio will follow the control room selection (including SOLO/PFL). The headphone jack of the M-3500 itself follows the Studio mix (except for the talkback).



Your M-3500 combines all these mixer systems—main, monitor, auxiliary and control room—into a complete audio production system. On the next page is the M-3500 Brief Guide, a quick index of the top panel controls of the M-3500. It can be removed from the binding if you wish. More detailed explanations of each control and jack are in the "Features and Controls" section on pages 24-36.

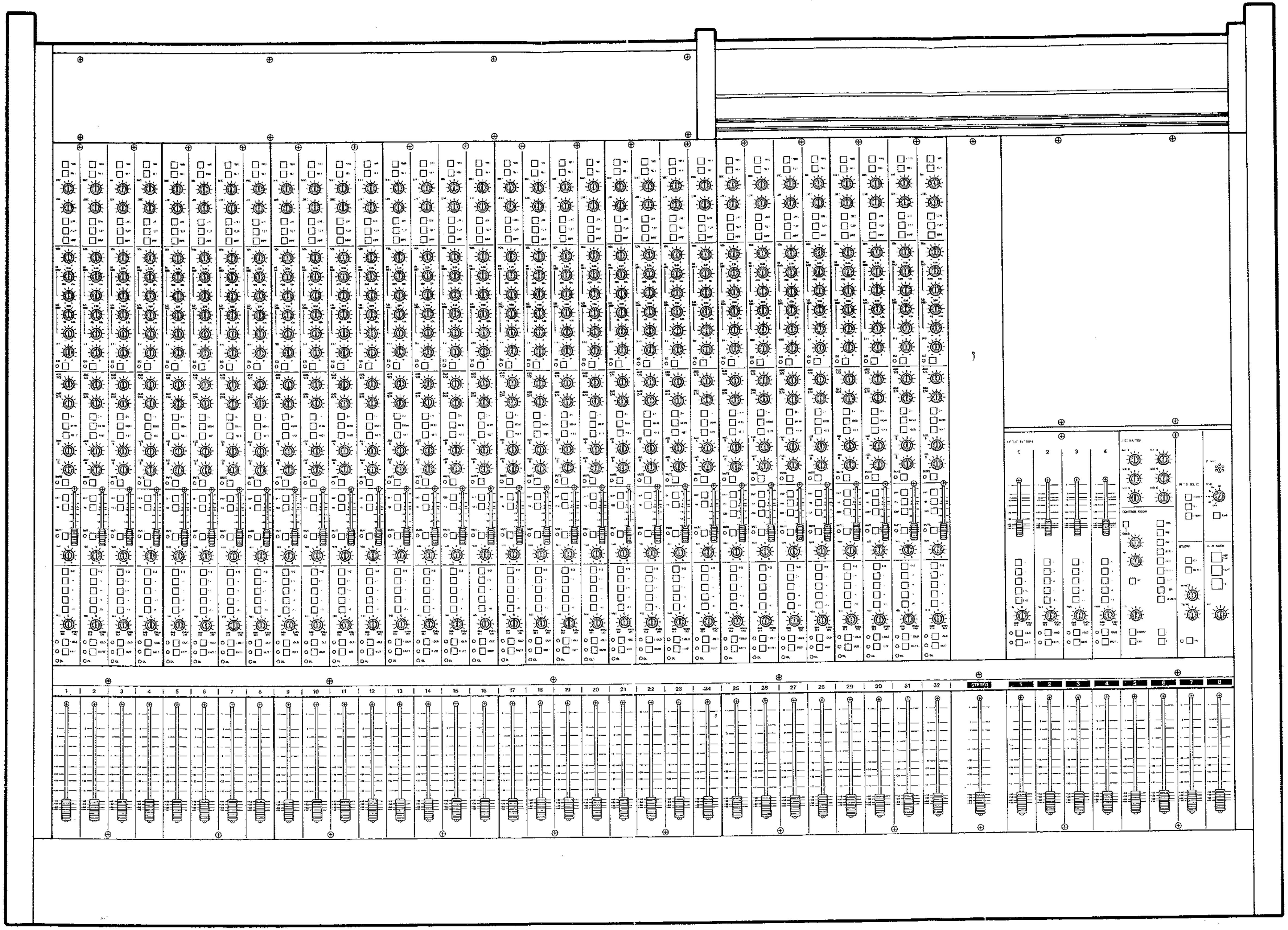
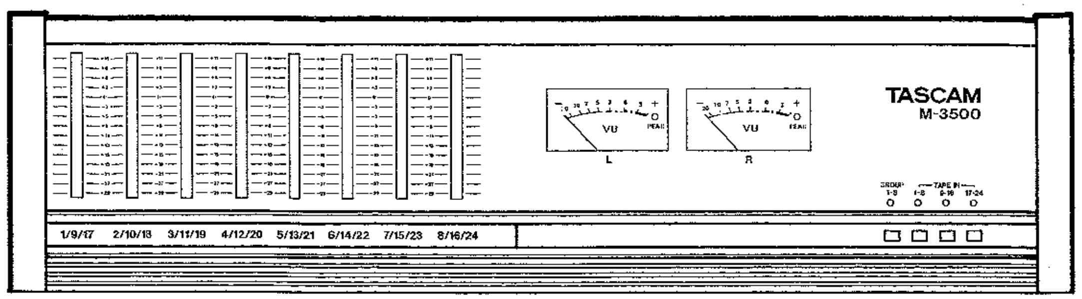
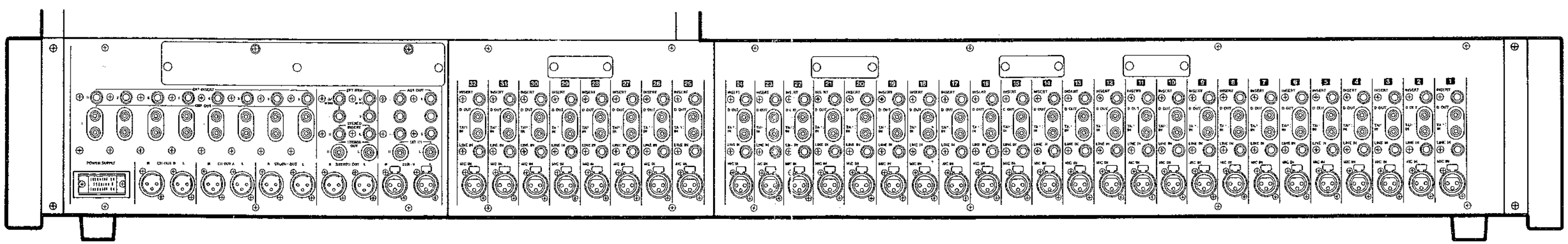


Illustration shows the 32-input version.


TASCAM
M-3500
VU
VU
L
R
1/9/17
2/10/18
3/11/19
4/12/20
5/13/21
6/14/22
7/15/23
8/16/24



This section of the manual is designed to help you connect your M-3500. The "Operations" section is based on these connection diagrams However, these methods are not the only way to use the M-3500. As you learn the features of the M-3500, you will discover alternative ways of connection and operation which may better suit your needs

When installing your system, make sure all devices are turned off and the level controls are turned down.

Power:

1. Place the PS-3500 power supply unit within easy reach of the cable to the console. Do not place it directly under or over signal processing gear in a rack, as it may induce electromagnetic hum in those units if they are not properly shielded.

2. Connect the multipin cable from the power supply to the M-3500.

3. Connect the AC plug of the PS-3500 to an outlet. Try NOT to use outlets that are on the same circuit as air conditioners and old refrigerators. These things may introduce noise to the system. All elements of your system should connect to the same circuit if possible, but not to the same plug. There are several multiplug power conditioners on the market that do a good job of filtering out spikes and noise that come through the AC line. Make sure that the outlet you connect to is properly wired; AC outlet testers can be inexpensively purchased at electronics stores. If other units in your system have three prong AC cables, it is possible they may introduce "ground loops" or hum in the system if the AC ground is not secure. If this happens, you may need to ground elements separately and "lift" the ground with a three prong to two prong adapter. Make sure you follow the manufacturer's recommendations before you do this, though.

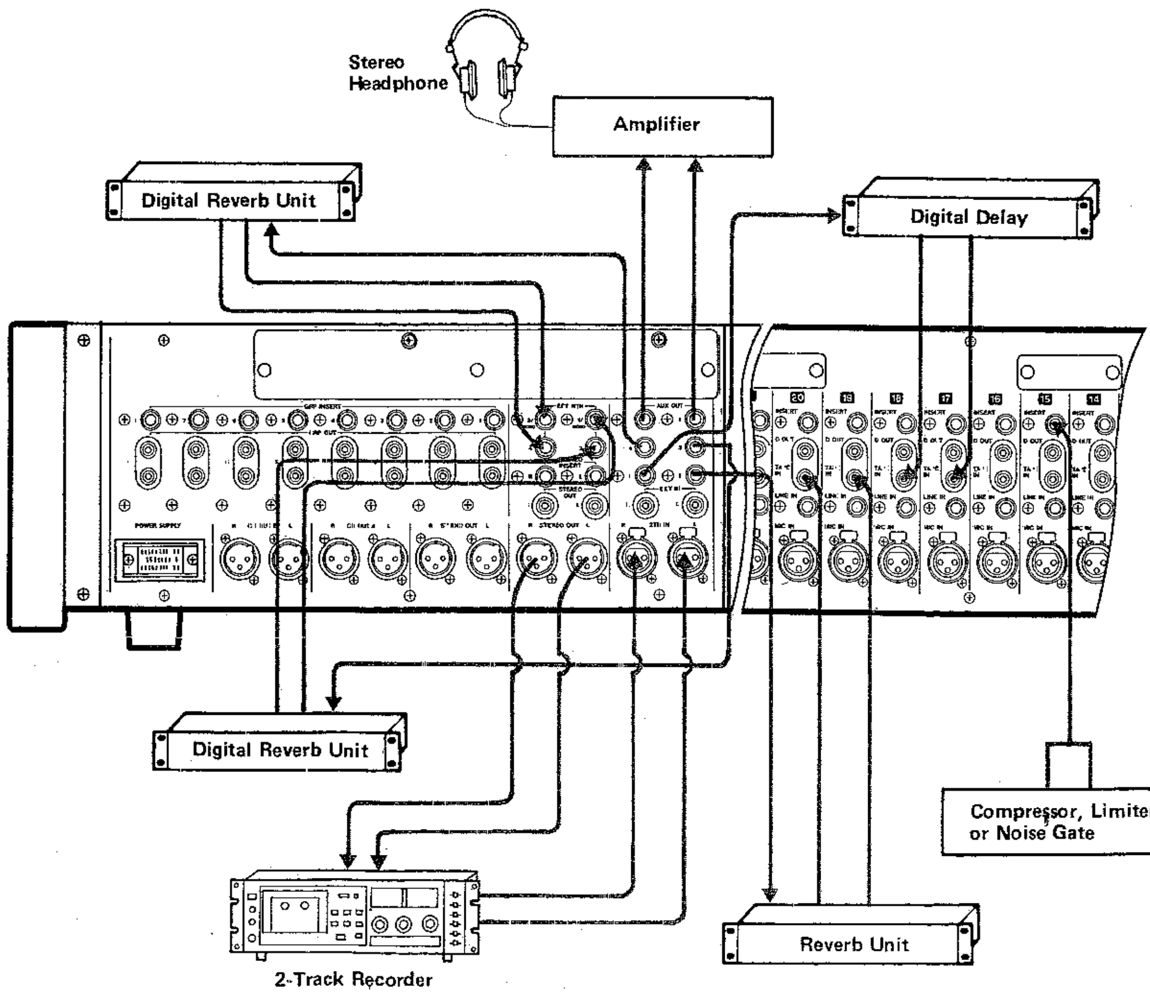
Proper audio grounding and shielding is a study in itself. Your number one priority is the safety of the system, and second is getting low noise. Standard industry practice has become the "star ground", where all system elements are grounded at one place only: the console or patch bay. If you have problems with noise, or any "shocks", consult a professional in studio electrical system design.

INPUTS:

In this example, we are going to assume you aren't using a patch bay. However, as your system grows, you may find it will make your life easier to acquire some TASCAM PB series patch bays.

MICROPHONES, low impedance (200 to 600 ohm) can be connected to the MIC jacks. If they are phantom-powered type, press the PWR switch for that channel; otherwise leave it off. Line level XLR outputs may also be connected to the MIC jack, if you press the PAD swtich, lower the MIC TRIM level, and make sure PWR is off.

LINE INPUTS, such as synthesizers, guitars, and drum machines, may be connected to the 1/4" LINE IN jacks. Some guitars may sound better if they're run through a preamplifier or direct box first.



TAPE RETURNS, the outputs of your multitrack tape recorder, should be connected to the TAPE IN jacks. They are designed for −10 dBV unbalanced tape returns. Use good quality shielded, low capacitance, high RF resistant cables such as the TASCAM Professional series. Cable lengths up to 20' are acceptable, but the shorter the better.

If you're not using all the tape returns, you can use your "spares" for other high-level line inputs if you wish —if you look at the block diagram you'll see the only difference between TAPE and LINE ins is that Line has an input trim and amplifier.

If your tape recorder has only XLR-type balanced (or unbalanced) +4 dBm connectors, an optional balansing amplifier LA-3500 is available.

OUTPUTS to Multitrack:

1. Using good quality cables, connect the GRP OUT jacks to the same numbered track of the multitrack recorder. Color-coded plugs help. Note that if you're connecting to a 16-track recorder, Group 1 will feed tracks 1 and 9, Group 2 will feed tracks 2 and 10 etc.

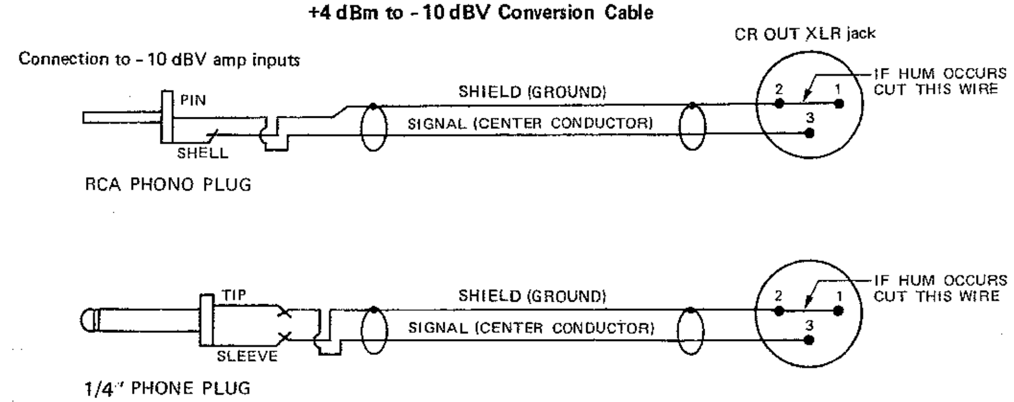
2 In some cases, your 16 or 24 track recorder will "normal" the connections, so you only need to run eight cables to your recorder. For example, the TASCAM MSR-24 has an "Input Link" feature that patches the track one input to tracks 9 and 17 if desired.

3. If you need to record more than eight tracks at once, you will use the D OUT (Direct Out) jacks on certain channels. See p. 16, Right, "Recording on more than 8 tracks".

Control Room and Studio Amplifiers:

1. Connect the CR OUT A XLR jacks to the input of your main control room amplifier. If your amplifier does not have XLR inputs, obtain an adapter cable or make one according to this diagram:

+4 dBm to -10 dBV Conversion Cable



2. If you plan to have a second, smaller set of control room speakers with a smaller amp, connect the CR OUT B jacks to it, as above.

3. If you plan to have speakers in the studio, connect the STUDIO OUT L/R jacks to its separate amplifier.

Mixdown Deck:

1. Connect the STEREO OUT L/R jacks to the input of the stereo mixdown deck. If it is a +4 dBm XLR machine, use those connectors; if it is a −10 dBV unbalanced machine, use the phono connectors above. Both connectors may be used at once, feeding multiple mixdown decks (for example, a DAT and a standard stereo cassette).

2. Connect the outputs of the mixdown deck to the 2TR IN XLR connectors, or to the EXT IN unbalanced connectors above them.

Effects Devices and Cue Amplifiers:

1. You will have to decide how you want to use your auxiliaries. In this example we will connect AUX SEND 1 and 2 to a stereo headphone amplifier, AUX 3 and 4 to two digital reverbs, AUX 5 to a digital delay, and AUX 6 to another reverb unit. Note that in this case, the digital reverbs are not "true" stereo even though they have a left and right input, so we only connect to their MONO input.

2. Connect the outputs of the effects devices to the effects returns. In this case, we wanted to use the synthesized stereo outputs of the digital reverbs, so the AUX 3 and 4 units connect to effects returns 1, 2, 3, and 4. The outputs of the other effects units are connected to the left over tape returns on channels 17, 18, 19, and 20.

3. If you have a compressor, limiter, or noise gate, it normally is connected to an INSERT point, using a PW-2Y or PW-4Y insert cable. Don't connect these yet, until you're up and running, because it's one more set of controls in the signal path that might cause confusion.

See p. 21, "Using Effects" for more detail on effects processing.

Preset the Controls: To avoid problems, begin with the power off. In each channel, "zero out" the controls as follows:

- Bring all the faders down.
- Set all MIC and LINE trims, AUX sends, AUX MASTERs, CR, SOLO, PFL and STUDIO levels to full counter-clockwise (7 o'clock).
- Set all EQs and PAN controls to center (12 o'clock). You may feel a "detent" at their center position, except for the EQ center frequency controls.
- Make sure all the switches (such as CHANNEL ASSIGN, EQ and MUTES, FLIP etc.) are up (off), except for any PWR switches you need for phantom power microphones.
- Depending on what inputs you have connected, set the channel input select switches (LINE) to the appropriate position: MIC (up) or LINE (down).

Testing Connections To and From the Multitrack

1. Once all the connections are made, begin energizing the system by first turning on any electronic instruments, then the mixer, then effects devices, and finally the power amplifiers. **NOTE:** When shutting a sound system down, turn the power amps off first. Wait at least 30 seconds for capacitors to discharge, then continue to turn off the remaining equipment.

2. On the far right side of the console, **set the oscillator frequency to 1 kHz. Press TONE** to the down position.

3. Make sure the switch on the master meter bridge is set to read "Group 1-8".

4. **Slowly raise the Group 1 master fader** to "0" (about 2/3 up). You should see the Group 1 LED meter on the M-3500's meter bridge go up.

5. **Put your multitrack recorder into INPUT or REC READY mode for track 1.** Its meter should rise to match the meter on the console. (If the recorder has input level controls of its own, raise them until the meter reads 0 VU.)

6. **Press INP** next to channel 1's monitor fader. **Raise the short monitor fader** to its nominal ("0 dB") position

7 **Slowly raise the STEREO master fader.** You should see the two VU meters start to move If not, make sure the METER SELECT (the 3 switches on the console, not the 4 on the bridge above) is in STEREO.

8. In the control room section, **select STEREO and press the A button.** Make sure that no SOLO lights are on anywhere on the console, and that DIM and MONO are off. **Slowly increase the CR LEVEL control. If you haven't already, raise the level controls on the monitor power amplifier.** At this point, you should be hearing a pleasent (?) 1 kHz tone in your monitors. Congratulations. If not, some connection is crossed or loose. Retrace your steps and try again. If you're really stuck, put a prerecorded tape on the multitrack and put it into play, to check that the tape returns are working. Or start at the amplifier inputs and work your way back (did you remember to wire the speakers?). Be logical and go one step at a time—eventually you'll find the knob that's set wrong, or the cable that's bad (even new ones are, from time to time).

Once you've gone through this procedure on track/group 1, do the same for all the other tracks and groups in turn.

Meter Calibration: It is normal for slight variations both in the meters and in the fader readings. If a fader must be set a little above "0" to get 0 VU, it's nothing to be worried about. The meter settings of the M-3500 are very stable, so recalibrating the meters of the mixer itself is not advised unless you have an extremely accurate voltmeter. If levels of your multitrack don't match those on the mixer, the multitrack may need calibration. If the tape meters don't play back at the same level you recorded, a complete alignment of the deck may be needed to bring everything back into "spec". Level differences of 1 dB or less may be due to other factors, and should not be of concern.

Set the 2 Track Level:
You can also set the input sensitivity of your 2 track mastering recorder. Adjust the STEREO Master fader until the M-3500's VU meters read 0 VU. Put your mastering recorder into INPUT or RECORD mode, and adjust its input control until it reads 0 VU on that unit (some digital recorders may use "–18" as their "0 VU" point). When you put a prerecorded tape on your two track, you should be able to hear it in your monitors when you press the 2TR or EXT switch in the control room.

Set the Control Room Amp Level
If the CR level control is set relatively low (below 12 o'clock), and the monitors are very loud, it's a good idea to turn down the controls on the power amplifier itself. This is a safety precaution to avoid damage to your monitor system, and it also leads to quieter operation.

Set the Input Levels
At last you're ready for audio. Plug a microphone into a channel and make sure the LINE switch is UP. Turn the MIC TRIM up to the 12:00 position. Press that channel's L-R ASSIGN switch. Slowly increase the channel fader to the 0 dB position while someone speaks into the mic. You should hear it faintly, if the STEREO fader is still at its nominal position and the CR is set to hear STEREO. Slowly increase the MIC TRIM until the stereo meters read 0 VU. (Watch out for feedback if the mic is in the control room.)

Setting LINE TRIM is basically the same procedure. Remember, these procedures are only initial. In actual performance, further adjustments are usually needed to fine tune the settings. For more information on setting gain, see p. 19, "Gain Staging and Mixing Advice."

If the channel signal level is too high, the channel OL (overload) LED will light If this indicator comes on, turn the MIC or LINE TRIM control down.

Setting Auxiliary and Effects Send Levels
Getting the right levels to your effects devices is as important as setting levels to the recorder. Once you have a line or mic signal working at the proper gain, the "2 o'clock" position of the AUX control and AUX master should make the meter of the effect device go to its "0 VU" indication, whatever it may be. Consult the manual for your effects device, and see p. 21, "Using Effects" for more information. At this point, just verify that you're getting signal to the device, and that it is returning to the console (either through the effects returns or the monitor spare inputs).

Multitrack recording is divided into three separate mixing tasks:

1. We must route the input signals to the desired tracks of the recorder at the proper level to achieve the best signal-to-noise ratio.
2. We need one or more cue mixes for the artists.
3. We must create a monitor mix in the control room for the engineer and/or producer.

To be most effective, these three mixes must be independent of each other.

Recorder Mix

Routing the inputs to the multitrack is a simple task. Once you have set the input trims properly, press the channel assign button to send the signal to the correct output group. Remember that the MAIN PAN control must be set correctly—signal won't make it to group 1 if the PAN is turned hard right.

Remember that although the M-3500 is an eight group design, the 16 outputs and in-line monitoring make it possible to perform 16 or 24 track recording and mixdown. Each group supplies signal to a pair of group output jacks. Each pair is controlled by a single group master fader, and the signal is identical at either jack. Typically, group one feeds track one, track nine, and track seventeen. There's no need for an on/off switch or additional assignment—those selections are made at the REC FUNCTION track switches on the recorder, and the high input impedance of modern recorders means one group output can feed more than one track anyway. Since the whole idea of multitrack recording is to build up tracks a few at a time anyway, it usually isn't necessary to have as many output groups as tracks.

As far as levels are concerned, if your equipment is properly calibrated the meters on the multitrack recorder will match the meter levels on the M-3500 itself. "0 VU" is the typical ideal level for recording, though peaks to +8 and above can be tolerated on some types of program material. It would be presumptuous to suggest a "best" level setting. The actual final setting for a performance can only be judged and set by the operator, and we encourage you to try different methods and submit them to the ultimate judge—your ears. In most situations, the magnetic tape itself is the limiting factor. The M-3500 outputs themselves do not distort until approximately +28 dB—long after the meters have pegged. The meters do not reflect mixer distortion, but distortion of a typical tape recorder. (Note, however, that it is possible to clip the input channel electronics by an improper TRIM setting, even though the following signal path is well within its limits. This is indicated by the channel OL lights).

If you want to see the level of the tape returns at the console, press the "TAPE IN" switch on the meter bridge. This will bring the tape inputs up on the meter, eight at a time, instead of the group outputs.

For tape ins 25-32 on the 32-input version to be monitored by the meters, an optional expansion meter unit needs to be mounted. See the ACCESSORIES section of this manual.

For important information about level setting, see p. 17, "Gain Staging".

Recording more than 8 tracks simultaneously: If you need to record more than 8 tracks at once, you can use the D OUT (Direct Out) jacks of certain channels to feed the multitrack recorder, instead of the group outputs. In this case, there is only one input per track, and the record level is controlled by the channel fader only, and the Group Meters will not show the record level. Press the TAPE IN switch on the meter bridge, or go by the meters on the recorder itself.

In most cases, you patch the direct outs of single channels (such as a lead vocal) to a track, saving group outputs for instruments (such as multiple keyboards) that have to be combined.

Cue Setting

In our example, AUX 1 and 2 are being used for headphone cue feeds to the studio. Your basic decision to make is what the source of the signals should be: the multitrack tape recorder outputs, or a "tap" from some section of the mixer itself.

Tape cue: For the performers to hear the output of a tape track, press INP and MON in that channel. (Make sure that FLIP is UP). If there is signal on the tape and it is in PLAY-REPRO mode, they should hear it. There are several advantages to using tape as the cue source:

1. The cue level will be the same during recording and playback.
2. If the tape recorder has an INSERT (pre-roll sync) feature, it will be easy for the performer to hear when punch-ins and outs are made.

The disadvantages of using tape for cue, when you would want to monitor the mixer channel instead are:

1. Any changes the engineer makes to the main faders (in order to lower the record level, for example) will also be heard in the cue mix.
2. The performer may not be able to hear at certain times depending on the settings of the tape deck itself. For example, taking the track out of REC READY mode can cut the feed of the instrument to the cue
3. When instruments that aren't going to be recorded to multitrack tape—for example, MIDI "virtual" tracks—must be heard, source cue is the only way to go.

Source cue: If you want to use the mixer channel as source for AUX 1-2, simply make sure that the MON switch is UP.

Tape/source mixed: Some artists need to hear both the pre-recorded track and their "live" instrument simultaneously when they are doing a punch-in to fix a section.

If you want a mix of the tape track and its source, make sure the source is in a different channel from the track. For example, while doing guitar overdubs onto track 7, the guitar can be patched into channel 24. To get a mix of both, press MON in channel 7 (Aux 1-2 will hear tape track 7) and make sure MON is up in channel 24 (Aux 1-2 will be a cue send of the guitar)

Group output cue: Another variation of "source cue" is to feed one of the eight group mixes to the headphones. This can be done on the highest 8 input channels on the console, the ones with GRP switches in the monitor, directly under the 8 Group meters. Press GRP, and MON in that channel. This will bring the mix of everything assigned to that group into the headphone mix. The advantages are when you have already done a complex mix (of percussion mics, for example) for recording and don't want to repeat the mix all over again on each individual AUX control

Effects to Cue: Will returning effects make the cue mix more cluttered, or is it needed for artists to perform properly? If you decide you need effects in the cue mix, there are two ways to go:

1. Patch the output of the effects devices into a line input, instead of the Effects Return inputs of the console. As a line input, it can be monitored like any other source. OR:
2. Leave the effects in the Effects Returns, but assign them to a group that is not going to be recorded. Then bring that group output into the cue mix (see "Group Output Cue", above).

Of course, if you are recording "wet" (effects to tape), using tape as the cue source brings the effects returns into the cue mix as well.

Speaking to the cue section: Press AUX 1-4 in the Talkback section, and the console's built-in mic will be patched to the performers' headphones as your control room monitors are muted to avoid feedback. Press also the TB switch if you have a studio speaker system, and your voice will go out over the speakers (regardless of whether ON is pressed).

Other decisions you may have to make include: Should the AUX 1 and 2 mixes be stereo (1 feeds left, 2 feeds right) or two separate mono mixes (one for the drummer, one for the vocalist)? Should I make EQ settings while I'm recording? (AUX 1-2 is always post-EQ, even when it is pre-fader—the musicians will be able to hear EQ changes in the headphones).

Monitor Mix

To create your control room monitor mix, you will usually rely on the STEREO switch in the Control Room section, and use the short monitor faders. A lot of the same options available to the cue mix are available in the control room as well. There is one important note, however. In most cases, the engineer should be hearing tape, not source. This allows you to hear the signal as it's actually being recorded, any verify the signal continuity to and from the mixing console and the recorder

To monitor tape: Press INP, and make sure FLIP is Up.
To monitor mixer channel: Press PRE
To monitor group outputs: On the highest 8 channels of the console, press GRP

Note that the "L-R" assign switch on the main channel will also bring the output of the channel into the control room monitors. In some cases this may be desirable; in other cases it can mislead you. Make sure that no "L-R" assign switches are down if you want to make sure you're hearing what's going on to the tape and only what's going on to the tape.

When the recording is complete, rewind the tape and play it back through the control room speakers. You don't have to change any control settings to do this, and should hear exactly the same mix as while you were recording. If playback is also desired on the STUDIO speaker system, press the studio ON switch and raise the STUDIO level. The studio will hear the same stereo mix you are hearing. Remember to turn the switch OFF before you cut more tracks to prevent feedback.

In some situations, you may find SOLO useful. If you want to solo a tape return, simply press FLIP and SOLO. Some engineers make a practice of playing back the newly recorded material after every pass. This will allow you to hear if any audio or acoustical problems are emerging, such as improper tuning, voicing timing errors, etc. If problems are found, correct the cause and re-record the track. While certain tonal characteristics can be equalized and adjusted later, many problems are impossible to "fix in the mix".

Monitoring on the large faders: Depending on how many inputs you are recording, and how many tracks you have, you may want to monitor via the main mix. This "trick" is to leave your lower channels open, using the higher channels of the console for overdubs. Since the lower channels aren't needed during the overdubs, you start your mixdown "early" by pressing FLIP on those channels, and assigning their outputs to L-R only. You can start marking levels, using EQ, even adding effects this way.

Operation: Mixdown

Once the multitrack master tape has been completed, the next step is to mix it down to a standard two track stereo format. The procedure is a reverse of the direction signals took during tracking: instead of going TO the multitrack, signals come FROM the multitrack. Typically, you want to bring signals through the main mixer channels so they can be EQ'd and processed. This is the job of the FLIP switch—press it to bring tape to the channel, and mic or line "flips" up to the short monitor fader.

To mixdown to stereo:

1. Press FLIP on all the tape channels.
2. Assign all the channels to L-R.
3. Make sure the monitors are muted.
4. Press 2TR (EXT) in the Control Room section.

Note that the monitor faders are still feeding the stereo mix. Unless you are using them for MIDI instrument inputs or effects returns (see Double Inputs, below), make sure they are muted or they will add noise to the mix. In fact, any unused faders should be muted.

During mixdown, press 2TR (or EXT) in the Control Room source. This allows you to verify that signal is indeed reaching the recorder. You may have to place the deck in REC READY, REC PAUSE, or INPUT to hear it. If you have the inputs of the two-track calibrated properly, the meters of the deck and the two VU meters of the M-3500 will track properly. The red PEAK LEDs of the M-3500's meters flash at approximately +10 VU, so you can see transients too fast for the meter to show. If you press CR in the METER SELECT, and 2TR is the control room source, the two VU meters will reflect the output level of your two track.

Note that if you SOLO or PFL channels during a mixdown, these solos will NOT affect the stereo output to the two track. PFL is useful during mixdown to get a cue from a track before you bring it in. For example, PFL a muted vocal track, and unmute it after the singer clears her throat before an entrance. SOLO is useful for checking the mix, for example, soloing two guitars to see if they balance each other in the stereo image

If you want compression or equalization of the entire stereo mix, there are two INSERT points on the stereo L-R output that can be used to insert signal processors in the signal path. These insert points are post-fader.

Mixdown with double inputs:

If you have a large number of MIDI virtual tracks and effects returns, or other sources in addition to tape tracks at mixdown, you can use the monitor short faders as additional inputs. It helps to think of them as "short faders" instead of as monitor faders, because they are going the same place the large faders are going in this case: the stereo outs.

FLIP brings the tape return to the large fader, and either the MIC or LINE input (as chosen by the LINE switch at the top of the channel) to the short fader. To use AUX 1-4 as effects returns, disconnect the headphone amp from AUX 1-2 and repatch those outputs to an effects device instead. Press MON and POST, and Aux 1-4 will act as an effects send from the short fader. Aux 5-6 are still effects sends off the large fader.

Subgrouping

Typically, the groups have no use in a mixdown. However, you can use them to subgroup when you have a large number of input channels which must be controlled through one fader as a group at mixdown. This subgroup gets to the stereo mix via the short faders GRP source capability.

1. Assign the channels you want to a group, and NOT to L-R directly. For example, if tracks 1-10 are all drum tracks, assign channels 1-10 to Group 1-2.
2. In the channels directly under the group meters (1/9/17 and 2/10/18), press GRP as the monitor source. (These are the first two short faders that have GRP switches).
3. Turn the first monitor PAN hard left, and the second monitor PAN hard right.
4. Bring the 2 monitor faders to the "0 dB" (nominal) position.
5. Bring the Group 1 and 2 Master Faders to their nominal position.
6. Play tape, set the CHANNEL PANs 1-10 to the desired positions, and get the mix you want on the channel faders.

When you want to change the level of the entire drum group, you can move either the group master faders (which we recommend, because they're larger), or the monitor faders. To mute the entire drum group, press the two MUTEs in the monitor.

Using the Equalizer

EQ can be used to change the tonality or timbre of a signal in an individual channel of your mixer. There is no specific control setting we can advise—the subjective art of applying EQ must remain the responsibility of the person performing the mix.

There are 4 bands of equalization in the M-3500, plus a High Pass Filter used to roll off low frequencies. The EQ points were chosen for the most musical sound and are useful on a wide variety of signals.

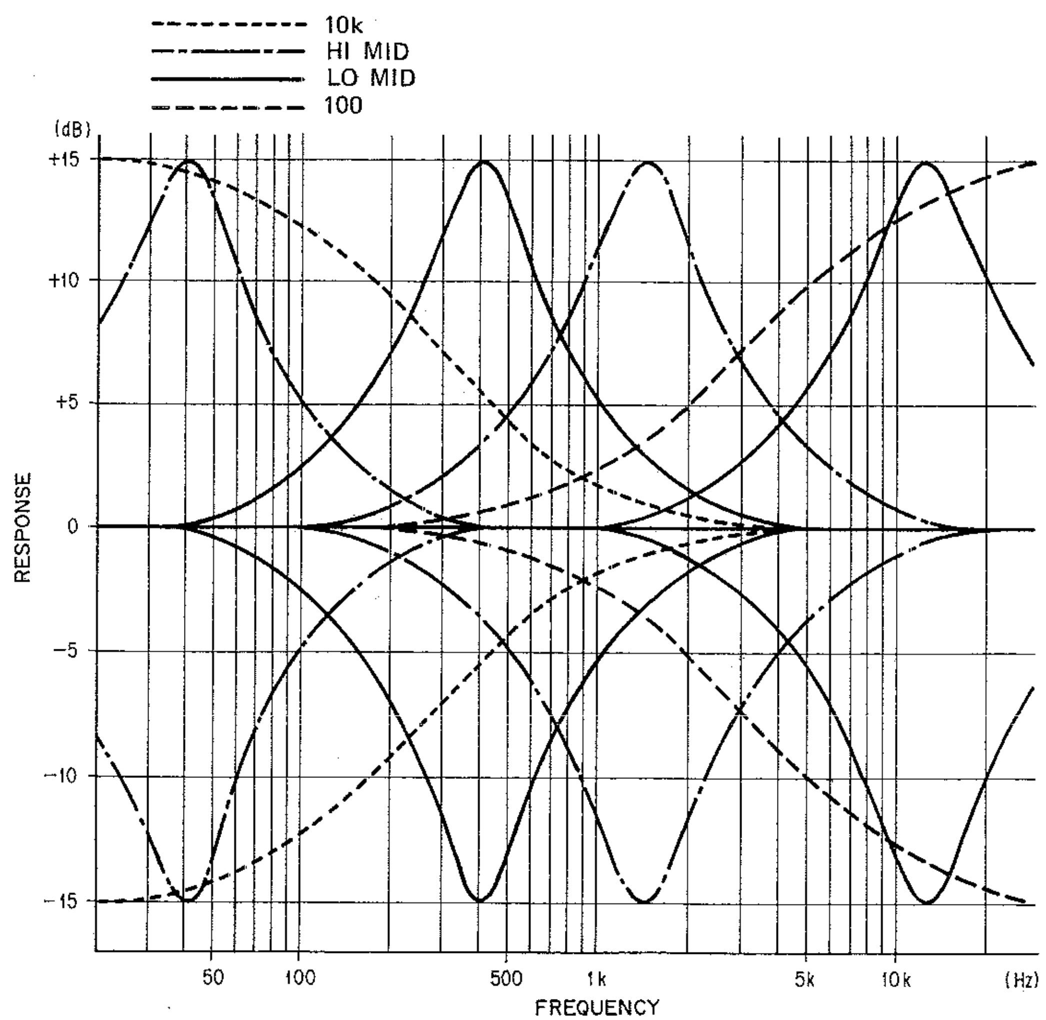
The top control (10 kHz) is a "shelving" type of control. This means there is a gradual increase or decrease from approximately 2.5 kHz to the shelving point, and frequencies above that point are all boost or cut by the same amount. This affects the relative brilliance or brightness of the signal, similar to a "treble" control.

The next band (HI MID) is a peak-and-dip type with a variable frequency, sometimes called a semi-parmetric or sweep-type. There are two controls. One determines the center frequency of the affected band while the second determines the amount of boost or cut applied to the band. It can boost or cut 15 dB. The center frequency can vary from 420 Hz to 13 kHz. The "Q" or bandwidth varys depending upon the amount of boost or cut.

The next band (LO MID) is similar to Hi Mid, except it covers the frequency band centers from 42 Hz (actually low bass) to 1.3 kHz.

The 100 Hz control is a shelving type, cutting or boosting frequencies uniformly below 100 Hz with a gradual increase/decrease up to 500 Hz depending on the setting.

Not actually part of the EQ itself is the HPF or high-pass filter. This is designed to out off low frequencies in a signal, when those frequencies represent undesirable rumble or noise. For example, vocal mics can pick up stand vibrations that are not part of the desired vocal sound, and the HPF with its 12 dB/octave filter at 80 Hz will filter them out



If EQ is desired, begin by determining which band requires alteration. Keep in mind that there are two ways to alter the tonality of a signal using EQ. One is to boost what you want to hear, the other is to cut what you don't. These two opposite approaches can both be effective depending on the situation, and sometimes can be combined. For example, if a vocal signal is a little too heavy, you could either reduce the low frequency content, or increase the amount of mid or high frequency signal.

The technique of using the sweep EQ is simple. Adjust the GAIN control of the band so there is an exaggerated amount of boost or cut, then slowly sweep the FREQUENCY control through its entire range. As the control is turned, you will hear the change in the signal's content. When the desired frequency is isolated, set the GAIN to the desired level. A few trials with different signals will give you a feel for the M-3500s capabilities.

The EQ bands perform in an interactive way. It is possible to raise the 100 Hz bass shelving control, and then cut the LOW MID at a certain point to remove an undesirable overtone. Remember, though, that you can't put back what isn't there in the first place. Boosting the treble on a synthesizer that has no signal above 7 kHz only boosts the noise of the synth. Not amount of tonal change can correct instruments which are out of tune or signals which are distorted.

Proper EQ can lower noise in a mix. Improper EQ can cause distortion. Note that the OL light is post-EQ; if a TRIM is set high, increasing EQ can make the light go on because it is adding gain to the signal. If this happens, simply lower the trim (or be more reasonable with your EQ setting).

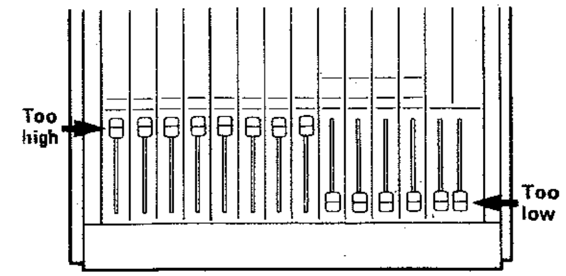
Of course, sometimes the best EQ is no EQ. If it's not needed, bypass the unused electronics by releasing the EQ ON switch.

In some cases, even more control is needed. Octave equalizers, 1/3 octave equalizers, notch filters, and full parametric EQs (such as the TASCAM PE-40) can be inserted into the signal path via the INSERT jack, when the on-board EQ of the M-3500 needs to be supplemented. Usually this is the case when a signal has particular noise bands (such, as a 60 Hz hum) that need to be filtered out

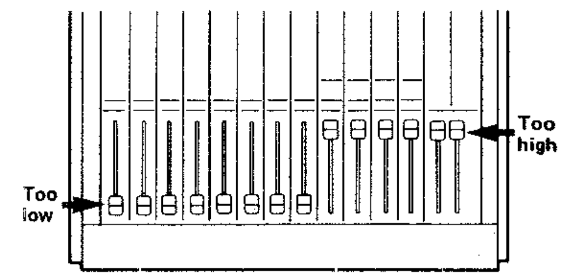
Gain Staging: A Word of Mixing Advice

How to set levels for best operation

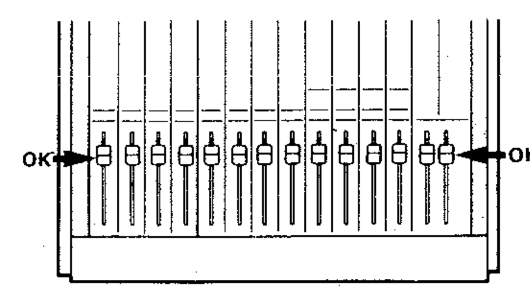
Good design, state-of-the-art components, and proper shielding makes the noise floor of each section of the TASCAM M-3500 superior to previous consoles in its class, but that does not mean there is no noise Every piece of electronic equipment has a noise floor, a varying random voltage that we hear as noise if it is amplified. Even a plain resistor has random molecular motion (except while being frozen to "absolute zero" temperature in a laboratory) that can be measured as noise. When it comes to operating a mixer, the trick is to operate the mixer with the proper gain staging so you don't needlessly amplify the noise that is there.



Similarly, every piece of electronic equipment has a limit of high voltage that it cannot pass. Even a piece of wire will melt if you push too much voltage through it—that's what fuses and circuit breakers are for. Less dramatically, electronic circuits will start to "clip" or distort when too much signal is passed through them. An amplifier with a 30 volt power supply cannot pass a signal of 32 volts, so the tops of the waveforms are "clipped off", resulting in the familiar square wave. This distortion is just as annoying (and in terms of equipment, more dangerous) as noise is.

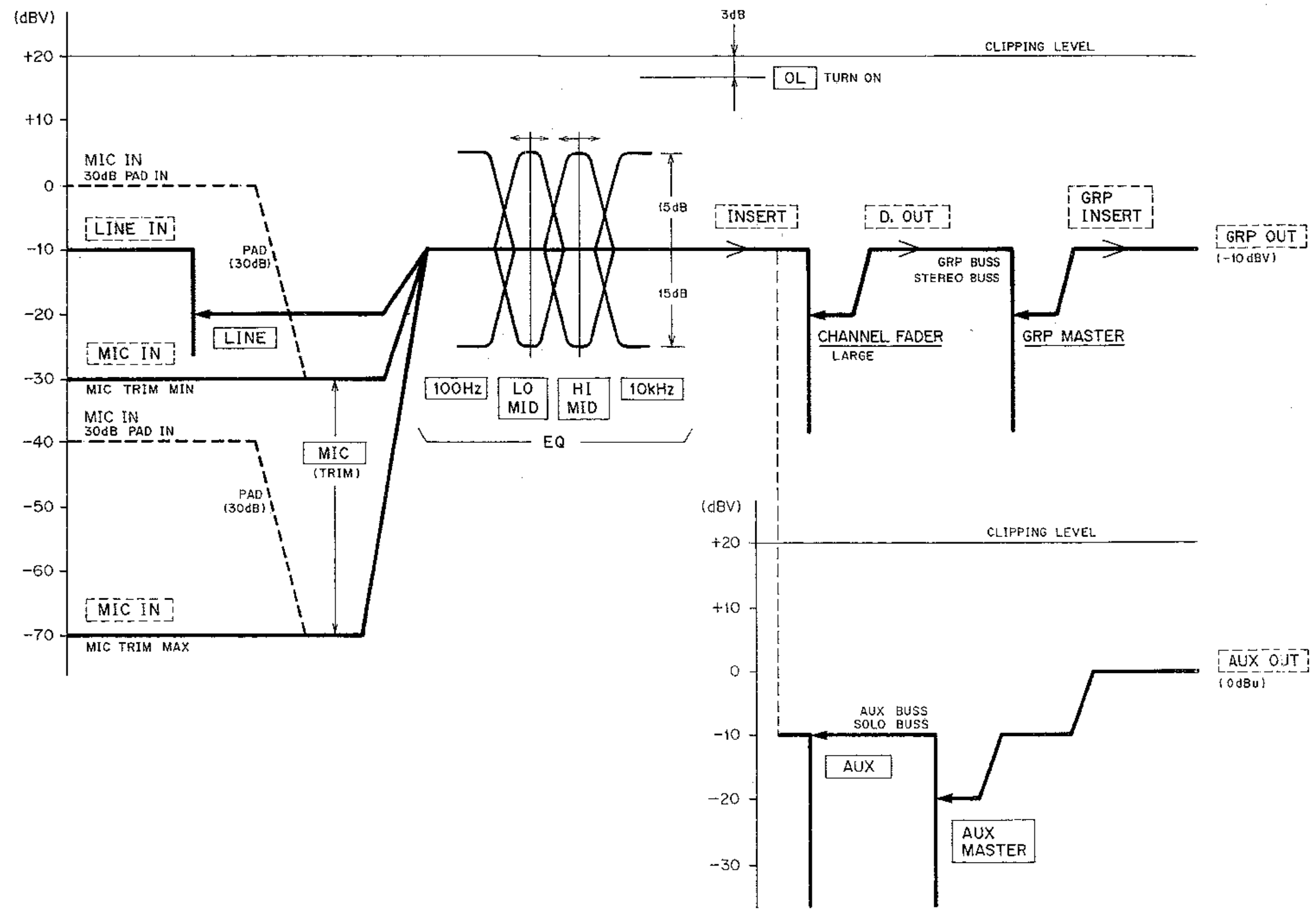


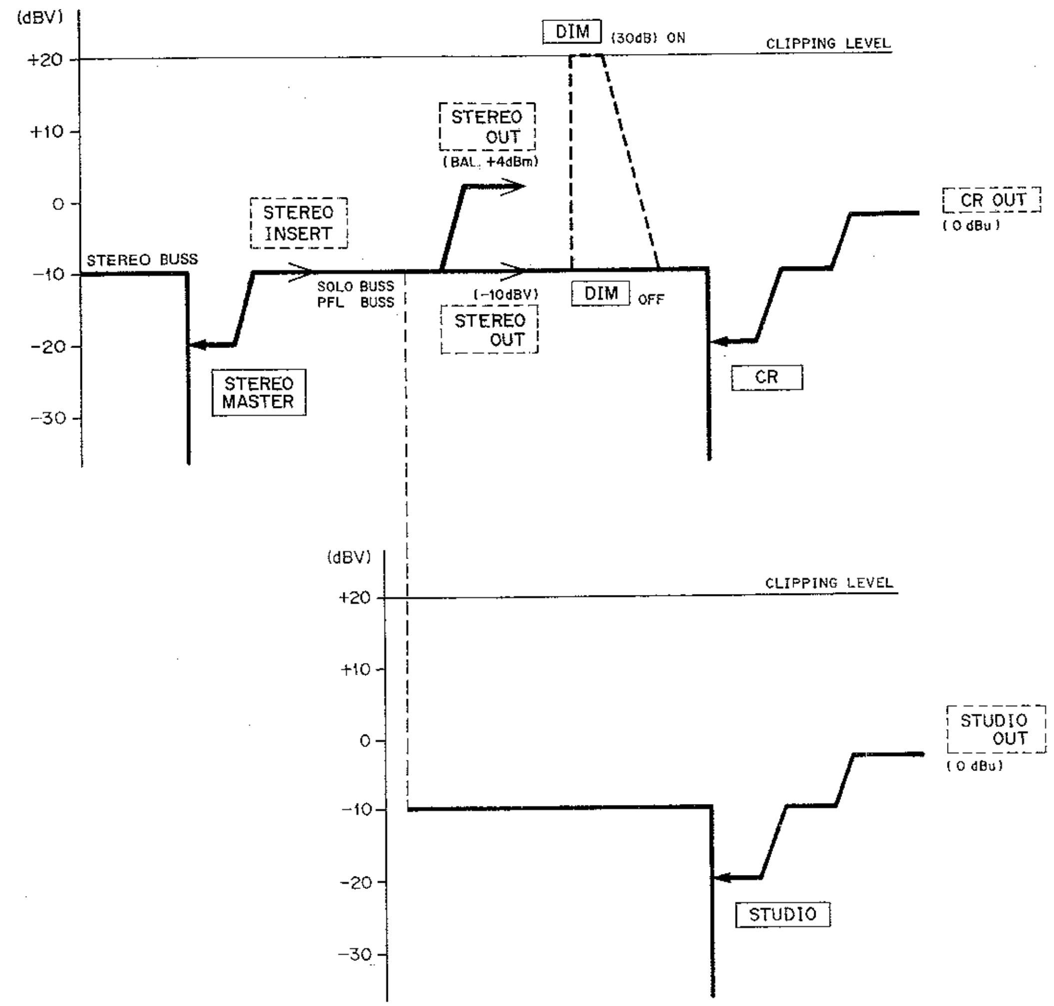
The M-3500, thanks to its outboard power supply and careful design, has a very high clip point and very low noise. In the case of the mic preamps, the noise is within 3 dB of the theoretical minimum. But it also can apply a lot of amplification to signals at various points in the path. It is your job to operate the recording system to get the best results. Your signal must be between the two limits—noise below and distortion above.

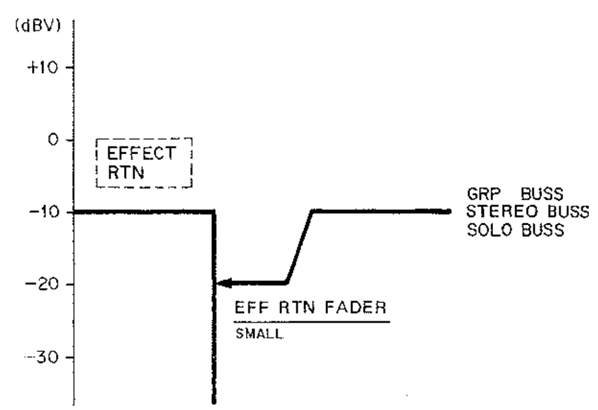


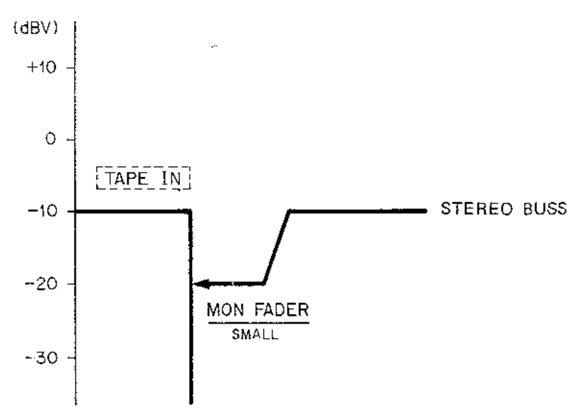
Level Diagram

A level diagram shows where gain is made and lost in a console. Note that even with a nominal −10 dBV input, it is possible to amplify signal quite a lot if faders are pushed above their nominal settings so each stage is providing gain. If the channel fader is up full, the direct output and the group feed will both be 10 dB louder than the input was. A group master pushed up full can add 10 dB to that, and if GRP is selected by a monitor, it too can add 10 dB, for a total gain of 30 dB (which, if the signal was nominal to begin with, will probably distort the stereo summing amplifier). On the diagram, the arrows show the nominal setting of a fader. The horizontal line at the −10 dBV shows what will happen if all faders are at their nominal position. In most cases, a rotary pot (for example, an Aux send) is at its nominal unity gain position at approximately "2 o'clock".









NOTE :
0 dBV = 1.0V (−10 dBV = 316mV)
0 dBm = 0.775V (+4 dBm = 1.23V)
0 dBu = 0.775V

Using Effects

Effects and signal processing allow you to develop your own unique recording style. But because there are so many possibilities, it can also be confusing. There are many different effects units on the market, all with different controls, types of inputs and outputs, and other characteristics. Read the manual for your signal processors, and the following section to get the complete story of what's possible in your own studio.

In-line Processing

The processing that's easiest to deal with happens before signal ever enters the M-3500. If a musician plugs his instrument directly into an effects device, and you plug the output of it into the console, the whole signal gets processed and only one instrument can use that processor. This is the typical use of effects pedals for guitar.

Insert Processing

This is closely related to the above method, and is typically used for compressors, limiters, and equalizers. Each channel, the output groups, and the stereo sends have special two-way send/recieve jacks called INSERT jacks. Each has the effect of inserting a signal processor into the signal path. In the channels, it is post-EQ, pre-fader; on the group and stereo outputs it is post-fader, just before the output jack. Whatever signal is travelling down that path gets diverted out the insert jack, sent to its own individual signal processor, and returned to the path it was on. This requires a special Y-cable with a stereo 3-conductor TRS (Tip-Ring-Sleeve) phone plug on one end, split to two cables with mono plugs, one for the input (send) to the device and one for the output (return). The TASCAM PW-2Y and PW-4Y can be used for this purpose. Using INSERT for processing has the limitation that only one signal can use a processor at a time. The advantages are:

1. The signal at the jack has already been preamplified and equalized. This means you can put a microphone signal through a line level processor (most processors can't preamplify microphones by themselves).

2. It's easy to move a processor from one channel to another, just by moving the insert cable from one jack to another. It's also easy to disconnect the effect by simply unplugging the insert cable.

3. Certain devices, notably graphic equalizers and compressor/limiters, are designed for in-line or insert use, dedicated to one instrument at a time. If a signal is compressed, ALL of it must be compressed (unlike reverb, where a mix of reverb and original signal is usually needed).

4. It's possible to have as many different effects devices going as you have channels--there is no limitation due to the number of auxiliary sends on a console For example, if there is a reverb setting only designed for a snare drum and not intended for anything else on the console, the insert jack may be the place for it.

Send Return Auxiliary Mix Processing

This is the most common method of effects processing, especially for reverb and delay. It allows a number of different channels to use the same effect, while allowing you to control how much effect is mixed with each channel. All input channels have access to six Auxiliary sends that can be used as effects sends. The Aux outputs can be connected to six different effects devices. The processed signals from the devices can come back into the mix via the EFFECT RETURNs, or a regular line input. In either case, they can only be heard or recorded if the processed signal is assigned to L-R or an output group.

This whole path--from the auxes to the processor and back into the console--is called an effects loop. The Aux system controls how much signal goes to the processor, and the effects returns control how much comes from it. See p. 9, "Auxiliary mixes" to see a diagram of the signal flow through the Aux system.

Setting Effects Send Levels

The goal is not to distort the device, while staying above the noise that effects units often generate. To get the best signal-to-noise from most signal processors, you should send it as strong a signal as it is designed for. With a properly set input signal (reading 0 VU on the meter when faders are set to nominal), the AUX will send the same level (nominal −10 dBV) to an Aux Out when the channel AUX and AUX MASTER are both set to approximiately 2 o'clock rotation.

If your effects device has an input level control of its own, it should be set so the meter or signal light of the effects device is just under the overload point when the M-3500 is sending it peak signals. When you want to hear or record less effect overall, make it a rule to reach for the EFFECTS RETURN first, not the Aux Master. When you reduce the effects return, you are also reducing whatever noise that effects device contributes to the mix If you cut down the Aux send, you are reducing the signal-to-noise ratio through that device.

Setting the Output Level of Effects Devices

If the effects send level has been set properly, in most cases the output level of the signal processor should be set to its nominal (unity gain) setting, or alternatively set it as high as possible without clipping the Effects Return of the M-3500 while keeping a reasonable range of control on the fader. If you can get the effects sound you want with the Effects Return (or other fader) set near 0 dB, that's ideal On the other hand, if your mix is drowning in effects even when the Effects Returns are in the low part of their range, turn down the output level of your effects device.

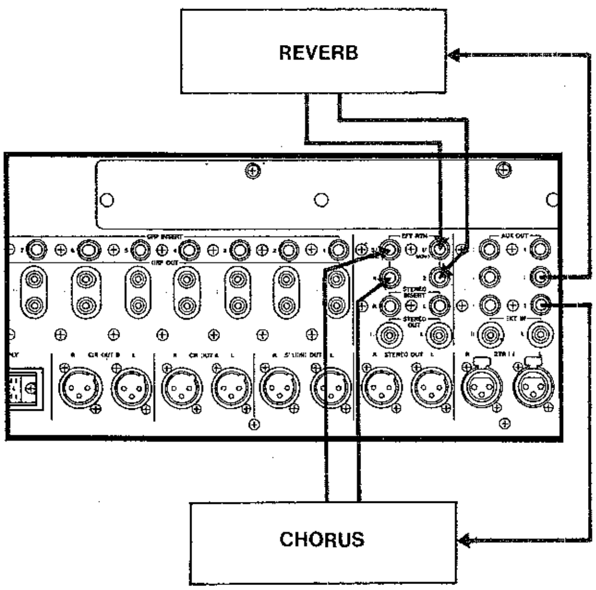
Some effects units have rear panel switches setting input and output level ranges between "+4" and "−20 dB". In this case, try setting the input to −20 (high sensitivity) and the output to +4 (full output level) to start

Setting the Mix/Balance Control on Effects Devices

If a device is being used in a send-return auxiliary mix, set its mix/balance control all the way to "wet", or full processing with no direct original signal. In send/receive processing, the "dry" (original unprocessed) signal goes down the channel path to be mixed with the effects return signal on a ground or stereo master. Therefore, you don't need any dry signal coming to the effects return. The mix balance control is set toward "dry" only when you're using the effects device at the INSERT jack or as an in-line processor.

How to Connect Effects Devices

There is no absolute "right" or "wrong" way to do this—there are several ways, each with its own consequences.



This is the most common method. Aux 3 feeds a reverb unit, which has a synthesized stereo output patched into effets returns 1 and 2. It would be used for effects off the monitor. Aux 5 feeds a chorus device with a stereo output patched into effect returns 3 and 4.

To record effects onto a track: Simply assign the effects return to the group of the track being recorded and adjust the controls for the sound you want.

To hear effects in the control room but not record them: Assign the effects return to L-R only.

To put effects into the Aux for performer cue: Assign the effects return to an unused group, (for example, Group 8), press GRP 8 as the monitor source, and MON in the corresponding AUX channel. Press PRE if you want the performers' reverb mix to be different from what you hear in the control room. Otherwise, as you move the Group 8 Monitor fader for your own needs, it will affect the send to Aux 1-4 headphones as well. See p. 16, "Cue setting: Group Output Cue" for more information.

Stereo from Mono sends: Please note that while many effects units have stereo outputs and inputs, in most cases they are not "true" stereo with separate processing on each side. The stereo inputs are designed to preserve stereo on the dry signal only on such units. In fact, the stereo inputs are typically mixed to mono, then run through a digital unit that creates a synthetic stereo reverb image. So there is no need to use two auxiliaries to feed such a unit—a single mono send will be converted by the unit into a stereo image. You do need to patch the outputs to separate effects rerutns, however, and pan them to opposite sides of the mix to get the stereo effect.

Using line or tape inputs for effects returns: If the four effects returns aren't enough, you can patch the output of effects devices either into an unused tape return or a line input. If you patch to a tape return, you will be able to monitor the effects return on the short monitor fader when FLIP is up, or send it through the main channel for monitoring (by assigning the channel to L-R) or recording (by assigning it to a group) by pressing FLIP down again.

If you patch to a Line input, you have the added feature of a Line Trim control so you don't have to reach back to the output level of the effects device. Also, depending on the position of FLIP, you can add EQ to the effects return.

In either case, be cautious of one thing: Make sure the AUX pots of the channel you're returning to are set to the off position, if that aux is feeding the same effects device. Otherwise, you will be sending the output of the effects device back to itself, which is a kind of feedback. If the effects device is a digital delay, this feedback has the same effect as a regeneration (number of echoes) control; in other cases it can cause real feedback.

Alternative Methods

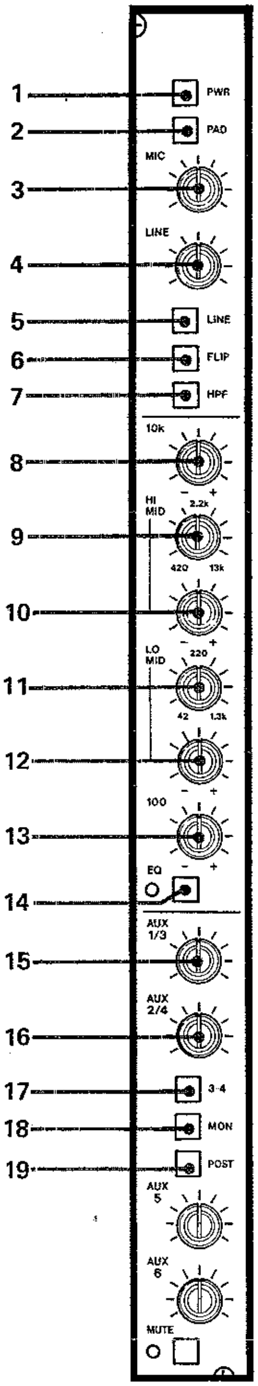
In addition to the standard methods earlier in the manual, here are some other ideas you may want to try:

Mixdown to groups, using L-R as an effects send: If you don't want to use the monitor as supplementary inputs at mixdown, it can be used as an effects send. The L-R (stereo) output will feed an effects device. The 2-track will get its signal from Group Outputs 1 and 2 instead. By pressing PRE in each monitor, the short fader becomes a stereo pre-send from the channel. The control room source must be 2TR (or EXT), since "stereo" is now meaningless as a mix reference.

Mixdown to Control Room outputs: If for some reason you need a lot of SOLO in a mix, repatch so the Control Room A or B stereo output is feeding the two track instead, and connect the control room amplifier to the outputs of the two track. This way, SOLO/PFL will affect the mix.

Use in PA applications: It's much easier to use a recording console for PA than it is to make a PA console work for recording. Since there are no tape returns, the Monitor section becomes either supplementary inputs, or an independent stage PA feed (again, by wiring the on-stage mix to the STEREO outputs, and the house PA being fed by two output groups.) The XLR outputs of the console are set to 0 dBu (.775 volts), the nominal input needed for most power amplifiers.

Use with an external submixer: If you have another mixer whose output you want to pass through the M-3500, plug it into a spare TAPE IN jack, press FLIP, and bypass the EQ. This way you can assign the output to any of the Groups or L-R, and the CHANNEL FADER will act as the submaster control for the external mixer. Use the −10 dBV (low level) outputs of the external mixer, or adjust the master output control of the mixer to that level. This path is as direct as the "Group Sub In" path on other consoles, except for the channel fader.



Input Channels

1. **PWR:** When pressed, this applies phantom power to the XLR microphone jack for that channel. 48 volts DC appears equally on pins 2 and 3 (per DIN standard #45 596) to be used by condenser-type microphones requiring external power. Since this potential is equal, it is "invisible" to standard balanced dynamic microphones, hence "phantom power". Check the manual for the mics you plan on using to make sure that his phantom method is correct before you apply power. While this 48 volt duplex phantom power is correct for most microphones, it will not operate "T power" or "AB power" mics such as the Sennheiser 405, 406 or 416.

Warning: This PWR switch must be off if any electronically balanced line input (such as the balanced output of a tape recorder or CD player) is connected to the MIC IN JACK; it could damage the output circuitry of such units. Imbalances in the cabling leading to microphones (such as an intermittent connection on one pin) can lead to voltage offset in the dynamic mic that can cause damage to the sound or to the mic itself. TURN OFF the PWR switch on all inputs that don't need it!

2. **PAD:** This inserts a 30 dB pad in the MIC signal path so that the MIC input can be used for high level signals. When it is on and the MIC trim is at minimum, the nominal input of the MIC jack increases to +4 dBm (1.23 volt).

3. **MIC TRIM:** This sets how much premplification level there is on the MIC input of the channel. It should be set high enough to amplify the source above the noise floor of the electronics, but not so high that it distorts them (indicated by the OL light in the channel). The gain range of the mic trim is 42 dB: when the MIC TRIM is set all the way to the left, the maximum input level is -25 dBm (0.044 volt). When the MIC TRIM is turned full clockwise it can amplify signals as low as -67 dBm (0.35 millivolt) to nominal level.

4. **LINE TRIM:** This sets the preamplification level on the LINE input of the channel, and works similarly to the MIC TRIM control. At full clockwise, there is 10 dB of gain in the preamp; at full "off" it is enough to attenuate any practical audio input (infinite attenuation).

5. **LINE Switch:** This switches between MIC (up) and LINE (down) as the input source for the channel. (The input routing after this to the short monitor fader or large channel fader is determined by the FLIP switch)

6. **FLIP:** This selects TAPE as the source of the channel. However, at the same time, it sends MIC or LINE (whichever is chosen by the LINE switch above) to the INP switch of the monitor fader. For example, if all the console's LINE and FLIP switches are pressed, all the main channel faders will be getting signal from their respective tape returns, and all the short monitor faders will be getting signal from the line input jacks. This

would be a typical mixdown patch. By releasing FLIP on all channels, the large faders would be getting signal from the line inputs, and the monitor faders would be getting tape returns, the typical patch for overdubbing. So by pressing FLIP, you are simply flipping the inputs from one path to another, which allows you to use the monitor section for supplementary inputs at any time, doubling the total inputs to the stereo mix

7. **HPF:** High Pass Filter. When pressed, a filter is inserted into the signal path which rolls off (attentuates) frequencies below 80 Hz at a rate of 12 dB per octave. A 20 Hz signal will be attenuated 24 dB when this is on. It is used to filter out subsonic frequencies which may be present in a signal. It is independent of the EQ ON switch.

EQUALIZER: The following controls allow you to adjust the tonality of the signal going through the main channel only. They get their signal after the HPF and send it on to the INSERT jack and CHANNEL FADER. For more information, see p. 19, "Using the Equalizer".

8. **10 kHz:** This is a high-frequency equalizer, shelving type, with a hinge point of 10 kHz.

9. **HI MID Frequency:** This changes the center frequency of the Hi Mid control, from a low of 420 Hz to a high of 13 kHz. The "Q" or bandwidth of the Hi Mid section is approximately 1.7, but varies according to the amount of cut or boost applied.

10. **HI MID Amount:** This controls how much cut or boost is applied to the band chosen by the HI MID FREQ control At the center detent position, there is no effect (flat response). Turning to the right amplifies the band, to a maximum of 15 dB. Turning to the left cuts the band, to a maximum cut of –15 dB.

11. **LOW MID Frequency:** This changes the center frequency of the Low Mid control, from a low of 42 Hz to a high of 1.3 kHz. The "Q" or bandwidth of the Low Mid section is approximately 1.7, but varies according to the amount of cut or boost applied.

12. **LOW MID Amount:** This controls how much cut or boost is applied to the band chosen by the LOW MID FREQ control. At the center detent position, there is no effect (flat response). Turning to the right amplifies the band, to a maximum of 15 dB. Turning to the left cuts the band, to a maximum cut of –15 dB.

13. **100 Hz:** This is a Low-frequency equalizer, shelving type, with a hinge point of 100 Hz.

14. **EQ On Switch:** This inserts or removes the above equalizer controls from the signal path. When the LED is ON, the EQ is in the signal path. This switch has no effect upon the HPF control.

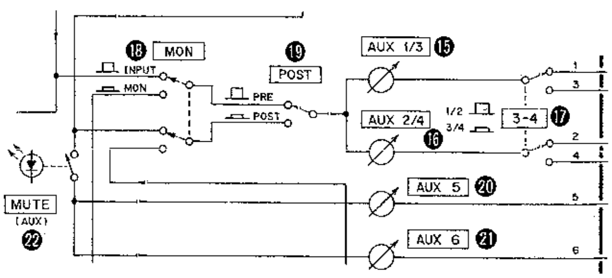
15. **AUX 1/3 Control:** This controls how much signal will be sent to either Auxiliary buss 1 or Auxilary buss 3, as set by the 3-4 switch below. Its input is determined by the settings of the MON and POST switches below. Its nominal unity gain position is approximately "2 o'clock"

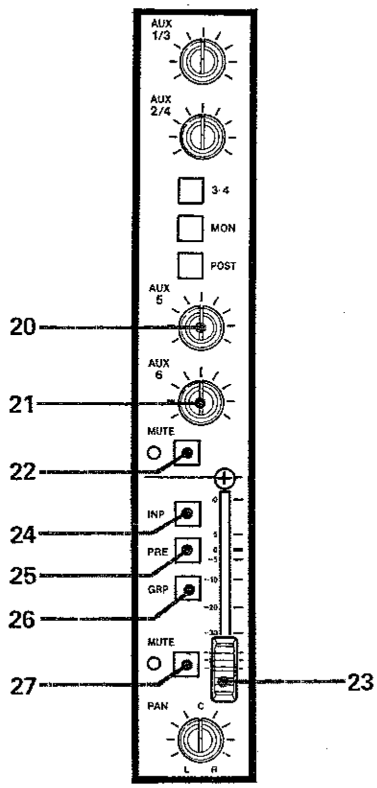
16. **AUX 2/4 Control:** This is similar in operation to AUX 1/3, except it sends signal to Auxiliary 2 or Auxiliary 4 (as set by the "3-4" switch). Its input is determined by the settings of the MON and POST switches below. Its nominal unity gain position is approximately "2 o'clock".

17. **3-4 (Auxiliary Assign):** When this switch is up, the Aux 1/3 control sends signal to the Auxiliary 1 buss, and the control beneath it sends signal to Auxiliary 2. When "3-4" is down, the two auxiliary controls above send signal to Aux 3 and Aux 4 instead. The reason for this switch is to allow you to use the auxiliary controls for different purposes on different channels (for example, two different types of effects) Another common application for this control is to use Aux 1 and 2 for headphone cue mixes during tracking and overdubbing, and use Aux 3 and 4 for effects sends at mixdown without repatching.

18. **MON:** This is an input select switch for the Auxiliaries 1-4 above. When MON is up, they get signal from the channel path. When MON is pressed, they will receive signal from whatever source is selected by the short monitor fader instead. If you are using the auxes for a headphone cue mix, MON would be pressed so the performers can hear Tape (which is usually in the monitor path). If MON is up, the cue feed will come from the channel itself. If you are using the auxes as an effects send from the monitor, MON would be down; if you need to have 4 different effects on the channel, MON should be up so all 4 aux controls can get their signal from the channel.

19. **POST:** This is a second input select switch for the auxiliaries 1-4 above. When POST is up, the signal to Aux 1-4 come from pre-fader, (either pre-channel fader if MON is up, or pre-monitor fader if MON is down). Press POST, and the auxes will get their signal from post-fader. POST means that when you change the level on the fader, the AUX 1-4 send will also change. POST is normally down if you wish to use Aux 1-4 as an effects send, because when you fade out a signal you (usually) don't want its reverb to stay in the mix. POST is up (PRE) if you are using the auxes as a headphone cue send, so the performer's mix doesn't change as you adjust the mix feeding the tape recorder.





20. **AUX 5:** This controls how much signal is sent to the Aux 5 buss and master control. It is typically used as an effects send off the main channel. Its signal always comes from post-channel fader, unless it is muted (by #22 MUTE below).

21. **AUX 6:** This is a second post-channel send similar to Aux 5

22. **MUTE (Aux):** This switch cuts off the signal feeding both Auxiliaries 5 and 6. When the LED is ON, the send is muted (off).

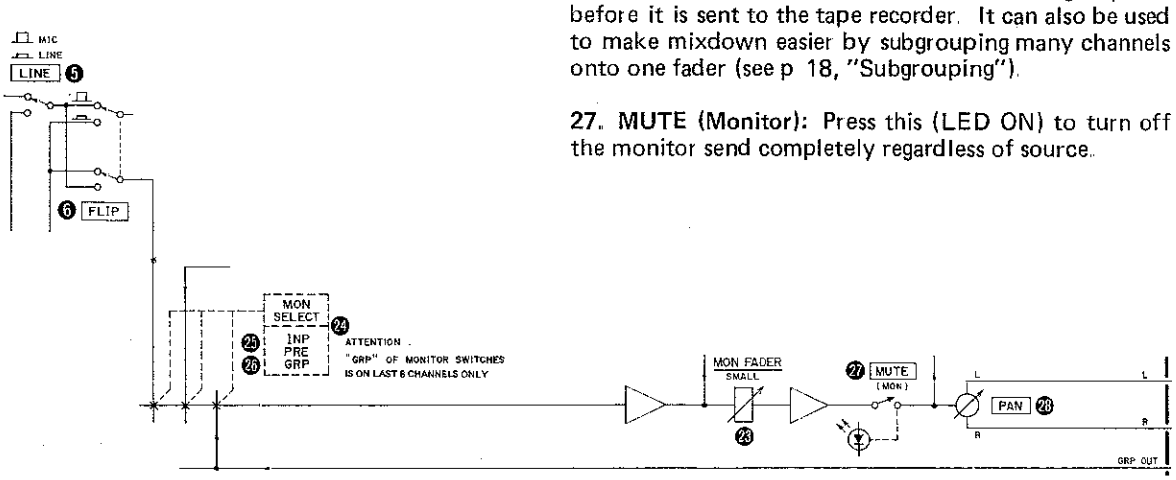
23. **MONITOR FADER:** The input to this fader is determined by the switches next to it, plus the FLIP switch (and sometimes by the LINE switch). Regardless of source, it always feeds the STEREO L-R mix, where it can be used as a control room monitor or as additional inputs to the final stereo mix.

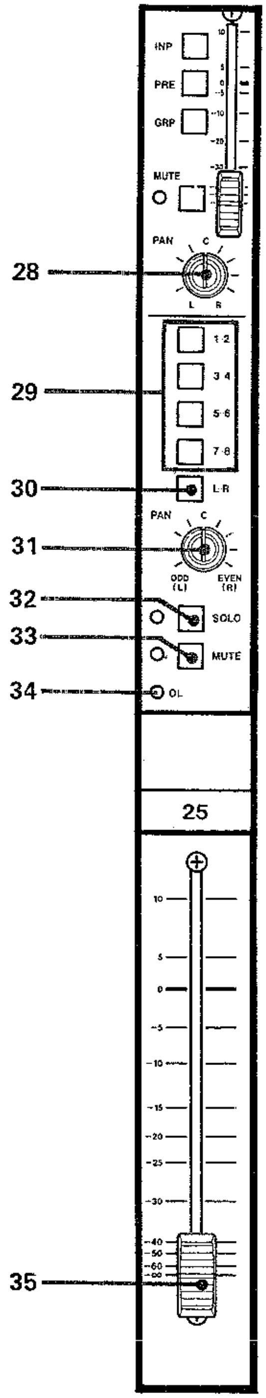
24. **INP (Input):** This is an input select switch for the monitor fader. During typical tracking and overdubbing, this switch is down so that the engineer can hear the Tape Return (provided FLIP isn't down, sending the Tape Return to the channel instead). If the FLIP switch at the top of the channel is pressed, INP gets its signal from MIC or LINE input (as determined by the LINE switch).

25. **PRE:** Pressing PRE selects the main channel pre-fader post-EQ signal as the monitor source. This is the position when you need to monitor the instrument's original signal before it passes through the channel fader, group outputs, or tape electronics.

26. **GRP (last 8 channels only):** This selects the corresponding Group Output (post-master fader and insert jack) as the source for the monitor. Note that the group meters are directly over the monitors they correspond to. This is used when you need to monitor a group mix before it is sent to the tape recorder. It can also be used to make mixdown easier by subgrouping many channels onto one fader (see p 18, "Subgrouping").

27. **MUTE (Monitor):** Press this (LED ON) to turn off the monitor send completely regardless of source.





28. **PAN (Monitor):** This allows you to create stereo control room mixes by sending the monitor signal in continuously variable degrees anywhere to the left or right of the STEREO mix.

29. **GROUP ASSIGNMENT SWITCHES:** These assign the output of the channel (large fader) to any of the eight output group busses. Each switch is an odd-even pair, and the amount of signal sent to odd or even numbered groups is determined by the CHANNEL PAN control. They may be used in any combination. Note that even if "1-2" is pressed, no signal will go from the channel to Group 1 if the CHANNEL PAN (#31) is turned hard right.

30. **L-R STEREO ASSIGNMENT SWITCH:** This sends the output of the channel directly to the stereo busses, depending on the setting of the CHANNEL PAN. This is normally pressed for final mixdown, or if the entire channel is being used for monitoring (see p. 16, Right, "Multitrack Recording and Monitoring").

31. **PAN (Channel):** This sends the output of the channel in continuously variable degrees to either side of the stereo mix (if L-R is pressed), or to odd-even sides of the Group Assignment switches (pan left for groups 1, 3, 5, and 7, pan right for groups 2, 4, 6, and 8). A PAN control is a combination "where to/how much" control, in that it controls both the level and direction of a signal.

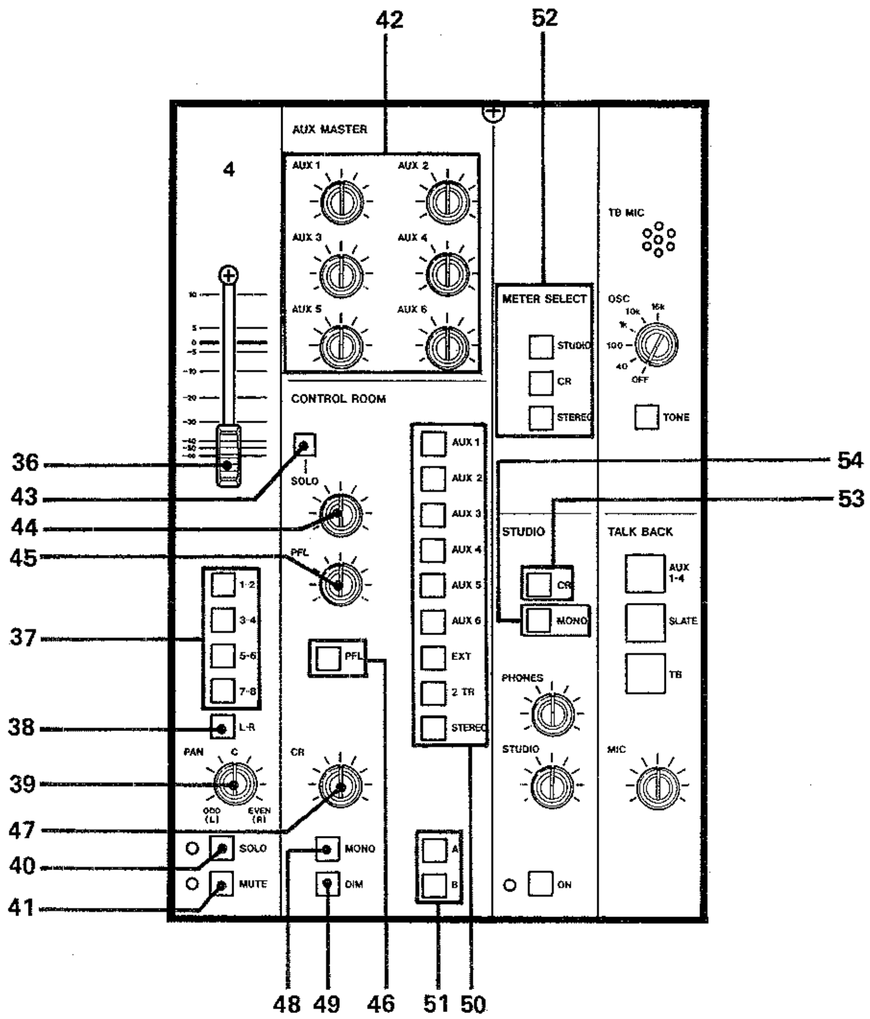
32. **SOLO Switch and LED:** When pressed, it will send that channel's signal (and only that channel's signal, if no other SOLO keys are pressed) directly to the control room monitors, cutting off any other signals to the control room. The soloed channel feeds also the studio and headphones outputs if the CR switch is pressed in the STUDIO section. The Master Solo light in the control room section will light, to alert you that Solo is active.

This SOLO can also act as a PFL (Pre Fade Listen) if that is chosen as the Solo status for the console. See p. 10, "Control Room Subsystem", and p. 29 "PFL switch".

33. **MUTE:** This completely disconnects the channel from the stereo outputs, any group outputs, and any POST auxiliary sends, when the MUTE LED is ON

34. **OL (Overload LED):** This LED will flash when the signal level in the channel (post-EQ, pre-insert, pre-fader) is 25 dB over nominal (−10 dBV) level. This is 3 dB before the channel electronics will distort. If it flashes, reduce the appropirate MIC or LINE TRIM until it stops flashing

35. **CHANNEL FADER:** This linear 100mm slide fader varies the level feeding the Channel PAN control and Group Assign switches, Aux 5 and 6, and Aux 1-4 if they are in the post-channel position (POST down, MON up). The fader is set for unity gain (level in = level out) when it is set at the "0 dB" level. When the fader is pressed to its maximum, there is 10 dB of gain added to the signal



MASTER Section

36. EFFECT RETURN FADERS: These control how much signal from the effects return inputs will be applied to the EFFECT ASSIGN switches via the EFFECT PAN control.

37. EFFECT RETURN ASSIGNMENT SWITCHES: These assign the effects returns to any of the eight output group busses. They work in the same way as the CHANNEL ASSIGN switches. Press these switches if you want to record an effect onto the multitrack. Remeber that the EFFECT PAN has an effect on the assignment.

38. EFFECT RETURN L-R STEREO ASSIGNMENT SWITCHES: These send the effects return inputs directly to the stereo busses, depending on the setting of the EFFECT PAN. This is normally pressed for final mix-down. It can also be pressed during recording to bring effects into the control room monitor mix without printing them to tape ("monitor wet, print dry").

39. EFFECT RETURN PAN: This sends the output of the channel in continuously variable degrees to either side of the stereo mix (if L-R is pressed), or to odd-even sides of the Group Assignment switches (pan left for groups 1, 3, 5, and 7, pan right for groups 2, 4, 6, and 8). Note that each effects return is MONO; a stereo effects return uses 2 separate effects return channels, and their pans are typically turned hard right/hard left in pairs.

40. EFFECT RETURN SOLO Switch and LED: These work the same as the channel SOLOs It's very handy to hear in isolation what the effects processor is doing.

Note that if you solo a channel, you will hear it "dry"; but soloing the Effects Return in addition, you will also hear all other instruments being sent to the effect at the time. If you want to hear just one instrument and its effect, you will have to solo the instrument, the effects return, and also MUTE any other channels feeding that effects device

41. **EFFECT RETURN MUTE:** This completely disconnects the effects return from the stereo outputs, and any group outputs, when the MUTE LED is ON.

42. **AUX MASTERS 1-6:** These are the last overall level controls for the Aux mixes. They get their signal from the AUX LEVEL controls in the channels. The signal then goes to the respective AUX OUT jacks on the back panel, and the AUX SWITCHES in the Control Room section beneath. Adjust these AUX MASTERS for the correct level feeding your external effects device. The nominal (unity gain) setting for these controls is approximately "2 o'clock".

43. **SOLO INDICATOR LIGHT:** If any SOLO is pressed anywhere on the console, this will light indicating that the Control Room is receiving signal from the Solo or PFL busses instead of the Control Room Source switch. Please note that if SOLO is engaged on a channel that has no signal in it, you will hear nothing in the Control Room until SOLO is turned off on that channel.

44. **SOLO LEVEL:** This sets the level you will hear in the control room when SOLO is pressed (if the PFL switch is up). Typically it is set a little above unity gain, so that a single instrument will sound about as loud in the control room as the entire mix does, allowing you to focus. The SOLO signal is stereo, coming from post-fader, post-pan.

45. **PFL LEVEL:** This sets the level you will hear in the contorl room when SOLO is pressed (if the PFL switch is down). Typically it is set a little lower, since PFL signals are often much hotter than Solo signals because they do not pass through the Channel Faders. Note that the PFL signal comes from a channel just after the EQ and Insert points, and is mono.

46. **PFL Switch:** This changes all the SOLO switches in the console between SOLO (Stereo-in-place) and PFL (Pre-Fade-Listen) modes. See p. 10, "Control Room Subsystem" for more information.

47. **CR—Control Room Level:** This sets the overall Control Room level, for both A and B outputs. Ideally the amplifier input controls should be set so that full rotation of this control does not exceed the maximum level you want in the control room.

48. **MONO:** This makes the stereo monitor mix into a mono mix. It is typically used in final mixdown so the engineer can check how the mix will sound when played back on a mono system, such as a car radio.

49. **DIM:** This cuts the control room outputs by 30 dB, and is useful when you need to answer the phone etc without disturbing the CR LEVEL control.

50. **CONTROL ROOM SOURCE SWITCHES:** These select the source of the Control Room mix (provided that no SOLO source is on to override them)

AUX 1-6: These allow you to hear the mix being sent to the AUX OUT jacks. Typically the're used when you want to check the mix being sent to the performers' headphones, or the mix being sent to an effects device. They are automatically mono.

EXT: Press this to hear the output of an external device plugged into the EXT IN jacks (typically a 2 track recorder, but also possibly a reference CD player or other device). If you are mixing down to an unbalanced recorder, you will probably monitor it here during mixdown.

2TR: This is just like EXT, except the 2TR IN jacks accept +4 dBm balanced XLR connectors from professional mastering decks

STEREO: This is the typical monitoring position since the Monitor Faders feed the stereo mix. It connects the output of the Stereo (L-R) master to the control room outputs

51. **A and B Switches:** These turn the corresponding Control Room jacks on and off. In a typical installation, Control Room A Out will be connected to an amplifier feeding large monitors so the engineer can hear a high-quality signal in detail, and Control Room B Out will be connected to a smaller system such as a hi-fi amp and bookshelf speakers so the mix can be checked on a more typical system. Only either A or B can be selected at a time ; don't try pressing both switches together. If they are both off it has the effect of a control room mute.

52. **METER SELECT:** The two mechanical VU meters on the bridge above can be switched between three outputs: STUDIO, CR (Control Room), or STEREO.

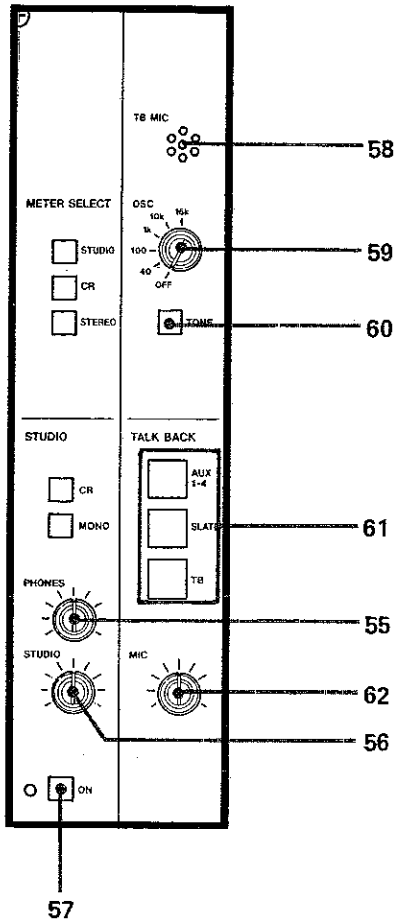
STUDIO: The meters will read the level appearing at the Studio Out jacks, including any Talkback signals. This is useful to make sure levels in the studio speakers are consistent.

CR: This is the most useful position, since it allows you to read the level of any of the Auxiliary mix outputs, either of the External 2-track inputs, or the Stereo Out, depending on the setting of the Control Room Source Switch. Meter readings will be unaffected by the CR LEVEL or DIM controls. Also, if the PFL or SOLO LEVEL controls are set to nominal, pressing any SOLO switch will show that channel's level on the VU meters.

STEREO: The meters will read the level appearing at the Stereo Output jacks.

53. **CR (STUDIO Select):** A separate pair of outputs on the back panel is designated to be connected to a separate sound system in the studio, for communication and so performers can hear playback without travelling to the control room. The normal Studio source is the STEREO mix. If CR is pressed, the studio will be fed by the source chosen by the Control Room Source Switch (Auxes, Ext, 2TR), or by SOLO or PFL, instead.

54. **MONO (Studio):** This makes the Studio outputs monophonic.



55. PHONES Level: This controls the output level of the headphones built into the M-3500 itself. This headphone mix follows whatever is chosen as the Studio source (except for talkback).

56. STUDIO Level: This sets the overall level of the Studio outputs. Note that it does not affect the level of Talkback to the studio, which is set by the MIC level alone.

57. STUDIO ON/OFF Switch: This turns on the send to the Studio amplifier. When the LED is on, the speakers are on. If it is off, it is still possible to use the Talkback system to communicate with the studio.

58. TB MIC: This is a built-in tralkback microphone for communication

59. OSC Frequency Select: This is a six-position switch choosing the frequency of the M-3500's built-in oscillator, or turning it off 16 kHz and 10 kHz are used to lay test tones on a tape so it can be adjusted for azimuth and frequency response at another studio. 1 kHz is the industry standard level reference tone. 100 Hz can be used to check low frequency response, and 40 Hz is used to lay a "slate tone" on a tape that can be heard as a marker when a tape is being fast wound.

60. TONE: This applies the tone of the built-in oscillator to the Group and Stereo busses (pre-Group and Stereo faders).

61. TALKBACK ASSIGN SWITCHES: These assign the output of the built-in Talkback mic to three different outputs:

AUX 1-4 sends the mic to the first four auxiliaries, allowing you to communicate with performers wearing headphones fed by any of those auxes.

SLATE sends the mic to all of the Group Busses and the Stereo outputs at once, so you can record an announcement on the multitrack and the mastering recorder ("Take 3", etc.).

TB sends the mic to the Studio amplifier, even if the Studio switch is off or the Studio Level is down

62. MIC LEVEL: This sets the output level of the built-in Talkback mic.

63. STEREO MASTER FADER: This fader adjusts the total output level of the stereo signal. It gets its signal from the Stereo busses, which are always fed by the short MONITOR FADERS, and by any CHANNEL FADERS whose L-R switch is down. It sends signal to the STEREO L/R OUTPUT jacks (both the unbalanced RCA jacks and balanced XLRs), the Control Room STEREO switch, the CR (STUDIO select) switch, and to the METER SELECT switch. It is set for unity gain at the "0 dB" mark, and at full up adds 10 dB of gain to the signal.

64. GROUP 1-8 MASTER FADERS: These adjust the total output level of all signals assigned to a group. They get their signal from the ASSIGN switches in the channels. They send signal to the 16 GRP OUT jacks on the back panel (2 jacks for each group, intended for 16 track recording). They also send signal to the GRP switch of the highest 8 Monitor faders, the GRP 1-8 switch on the Meter Bridge. There is 10 dB of gain available on each Group when the Master Fader is at full up. Note that although the Group Meters peak at +10, there is 28 dB of headroom in the M-3500 itself. This means that even when the meters are peaking, the circuitry of the M-3500 itself still has 18 dB of available headroom (maximum output is 8 volts before distortion). See p 19, "Gain Staging"

65. METER BRIDGE SELECT SWITCHES: These switches and indicators allow you to use the 8 LED meters of the M-3500 to measure four different sources:

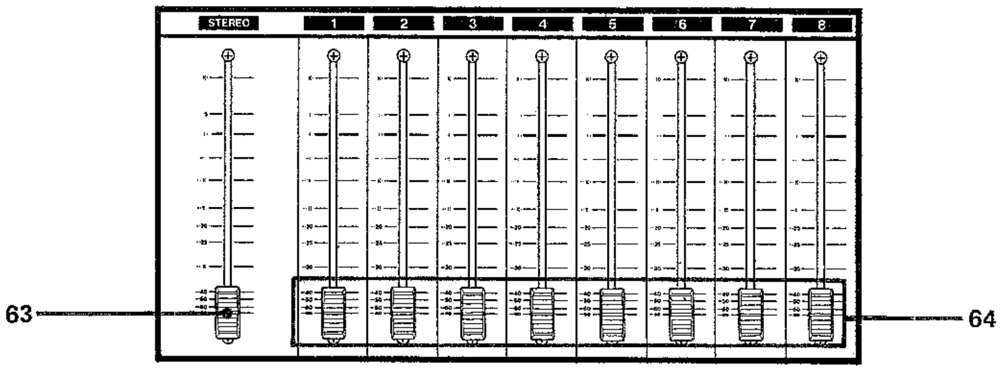
GROUP 1-8 (the normal position) measures the output of the groups

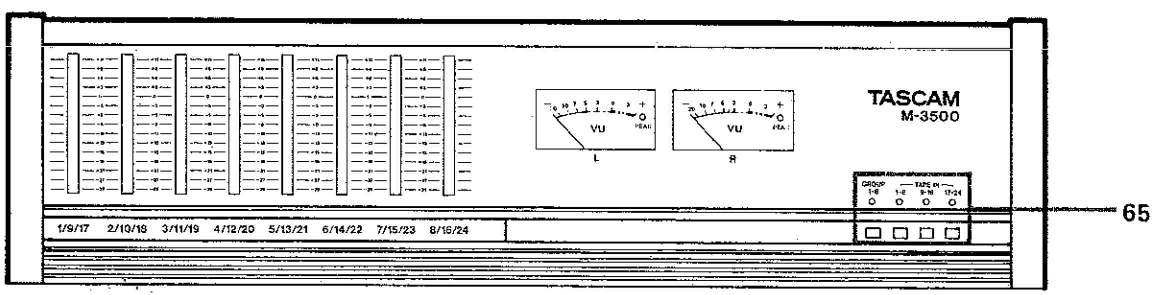
TAPE 1-8 meters the signals arriving at the first 8 TAPE IN jacks;

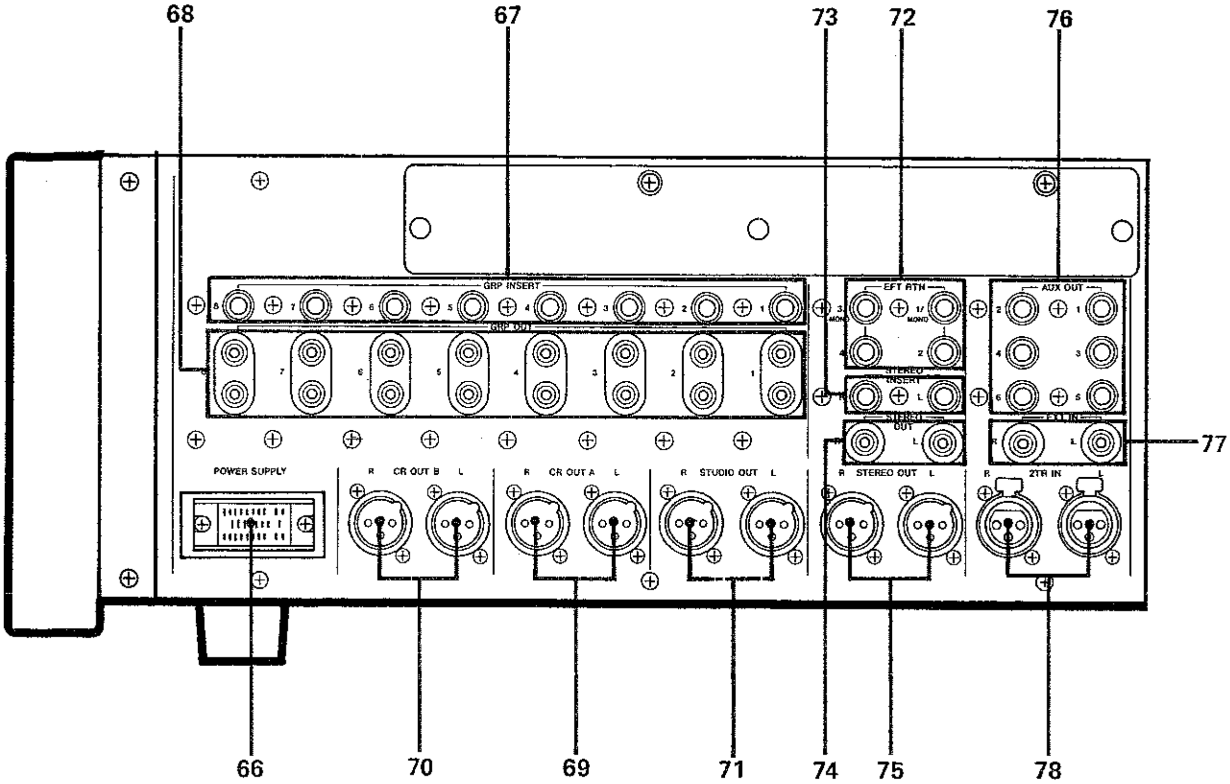
TAPE 9-16 meters the signals arriving at TAPE IN jacks 9-16;

TAPE 17-24 meters the signals arriving at TAPE IN jacks 17-24.

If you have the 32-input model, an optional expansion meter unit is available to measure the TAPE IN 25-32 signals. For more details on the expansion meter, see p 37, "Optional Accessories".







66. POWER SUPPLY: This is a multipin connector for connection to the PS-3500 Power Supply Unit only. The POWER On/Off switch is located on the external power supply.

67. GRP INSERT Jacks: These are three-conductor (TRS Tip-Ring-Sleeve or "stereo") 1/4" phone jacks inserted into the Group Out signal path between the Group Master Fader and the Group Out jacks. If nothing is plugged into this jack it is bypassed, but with the proper cable (a 1/4" TRS "stereo" to two 1/4" "mono" phone splitter cable, such as the TASCAM PW-2Y or PW-4Y Insertion Cable) you can route the entire group through an external signal processor (such as a compressor, equalizer, or reverb unit) before it goes to the tape recorder. For more information, see p. 21 "Using Effects". The INSERT point is before the feed to the GROUP METERS, and the GRP position of the highest 8 Monitor channels. Its nominal level is −10 dBV

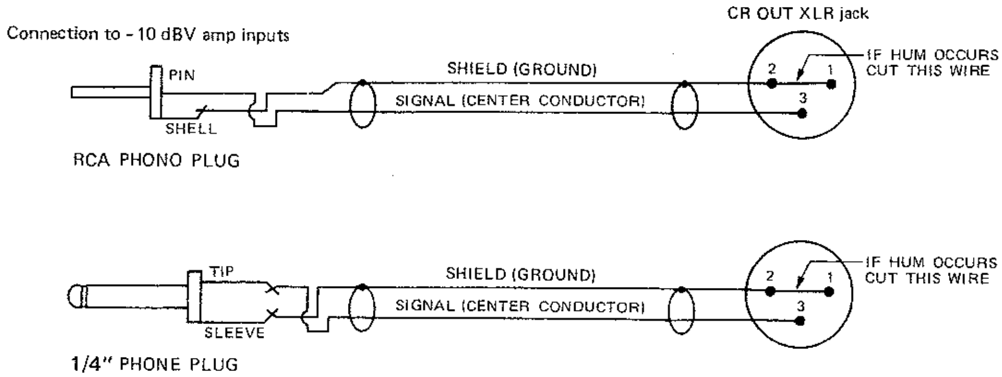
68. GRP OUT Jacks: These jacks are the outputs of the eight output groups, receiving signals from channels assigned to the groups via each GROUP MASTER FADER. The GROUP OUT jacks typically connect to the unbalanced inputs of an multitrack recorder whose nominal input level is −10 dBV (.316 volts = 0 VU). There are two output jacks for each group. These two jacks carry exactly the same signal and are intended for connection to 16 track recorders (Group 1 feeding both tracks 1 and 9, etc.). The GRP OUTS have an output impedance of 100 ohms, and a maximum output level of +18 dBV (8 volts), so they may be connected in parallel to 3 or more tape recorder inputs without signal degradation. They may also be connected to power amplifier inputs for multichannel (matrix) PA applications.

An option is available to add +4 dBm balanced outputs to each output group if required. See p 37, "Accessories".

69. **CR OUT A Connectors:** These +4 dBm balanced XLR jacks are typically connected to an amplifier powering the main reference loudspeakers. Signal comes from the Control Room Level control, via the "A" on/off switch. If your power amp does not have XLR inputs, obtain or build an adapter cable according to the diagram below:

70. **CR OUT B Connectors:** These are the same as the CR OUT A connectors, except their signal comes from the "B", and they are intended for connection to a secondary Control Room amplifier. This amp typically powers a pair of bookshelf-type reference monitors, or a headphone network.

+4 dBm to - 10 dBV Conversion Cable



71. **STUDIO OUT:** Connect these to the inputs of an amplifier powering speakers in a separate studio. Signal comes here from the STUDIO LEVEL and ON/OFF switch, except when the TB (Talkback) switch is pressed to override all switches and controls in the STUDIO section.

72. **EFT RTN Jacks 1-4:** These jacks send signal directly to the corresponding EFFECTS RETURN faders Connect the outputs of your effects devices to these jacks, although you can connect any other line input if desired. Each EFT RTN is normally independent and may be assigned to any of the Groups or to Stereo via the EFFECTS ASSIGN switches However, if a signal such as the output of a reverb is plugged into RTN 1, but nothing is plugged into return 2, the signal will go to both EFFECTS RETURNS 1 and 2. Effects Returns 3 and 4 work the same way. The input level expected by these jacks is - 10 dBV, although higher levels can be accomodated.

73. **STEREO INSERT Jacks:** These jacks are similar in operation to the GRP INSERT jacks (#67 above) They are typically used to insert a stereo compressor or other signal processing into the signal path at mixdown The insert point is between the STEREO MASTER fader and the STEREO OUT jacks below.

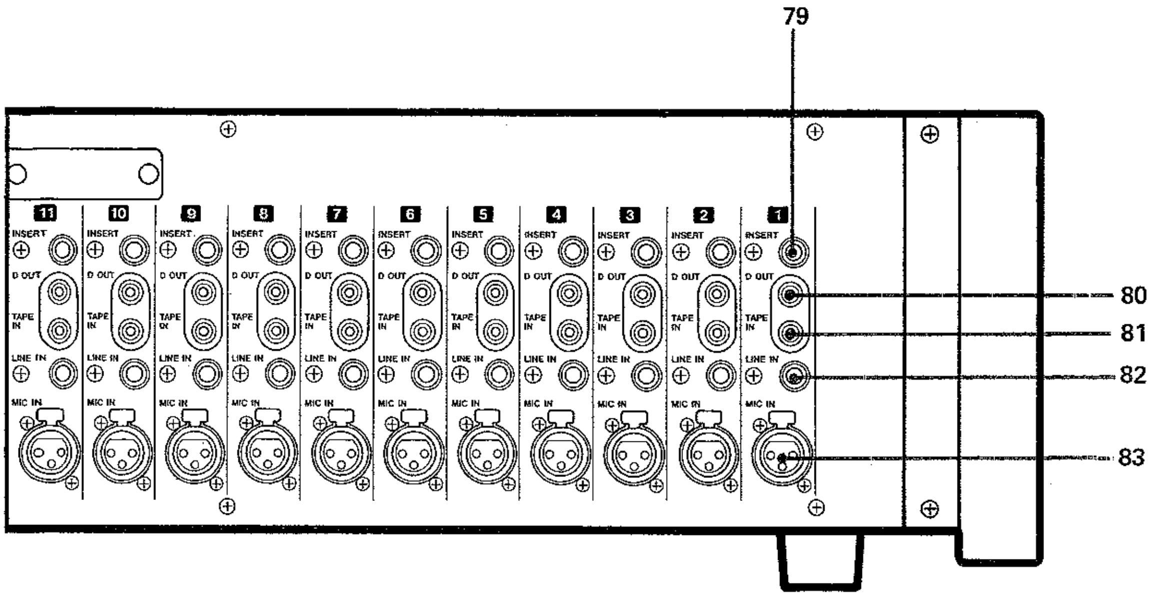
74. **STEREO OUT Unbalanced Jacks:** These jacks are typically connected to a mixdown deck with –10 dBV nominal input level These jacks and the balanced jacks can be used simultaneously.

75. **STEREO OUT Balanced Jacks:** These are for connecting the Stereo L-R signal to any +4 dBm balanced input device, such as a mastering recorder.

76. **AUX SEND 1-6 Jacks:** These are the output connectors for the six Auxiliary mix outputs of the M-3500. Signal comes here directly from the AUX MASTER controls. They are typically connected to the inputs of external devices such as reverbs, digital delays, etc. Aux 1-4 may also be used to feed a separate headphone (cue) mix. Their nominal output level is 0 dBu (.775 volts).

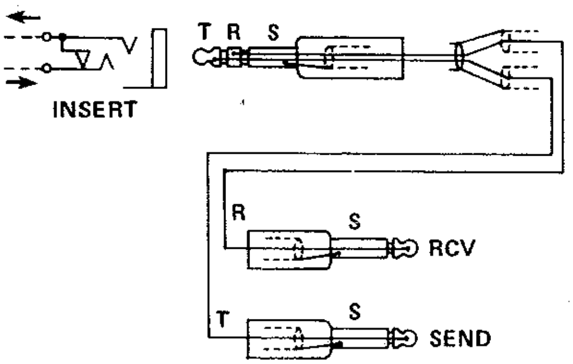
77. **EXT IN Jacks:** These provide a route from an external source (typically a –10 dBV level mastering recorder) directly to the Control Room Select EXT switch It may also be used for any source you want to hear in isolation in the control room, such as a CD player Note that since the EXT IN source can only feed the Control Room (not the Groups or Stereo busses), it is not used for any source you wish to mix together with other signals.

78. **2TR IN:** Like EXT IN, these jacks connect directly to the Control Room Select 2TR switch. They are typically connected to the +4 dBm balanced outputs of a two track mastering recorder, so playback can be heard in the control room without disturbing any settings or risking feedback by bringing the two track returns into a channel.



Input Channel Connectors

79. INSERT: The INSERT jack of each channel allows you to insert an external signal processor (typically a compressor or equalizer) into the channel path of the mixer, in between the EQ section and the channel fader. If nothing is plugged into this jack it has no effect, but with the proper cable (a 1/4" TRS "stereo" to two 1/4" "mono" phone splitter cable, such as the TASCAM PW-2Y or PW-4Y Insertion Cable) you can send signal to and from the external device from the channel. Note that the channel OL (overload) light is just before the insert jack; if the external device adds gain it is possible to clip the input electronics without the overload light flashing. For more information, see p 21, "Using Effects".



T – Tip, send signal

R – Ring, receive signal

S – Sleeve, ground

80. D OUT (Direct Out): This allows you to connect the post-fader signal of a channel directly to a recorder track or other device, without passing through a group output. It is used whenever more than eight outputs are needed simultaneously (see p. 16, Right, "Recording More than 8 tracks"). This output can also be patched into an outboard mixer for an additional post fader mix (similar in function to Aux 5 and 6).

81. TAPE IN: Every channel has an input jack designated for Tape Input (also called Tape Return) This signal normally goes to the MONITOR INP switch in the channel, so that tape may be monitored in the control room. When FLIP is pressed, the tape signal is sent through the main channel path (EQ, large fader, assignment switches) instead, for mixdown. This jack is designed for the unbalanced –10 dBV outputs. If you need +4 dBm balanced tape inputs, an optional balancing amplifier LA-3500 is available

82. LINE IN: These jacks are intended for unbalanced line level inputs (nominal signal level -10 dBV or 0.316 volts) such as synthesizers and other audio equipment. Signal goes from this input directly to the LINE TRIM control, and may be sent down the channel by the LINE switch. If will accept a minimum input level of –20 dBV ("guitar level") when the trim is at maximum gain. The maximum input level, when the trim is at minimum, exceeds any practical line level input (the TRIM comes before the input electronics, and has attenuation of over 80 dB). +4 dBm balanced signals may be connected to this jack if the source device can safely be unbalanced by an adapter (see CR OUT B for diagram). Check the manual for the device to see if this can be done.

83. MIC IN: These XLR balanced connectors are intended for connection of low-impedance microphones. However, they can also be used for line-level sources up to +4 dBm nominal level if the PAD and MIC TRIM controls are set to minimum. Signals go from this jack to the PAD, MIC TRIM, and (if the LINE and FLIP switches are both up) proceed down the channel path

The PWR switch applies 48 volt phantom power to pins 2 and 3 of this connector. Make sure PWR is OFF if any source except a phantom-powered microphone is connected. The pin assignment is: Pin 3 high, Pin 2 low, Pin 1 shield (ground).

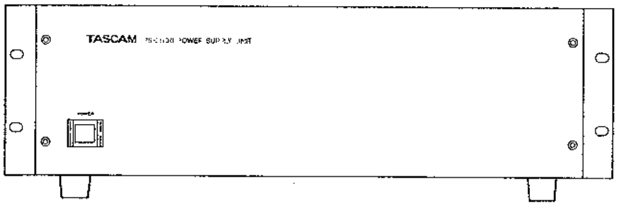
Front Panel

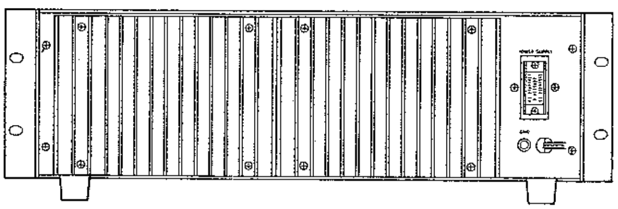
Headphone Jack: This built-in headphone amplifier gets signal from the CR (STUDIO select) switch. If the CR switch is released, the signal source is the STEREO output. This allows the engineer to continue to monitor the stereo mix in headphones, while a second engineer uses the Control Room speakers for other sources, or for SOLO. If the CR switch is pressed, the headphone signal comes from the Control Room Source Select switch, or from the SOLO or the PFL switch, although it will not be affected by DIM, or the Control Room MONO switch. To hear a mono mix in the headphones, press the STUDIO MONO switch. The headphone amplifier is rated at 100 mW. per channel into an 8 ohm load

Power Supply

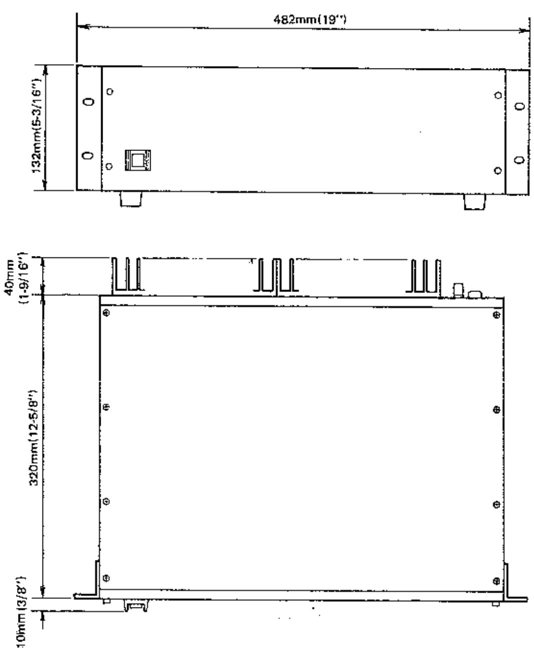
The PS-3500 Power Supply should be located within easy reach of the console. However, do not mount it directly over unshielded audio equipment, to avoid electromagnetically induced hum in such units. Do not use any multipin cable between the power supply and the console except that provided with the unit. Mount the PS-3500 in an area with proper ventilation. Connect the power supply to a good quality, noise-free AC power source of the proper rating.

The POWER switch is located on the front of the power supply.





PS-3500 Dimensions



VOLTAGE CONVERSION

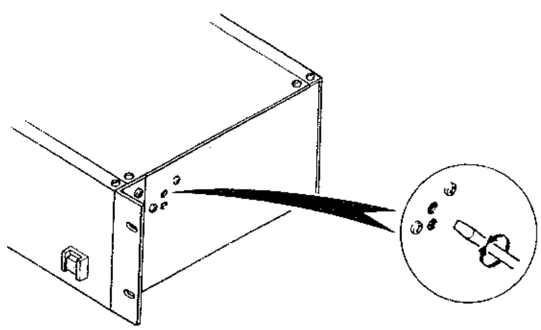
The M-3500 is adjusted to operate on the electric voltage specified on the PS-3500 Power Supply Unit.

NOTE: This voltage conversion is not possible on models sold in the U.S.A., Canada, U.K., Australia, or Europa.

For "general export" models, if it is necessary to change the voltage requirements of the M-3500 to match your area, use the following procedures.

WARNING: ALWAYS DISCONNECT THE POWER CORD BEFORE MAKING THESE CHANGES.

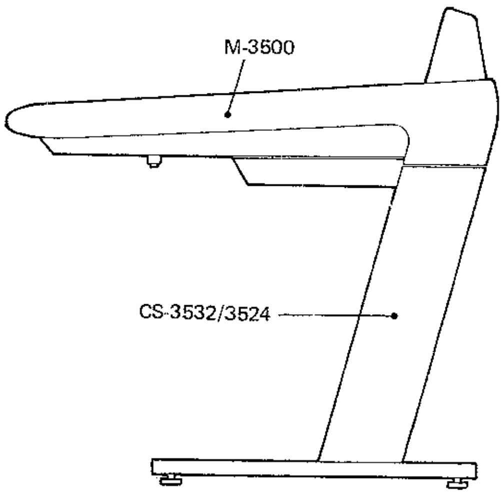
1. Locate the voltage selector on the side panel of the PS-3500 Power Supply Unit.
2. Using a regular (slot blade) screwdriver, turn the selector until the numerals corresponding to the voltage requirements for your area appear.



Optional Accessories

CS-3532/CS-3524 Pedestal

The CS-3532/3524 is a stand especially designed for supporting the M-3500 console. The CS-3532 is for the 32-input version of the M-3500, and the CS-3524 is for the 24-input version.



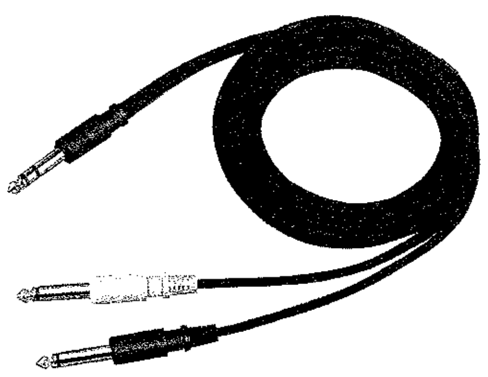
MU-3532/MU-3524 Meter Expansion Unit

The MU-3532/3524 is a meter expansion unit to be added to the standard 8 LED meters. The MU-3532 is for the 32-input version of the M-3500 and has 24 LED meters; in total you'll have 32 LED meters and be able to switch them to show all 32 channel inputs or tape returns at once. The MU-3524 is for the 24-input version and has 16 LED meters, allowing, together with the standard 8 LED meters, all 24 channel inputs/tape returns to be registered The expanded meters can also be switched to show levels feeding auxes 1-6, in addition to groups 1-8.

LA-3500 Balanced Amplifier Kit

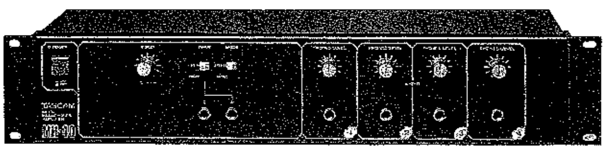
The LA-3500 is a user mountable circuit board kit, and accommodates the M-3500 console to +4 dBm group outputs and tape returns. Included are 2 input and 2 output circuit boards, each equipped with a 25-pin D-sub connector. One connector handles 8 audio signals; 2 inputs are for 16 tape returns, and 2 outputs both issue the same 8 mixes from the eight group busses to two different outboard equipment at a time.

PW-2Y/PW-4Y Insertion Cable



The TASCAM PW-2Y/PW-4Y is a connecting cable that allows signal processing such as a graphic equalizer to be inserted at specific points of the signal path of the M-3500. Its trip-ring-sleeve plug connects to the INSERT jack while its "Y'ed" end accommodates connection to the input and output terminals of the outboard equipment being used. Available in two lengths – 2 m (PW-2Y) and 4 m (PW-4Y).

MH-40B Headphone Amplifier



The MH-40B is a headphone distribution amplifier that can be used to feed four sets of studio cue headphenes and can be mounted in a 19" EIA rack

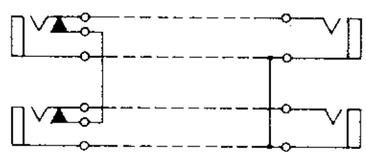
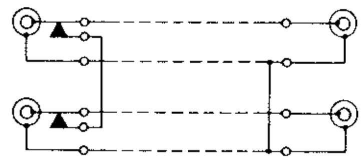
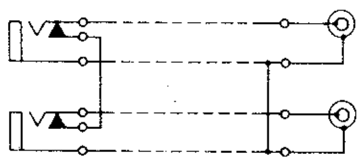
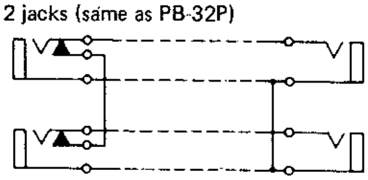
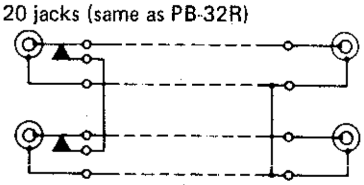
TASCAM PB-32 Series Patch Bays

(Model PB-32P)

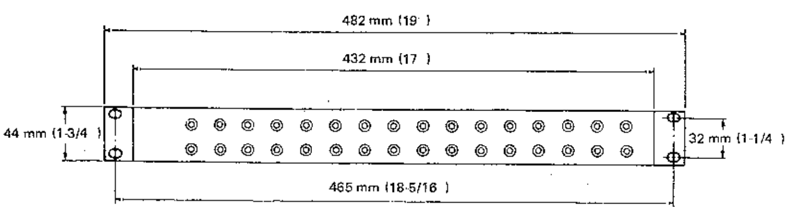
The PB-32 Series Patch Bays are ideal for any application in multitrack recording process. They are available in four basic configurations, and can be mounted in 19" EIA rack. They also feature "normalled" connections to provide the maximum in patching convenience without the need to patch through unused circuits.

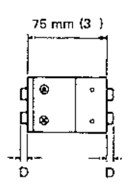
Specifications

Number of circuits:	16
Type of jacks:	RCA and/or 1/4" phone Front jacks with switch (white) Rear jacks without switch (red)
Dimensions: (W x H x D)	482 x 44 x 75 (19" x 1-3/4" x 3")
Weight:	1.3 kg (2-14/16 lbs.)

Model name	Type of jacks	Internal circuit connection (FRONT) (REAR)
PB-32P	1/4" phone jack (front and rear)	Upper Lower 
PB-32R	RCA jack (front and rear)	Upper Lower 
PB-32H	1/4" phone jack (front) RCA jack (rear)	Upper Lower 
PB-32W	1/4" phone jack (leftmost 12 jacks front and rear) RCA jack (rightmost 20 jacks front and rear)	Leftmost 12 jacks (same as PB-32P) Upper Lower  Rightmost 20 jacks (same as PB-32R) Upper Lower 

External Dimensions





D : 2 mm (1/16") (1/4" phone jacks)
7 mm (1/4") (RCA jacks)

TASCAM Cables (U.S.A. Market Only)

Cable, because of its inherent capacitance and resistance, is an active component in an audio system. There are vast differences in cable design and performance that have significant effect on the sound quality you'll get from your equipment. TASCAM Professional Audio Cables are the best available

Our cables feature very low capacitance (under 15 picofarads/foot) so they don't act as low pass filters and roll off high frequencies. The capacitance is also consistent; it doesn't change when the cable is bent or compressed. You don't get noise or degraded results when the cable has been used a while. Our cable's long term stability is provided by a special insulator that is as flexible as foam core dielectrics, but far more resistant to extreme cold or heat, and it doesn't let the center strands migrate. It also avoids the possibility of shearing th center conductor when the cable is crushed, so the cable does not suddenly fail.

Rather than loosely braided shield or spiral wrapped shield that can open up, we use bare copper braided shield with 97 % coverage. This excludes electrostatic noise (buzz) and RFI (CB interference, etc.). We also use a 7-strand center conductor: 4 pure copper strands for minimum resistance and 3 copper weld stainless steel strands for strength. The multiple strands increase flexibility and strength while offering less resistance at ultra high frequencies due to increased surface area for the "skin effect." This improves transient response.

The outer PVC insulating jacket resists abrasion, and is tightly fitted to the shield so it will not elongate The connectors are special, too. Their nickel plated brass center pins are a bit longer than most to establish good contact in all RCA jacks. The cadmium plated steel outer shell includes a gentle ridge which burnishes the mating jack when the connector is twisted to ensure good contact. For maximum RF shielding, the braid is terminated inside the shell and 2-radian soldered, not just spot soldered, for maximum strength. The plugs are clad with an oval jacket of molded plastic to further increase strength and make the ends easier to handle. TASCAM cable is available in lengths from 6 inches to 20 feet, or in color-coded sets of 8 for fast channel or function identification. TASCAM cable is also available in 500 foot spools.

If TASCAM professional cables are not available in your area, please try to find the next best cables. It really does make a difference in system performance.

ELECTRONICS

MIC IN (XLR Type Connector, Balanced)	
Input Impedance:	2.2k ohms
Nominal Input Level :	-57 dBm (1.10 mV)
Minimum Input Level;	
Trim Min:	-67 dBm (0.35 mV)
Maximum Input Level;	
Trim Max:	-25 dBm (0.044 V)
With Pad:	+4 dBm (1.23 V)
Trim Range:	42 dB
Pad:	30 dB attenuation
LINE IN (1/4" Phone Jack, Unbalanced)	
Input Impedance:	16k ohms
Nominal Input Level:	-10 dBV (0.316 V)
Minimum Input Level:	-20 dBV (0.1 V)
TAPE IN (Unbalanced)	
Input Impedance:	10k ohms
Nominal Input Level:	-10 dBV (0.316 V)
Minimum Input Level:	-20 dBV (0.1 V)
Channel INSERT (TRS 1/4" Phone Jack, Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	-10 dBV (0.316 V)
Maximum Output Level :	+18 dBV (8.0 V)
Input Impedance:	40k ohms
Nominal Input Level:	-10 dBV (0.136 V)
Maximum Input Level:	+11 dBV (3.5 V)
D OUT (Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	-10 dBV (0.316 V)
Maximum Output Level:	+18 dBV (8.0 V)
EFT RETURN (1/4" Phone Jack, Unbalanced)	
Input Impedance:	7k ohms
Nominal Input Level:	-10 dBV (0.316 V)
Minimum Input Level:	-20 dBV (0.1 V)
2TR IN (XLR Type Connector, Balanced)	
Input Impedance:	37k ohms
Nominal Input Level:	+4 dBm (1.23 V)
Minimum Input Level:	-6 dBm (0.39 V)
EXT IN (Unbalanced)	
Input Impedance:	20k ohms
Nominal Input Level:	-10 dBV (0.316 V)
Minimum Input Level:	-20 dBV (0.1 V)
GRP OUT (Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	-10 dBV (0.316 V)
Maximum Output Level:	+18 dBV (8.0 V)
AUX OUT (1/4" Phone Jack, Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	0 dBu (0.775 V)
Maximum Output Level:	+20 dBu (8.0 V)
CR OUT (XLR Type Connector, Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	0 dBu (0.775 V)
Maximum Output Level:	+20 dBu (8.0 V)
STUDIO OUT (XLR Type Connector, Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	0 dBu (0.775 V)
Maximum Output Level:	+20 dBu (8.0 V)
STEREO OUT (XLR Type Connector, Balanced)	
Output Impedance:	75 ohms
Nominal Output Level:	+4 dBm (1.23 V)
Maximum Output Level:	+25 dBm (13.8 V)

STEREO OUT (Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	-10 dBV (0.316 V)
Maximum Output Level:	+18 dBV (8.0 V)
GRP INSERT (TRS 1/4" Phone Jack, Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	-10 dBV (0.316 V)
Maximum Output Level:	+18 dBV (8 0 V)
Input Impedance:	20k ohms
Maximum Input Level:	+11 dBV (3 5 V)
STEREO INSERT (TRS 1/4" Phone Jack, Unbalanced)	
Output Impedance:	100 ohms
Nominal Output Level:	-10 dBV (0.316 V)
Maximum Output Level:	+18 dBV (8.0 V)
Input Impedance:	6k ohms
Maximum Input Level:	+11 dBV (3 5 V)
Headphone Output (TRS 1/4" Phone Jack)	
Nominal Load Impedance:	8 ohms
Maximum Output Level:	100 mW + 100 mW
Equalizer	
Type:	4-band, 2-sweep
Frequency;	
HI:	10 kHz
HI MID:	420 Hz to 13 kHz
LO MID:	42 Hz to 1.3 kHz
LO:	100 Hz
Boost/Cut:	15 dB
HPF (High-Pass Filter):	12 dB/octave at 80 Hz
OL (OverLoad) Indicator	
Flashing Level:	+15 dBV
DIM	
CR OUT Attenuation:	30 dB
Test Tone	
OSC Output:	40 Hz, 100 Hz, 1 kHz, 10 kHz, and 16 kHz
Meter	
Type:	8 LED and 2 VU meters
PEAK Flashing Level:	+10 VU
Fader	
Attenuation:	90 dB (at 1 kHz) or more
Power Requirements	
USA/CANADA:	120 V AC, 60 Hz
EUROPE:	220 V AC, 50 Hz
UK/AUSTRALIA:	240 V AC, 50 Hz
GENERAL EXPORT:	100/120/220/240 V AC, 50/60 Hz
Consumption	
32-in Model:	135 W
24-in Model:	122 W

TYPICAL PERFORMANCES

Equivalent Mic Input Noise:	DIN AUDIO/IHF "A" (150 ohm source) -130 dB/-132 dB
Signal-to-Noise Ratio:	DIN AUDIO/IHF "A"
32 MIC INS to GRP OUT:	50 dB/53 dB (150 ohm source)
24 MIC INS to GRP OUT:	52 dB/55 dB (150 ohm source)
1 LINE IN to GRP OUT:	82 dB/85 dB
32 LINE INS to GRP OUT:	62 dB/65 dB
24 LINE INS to GRP OUT:	63 dB/67 dB
1 LINE IN to AUX OUT:	70 dB/73 dB
1 LINE IN to CR OUT:	83 dB/86 dB
1 LINE IN to STUDIO OUT:	83 dB/86 dB
32 LINE INS (Monitor) to GRP OUT:	62 dB/65 dB
24 LINE INS (Monitor) to GRP OUT:	63 dB/67 dB
Headphones:	72 dB/75 dB

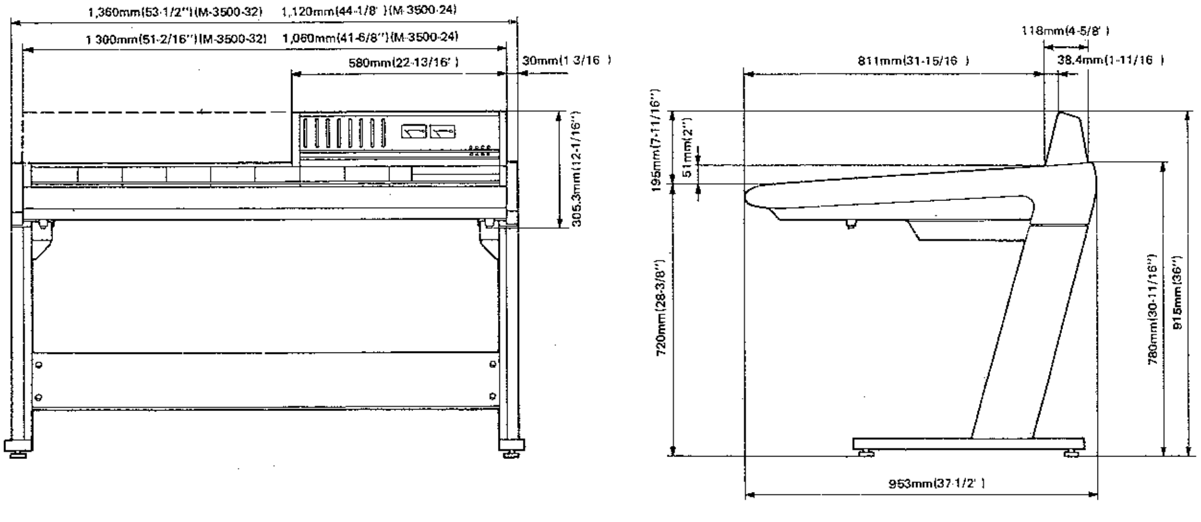
Total Harmonic Distortion (THD)	
1 MIC IN to GRP OUT:	Less than 0 018 % (at 1 kHz)
1 LINE IN to GRP OUT:	Less than 0.018 % (at 1 kHz)
IMD (Intermodulation Distortion)	
1 MIC IN to GRP OUT:	Less than 0.05 % (100 Hz and 5 kHz in the ratio 4:1)
1 LINE IN to GRP OUT:	Less than 0 05 % (100 Hz and 5 kHz in the ratio 4:1)
Frequency Response (at nominal input level)	
MIC IN to GRP OUT:	20 Hz to 20 kHz +0.5 dB/-1.5 dB
LINE IN to GRP OUT:	20 Hz to 20 kHz +0.5 dB/-1.5 dB
Headphones:	50 Hz to 20 kHz +0.5 dB/-3.0 dB
Crosstalk	
GRP OUT:	Better than 59 dB (at 10 kHz)
STEREO OUT:	Better than 59 dB (at 10 kHz)
Other Outputs:	Better than 60 dB (at 1 kHz)
Click Noise:	Less than -35 dB

OTHERS

Dimensions (W x H x D)	
Console;	
32-in Model:	1,360 x 305 x 953 mm (53-1/2" x 12" x 37-1/2")
24-in Model:	1,120 x 305 x 953 mm (44-1/8" x 12" x 37-1/2")
Power Supply Unit:	482 x 144 x 953 mm (19" x 5-11/16" x 37-1/2")
Weight	
Console;	
32-in Model:	70 kg (154-5/16 lbs.)
24-in Model:	60 kg (132-4/16 lbs.)
Power Supply Unit:	13.5 kg (29-12/16 lbs.)

In these specifications, 0 dBV is referenced to 1.0 V, and 0 dBm/dBu is referenced to 0.775 V

Changes in specifications and features may be made without notice or obligation.



M-3500 mounted on the CS-3532/3524 Pedestal Stand