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READ THIS PAGE!

We realize that you must have a powerful urge to try out your new mixer. All we ask is that you read this page NOW, and the rest can wait until you're good and ready. But do read it – You'll be glad you did.

We'll talk about gain structure a little further on, but proper setting of input gain is very important to get the lowest noise and distortion when mixing. Keep this in mind and you won't go astray.

LEVEL-SETTING PROCEDURE

Message to seasoned pros: do NOT set levels using the old "Turn the trim up until the clip light comes on, then back off a hair" trick. When a Mackie Designs mixer clip light comes on, you really are about to clip.

This procedure really works—it assures low noise and high headroom. Proper level setting is important whether you're mixing your band's PA, tracking a recording, or doing a multitrack mixdown. Please read on.

It's not necessary to hear what you're doing to set optimal levels, but it helps you to understand what you're adjusting. Plug a set of headphones into the PHONES output jack, then turn the PHONES level control about one-quarter of the way up.

Perform the following steps for each channel—*one channel at a time*:

1. Turn the TRIM, AUX send and all fader controls fully down. This prevents listeners from hearing what you're doing. It won't be interesting.
2. Set the EQ knobs to their center (usually detented) position.
3. Connect the signal source to the MIC or LINE channel input.
4. Press the channel's SOLO switch.
Note: Some of our mixers have a SOLO MODE switch in the output section. If applicable, select the LEVEL SET (PFL) mode—The LEVEL SET LED will light.

Note: The PPM series Powered Mixers are the exception since they don't have a SOLO switch. Watch the INPUT LEVEL SET light. When it blinks on peaks, that's the equivalent of a 0 VU reading.

5. Play something into the selected input, at real-world (as loud as you'd expect it to be) levels.
6. Adjust the TRIM control so that the display on the meter stays around 0. (Only the left meter will be active)
7. If you'd like to apply some EQ, do so now and return to the previous step to check the level again. **THIS IS VERY IMPORTANT!** Equalization can add several dB of boost to the channel signal, resulting in too high a level going into the mixer if you don't trim it back down.
8. Disengage that channel's SOLO switch.
9. Repeat for all of the other channels that you're using in the mix.

And now on to our regularly scheduled program.

Compact Mixer Reference Guide Introduction

Thank you for choosing a professional mixer from Mackie Designs. It has been designed to provide top quality sound, fill a wide range of applications, and be dependable night after night and session after session. Audio engineering is a combination of science and art. This book will help you to thoroughly understand what's under the hood so that you can apply the science as you develop your art.

Goal of this book

The intent of this book is to tell you a whole lot more about mixers than you'd learn from a typical instruction manual. This is a technical reference, so at times we'll get technical.

Organization

There are many interrelated topics to visit when learning about a mixer. This book is divided into three major sections. The first section is an introduction to mixers in general with an emphasis on signal flow – how a signal gets from input to output, followed by a detailed discussion of each control you'll find on a Mackie mixer. The next section is a series of application notes including hookup diagrams and suggestions for operation. The final section contains some general tips, more detailed articles on important technical subjects, a glossary of terms, and specifications and block diagrams of all the Mackie analog mixers.

Because of the way things are interrelated, you'll find some repetition in the different sections. This is isn't because the editor had a bad day, it's to allow you take it from the top or jump in wherever you're comfortable. You can expect to learn:

- The names and functions of the major building blocks of a mixer
- How to read and follow a block diagram
- Signal paths through the mixer
- Function of all the controls and some special features
- Hookups for various applications
- Mixing and operating tips

Much of what's written here applies to any mixer, even if it's from another manufacturer, although not everything you read

here may apply to your particular mixer. Different applications call for different features, so you may not have every control, input, or output described here. If you're not familiar with mixing consoles, we suggest that you read through this book at your leisure, and keep it handy as a reference when you're puzzled or you want to learn more.

Other Manuals

Since you're probably eager to hook up your mixer immediately, we've provided a Quick Start guide which illustrates the features of the specific model that you've purchased. It doesn't contain as much technical information as this book, but it has illustrations unique to your mixer designed to get you up and running quickly. Its purpose is different from that of this book, but equally important for a successful mix.

Section I: Grand Tour

Chapter 1: Mixer Anatomy IOI—The Mixer in a Nutshell

In reading about mixers, you'll run across a lot of technical terms that authors toss around as if the readers knew what we were talking about. In this section, you'll learn the terms used to describe the workings of a mixer such as preamp, bus, channel, insert, path, group, equalizer, and in-line and split monitoring. Then we'll take a signal flow safari using the block diagram of a Mackie 1604VLZ-PRO as our map.

Mixers come in all shapes and sizes from palm-sized to yards-long studio consoles, but they have a lot in common. A simple stereo mixer you'd use to record your band with two mics has the same basic function as the gargantuan consoles monsters required for building a movie sound track from 48 tracks of audio, a dozen tracks of dialog, and hundreds of sound effects. The bigger console has more inputs and outputs, but it mixes just the same.

When is it a mixer and when is it a console? That's like asking is it a violin or a fiddle? Generally, we use the term "console" when it's in the studio because it serves as the control center. In sound reinforcement or industrial applications, unless the operator is trying to impress a new-found friend hoping for some after-the-show extra curricular activity, it's usually called a "mixer" since in that application mixing is its main function. Like the concert violinist who refers to his instrument as "my fiddle", we'll use both terms here so we don't get too stuffy.

What a Mixer Does

The two basic functions of a mixer are routing and summing of audio signals. The mixer brings signals in from the outside world, allows you to manipulate them with a set of tools, and sends them back out to another unit in the recording or sound reinforcement system.

What happens along the path between in and out is a function of the controls you have available and how you use them. Faders or volume controls adjust the relative loudness of the incoming signals. All Mackie mixers include an Equalizer (EQ) section which allows you to shape the frequency balance of individual sounds to correct for deficiencies at the input, create new sounds, or help blend sounds. Other con-

trols allow you to add effects such as reverberation or delay.

A mixer is a collection of controls and indicators—controls that allow you to route, modify, and combine signals, and indicators to show you how things are set. Controls consist of knobs and switches. An Indicator can be a dot on a knob indicating its position, a button up or down, a light on or off, or a meter.

There are a lot of knobs, buttons, and lights on a mixer, but don't let that intimidate you. Many of them are repeated many times over, so all you need to learn is one set and you'll have it licked.

Playing With Blocks

A block diagram can tell you nearly everything about the functional capability of a mixer and how to use its various inputs, outputs and controls. It's like a road map. Learn to follow a block diagram and you can figure out how to get a signal through just about any console (or any other piece of equipment for that matter).

A block diagram is neither a schematic diagram, pictorial, or engineering drawing, but it contains elements of each. It won't tell you the noise figure of an amplifier or how many bits an A/D converter uses, but it will show you how a signal gets from one point to another, what switches and controls are in its path, and what inputs and outputs are available.

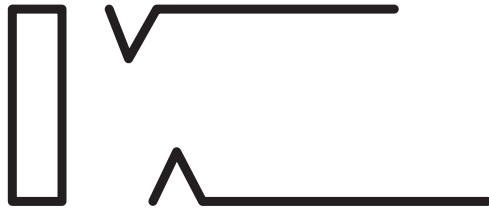
Let's tour a mixer using the 1604VLZ-PRO Block Diagram as our road map. It contains just about all the elements you'll encounter on any mixer. We won't trace every path, but enough to get you familiar with the symbols and the process, and you can take it from there.

Since a mixer contains the same circuitry duplicated many times, a block diagram will typically show only the first of many identical circuits. If a mixer has 16 input channels, expect to see full details on only one channel. If there are four subgroup busses, although they'll all be shown, you may see only one subgroup fader, insert jack, and output. You'll have to go to the spec sheet to find out how many "duplicates" there are.

Symbols

The electronic industry has developed standard symbols for components in schematic diagrams, but block diagrams aren't quite so standardized—you may have to make some guesses when first seeing an unfamiliar diagram.

In Mackie's block diagrams, input and output connectors are shown using their standard schematic symbols. Nearly all of the 1/4" jacks on the console are of the tip-ring-sleeve construction—they're illustrated like this on the diagram.



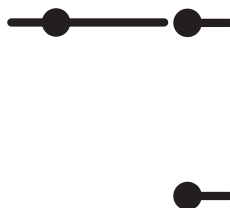
Not everybody uses such descriptive symbols, so if a spec sheet says an input is balanced and the block diagram shows a single circuit jack (or just a block or an arrow), you have to trust 'em.

We use a circle with an arrow diagonally across it to represent a rotary control. A vertically oriented rectangle with a diagonal arrow represents a slide fader.



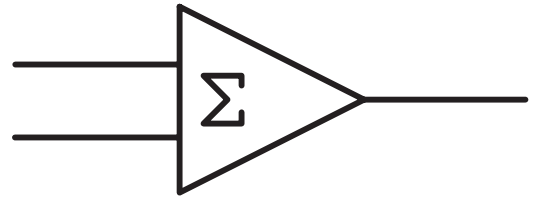
In general, an arrow through a symbol indicates that whatever the symbol represents is adjustable.

Switches are represented by a "wiper" pivoted at one end, with the other end pointing to one of the possible switch positions.



All the switches used in Mackie mixers are two-position switches, meaning that the wiper connects to one or the other of two points, one of which might go nowhere to represent that the switch in the "off" posi-

tion. The wiper can be either an input or output depending on the switch's function. You may find multi-position switches on other block diagrams.



An amplifier is represented by a triangle with the point going in the direction of the signal flow.

Mackie represents amplifiers with balanced inputs by showing two leads going into the triangle, labeled + and -, indicating which input is "hot".

An amplifier with the Greek letter Σ (sigma is the standard mathematical symbol for summation) inside the triangle is a *summing* amplifier. A summing amplifier has a gain of 1 (unity, or 0 dB) unless it needs to make up for a loss of gain in preceding circuitry. Unity gain amplifiers are often used as *buffers*, isolating an input or output from the outside world.

Mackie represents resistors or attenuators on a block diagram as rectangles, capacitors use their standard schematic symbol, and lights look like little suns. If you have an idea as what to expect, you can usually guess the meaning of a symbol.

Block diagram convention is with signal flow from left to right and from top to bottom, so you'll usually find inputs on the left side of the diagram and outputs on the right.

A dot at the intersection of two lines indicates a connection, while lines crossing without a dot are just crossing en route.

The Bus—A Mixer's Basic Framework

The term *bus* is common in many electrical-based technologies. Your computer has several busses which distribute data and control signals among the plug-in cards. The circuit breaker box in your basement has busses which distribute electrical power from the pole on the street to the various circuits in your home or studio. The word (in the singular form) has only one "s"—it

comes from the British “omnibus”, a common carrier.

The fundamental purpose of a mixer is combining signals from several sources to one or more outputs. The place where they come together is the bus. In a monophonic mixer, all of the input signals are sent, at their desired volumes, to a single bus, to which the output is also connected.

In a stereo mixer, incoming signals are combined to a pair of busses, corresponding to the mixer’s left and right outputs. In a multi-bus mixer, each input can be assigned to one or more busses, with those busses feeding individual tracks of a multitrack recorder, effects processors, the mixer’s main stereo outputs, or even another mixer.

Busses are represented by a series of parallel vertical lines, usually somewhere near the center of the block diagram. You’ll find a series of horizontal lines connecting to each bus and going off in both directions. Those arriving from the left represent paths from input sources to the bus while the lines going off to the right represent paths leaving the bus, heading toward an output.

The number of busses is one measure of a mixer’s complexity. A stereo mixer has a main left and right bus and, perhaps a couple of auxiliary busses to feed signals to effects devices or monitor speakers. A multitrack recording console typically has a large number of auxiliary busses, as well as several subgroup busses.

Hey! What gives with this 1604? The ads say it’s a 4-bus mixer, but yet there are eighteen vertical, parallel lines on the block diagram!

No, you’re not looking at the wrong diagram. The folks who write the ads are talking about one specific type of bus, the subgroup bus. Rest assured. It has the main left and right busses, several auxiliary send busses, a solo bus, and a headphone cue bus, too.

The important thing to recognize is that every bus has several inputs and one output, though that output may go to more than one destination (including another bus).

Your Tour is About to Start—Please Board the Bus

The primary signal path through the mixer begins with a mic or line input, and ultimately ends up at the main outputs. We’ll trace through this path first. We’re using the 1604VLZ-PRO as an example here, but what follows will apply to any mixer. Open up the 1604VLZ-PRO block diagram and sing along.

The channel input section is located in the upper left corner of the diagram. The signal from the microphone or line input is first amplified by the mic preamp.

Note that the MIC input is also connected, through a pair of resistors and a switch, to the *phantom power* supply which provides operating voltage for condenser microphones right from the mixer. The “global switch” note on the diagram means that a single switch controls phantom power for all of the microphone inputs together.

The rotary control above the mic preamp is the input TRIM control. The preamp boosts the low level microphone signal up to the operating level of the console. TRIM allows you to adjust the preamp for optimum gain. (we’ll tell you what’s optimum later)

Just below the mic connector is the LINE INPUT jack. Note that the signal goes from this jack through a pair of resistors, directly to the mic preamp input. These resistors form an *attenuator*, which reduces the line level signal to approximately that of a microphone. This allows the mic and line inputs to share the same high quality circuitry.

The channel INSERT jack follows the preamp. This jack looks like the line input jack though it’s used in a different way.

The arrows on the jack denote switch contacts which are normally closed, but which open when a plug is inserted. Following the signal path through the switch, you’ll see that without a plug inserted in the jack (the *normal* state), it’s as if the jack wasn’t there. Inserting a plug pushes the switch contacts apart, breaking the normal signal path. The tip contact of the jack now carries the preamp *output* signal while the ring contact becomes an *input* to the next stage.

Using what's commonly known as an *Insert cable*, (See the Tips section for a wiring diagram) you can insert a compressor or some other signal processor into the console's signal path.

A LOW-CUT filter follows the INSERT jack. The switch immediately following the filter selects whether it's engaged or bypassed. The equalizer (EQ) section follows the filter. The 1604's equalizer is always in line, but if you peek at the 8-Bus block diagram you'll see a switch that completely bypasses the EQ. This bypass switch allows you to compare the equalized and "flat" signal, and provides a more direct signal path if no EQ is required.

The MUTE switch follows the equalizer. This is simply an on/off switch which disconnects the input section from the rest of the circuit.

Next comes the FADER which adjusts the volume of the channel signal. The fader feeds the PAN control where the signal splits into two branches, each going to a separate bus. The bus ASSIGN switches tell it where to go—to the main stereo busses or pairs of subgroups. (Note: Mono mixers have no PAN controls or ASSIGN switches)

Hopping Off The Bus

The ASSIGN switches operate in pairs, as indicated by the dotted lines connecting two switches. The topmost switch connects the panned signals to the main Left and Right busses. Other bus assign switches send the signals to subgroup busses 1-2 and 3-4.

To save paper, the block diagram shows only one input channel. In reality there are sixteen identical channels, with their associated EQ, faders, and bus assign switches, each capable of being switched to your choice of busses.

Signals assigned to each bus are added together by a summing amplifier. Follow the lines going to the right from the bus and you'll find it.

The main mix summing amplifiers go through another set of INSERT jacks where a final outboard processor (typically a stereo compressor or equalizer) can be patched into the stereo mix. From here, the output level is adjusted by the left and right MASTER FADERS, and finally through line

driver amplifiers to the output connectors.

The amplifiers feeding the main L/R outputs are represented by a pair of overlapping triangles rather than a single triangle. This is Mackie's way of indicating that the amplifier has a balanced, differential output.

The left and right outputs are mixed at equal levels through two resistors to provide a monophonic mix. The MONO level control adjusts the volume of this mix, then another amplifier follows, providing the separate mono output. From the block diagram, you can see that the level of the mono output is affected by both the main faders and the MONO level control.

An Auxiliary Route

In a mixer, a signal is often "copied" so that it can be used for more than one purpose. A copy of the channel input signal is placed on the AUX busses which feed the AUXiliary outputs (or *sends*).

Locate the AUX 1 and AUX 2 controls on the block diagram and look back toward the channel input. Notice that the source for these two controls is switch-selectable. They can get their input either directly from the mic preamp or after the equalizer and channel fader, depending on the position of the PRE/POST switch. The AUX 3 and AUX 4 controls are permanently connected to the post-EQ/fader signal. The outputs of the AUX 3-4 controls can be switched as a pair to either the AUX 3-4 or AUX 5-6 busses, allowing the controls to do double duty.

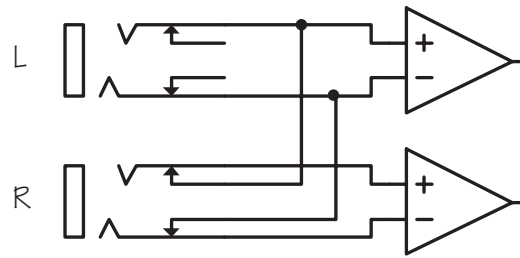
The DIRECT OUTPUT is a copy of the signal taken from the same point that feeds the post-fader auxiliary sends, therefore anything that you connect to the DIRECT jack will be affected by the fader and EQ controls.

Auxiliary returns are normally additional inputs to the main left/right bus, but they can often be switched to other busses. All of the Mackie auxiliary returns (with the exception of the PPM mixers) come in pairs since most of the things you'd connect to those inputs—effect processors or synthesizers—have stereo outputs.

The AUX RETURN jacks have normalling contacts like the Insert jacks, but they're used in a different and clever way—normalling the two jacks together allows you to

send a mono signal, via an AUX RETURN, to both channels of the stereo bus.

Notice the connections between the two AUX RETURN 4 jacks.



The Left input jack, in addition to being connected to its buffer amplifier, is connected to the switch contacts of the Right input jack. In patchbay lingo, we'd say that the left and right jacks are *half normalled*.

A mono signal plugged into the Left jack (only) gets connected, through the Right jack's switch contacts, to the right input as well. This splits the mono signal so that, to the RETURN inputs, it looks like a stereo source.

When connecting a stereo source to the AUX RETURNS, inserting a plug into the Right jack breaks the normalled connection between the jacks, feeding the left and right amplifiers independently.

Auxiliary returns on the 1604 have some specialized routings that we'll describe in detail in the section on controls, but as an exercise, follow them through the block diagram and see where they go. You'll discover that with the press of a button, you can add reverb from Aux Return 1 or 2 to a stage monitor or headphone cue fed from Auxiliary Send 1 or 2.

The Long Road from In to Out

The block diagram is one good way of getting an overview of a console. Another way is to take each building block, each control, each input, and each output and study it individually. That's where we'll go next.

After this brief introduction, when reading about a particular part of the mixer, you should be able to turn to the block diagram and see where that part fits into the big picture.

Being able to follow a signal as it flows from input to output really helps when you know something's going in and it's not coming out where you expect it—and that happens

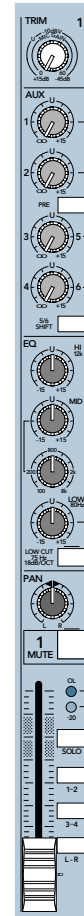
to even the most experienced engineers at times.

Now that you've looked at the map, let's hit the road, Jack. On to the hardware:

Strolling Down The (Channel) Strip

The vertical column of knobs, buttons and circuitry associated with each input channel of the mixer is called a *channel strip* or *module*.

On older consoles, it was truly an independent and removable module. Maintenance was easy, but it's expensive to build a console with so many removable parts. At Mackie, we've improved reliability and reduced maintenance to near zero, allowing us to build our mixers with of the all channel strips in a single unit or a large module comprised of several channels. The familiar term *channel strip* or *module* tends to hang around, however, since regardless of actual construction, it just looks like a module.



The Amplifier—The Mixer's Basic Building Block

You can't read about a mixer for very long without encountering the term *gain*. An amplifier is the building block which supplies that gain. Gain is the amount a signal is amplified. It's always a ratio (10 times or 100 times), and, in audio, is almost always expressed in decibels (dB).

$$\text{Gain} = \frac{\text{VoltageOut}}{\text{VoltageIn}}$$

Or, in dB:

$$\text{Gain(dB)} = 20\log \frac{\text{VoltageOut}}{\text{VoltageIn}}$$

An amplifier that boosts a 1 millivolt (0.001 volt) microphone signal up to the mixer's internal operating level of 0.775 volt has a voltage gain of 775, or about 58 dB. (use your scientific calculator or dig up a dusty old math book). A handy rule of thumb is that doubling the gain adds 6 dB, while cutting it in half is the same as subtracting 6 dB.

Some amplifiers have a gain of 1 or *unity* (no amplification), but serve to isolate one circuit from another or to match impedance. A unity gain amplifier is often called a *buffer*.

The Mic Preamp

The first thing that a microphone signal encounters when entering the console is an amplifier with up to 60 dB of gain, adjustable with the TRIM control. It's called the preamplifier since it's the first amplifier in the chain. The preamp is a hard worker. It must amplify the mic signal from a quiet dulcimer or a loud kick drum, doing both without adding a lot of noise or distortion. This requires that it work well providing as much as 60 dB of gain or as none (unity gain).

A line-level input typically has a gain range of -20 (that's 20 dB of attenuation) to +20 dB. On a Mackie mixer, line inputs pass through the preamplifier stage, but are first attenuated to bring the line level signal down to the working range of the TRIM control.

When the gain is properly adjusted with the TRIM control, peaks as much as 20 dB above the average level can be amplified without distortion. Too much gain and loud peaks can distort, too little gain and the preamp's internal noise will be amplified further on in the chain. Just right is . . . just right.

Pots—The Knobs We Love To Twist

A *potentiometer*, or *pot* for short, is a variable resistor which divides a voltage in relation

to the position of a contact sliding along a resistive element. Most potentiometers are round and the resistance changes when you rotate a knob. The *linear faders* used as main level controls on most Mackie mixers are potentiometers built with the resistive element in a straight line. Equalizer controls are also pots, too—they adjust the gain at specific frequencies.

PAN Pots

All stereo mixers have Pan controls in the channel strip. The pan pot (short for “panoramic potentiometer”) consists of two volume controls ganged together to a single knob, operating on the same signal. With the knob turned fully in one direction, the signal going into one of the potentiometers comes out at full volume while the signal through the other is completely off.

Between the extremes, the signal is divided between two outputs which ultimately end up at the left and right speakers. The difference in volume level of the signal between the two speakers creates a “phantom image” of the sound source in the space between the speakers.

Groups and Subgroups

In a basic stereo mixer, the pan controls route the input channels to a pair of busses which feed the main outputs. The eight busses we speak of when we refer to an 8-Bus console are called *group* (or sometimes *subgroup*) busses. In a multi-bus console like the Mackie 1604VLZ-PRO, switches connect the panned outputs to selected pairs of subgroup busses or the main stereo busses. In the studio, subgroup busses are often connected to the inputs of multitrack recorders.

The Mackie 8-Bus, SR, and 1642VLZ-PRO mixers provide multiple output jacks for each subgroup bus for convenient connection to multitrack recorder tracks. See the chapter on Hookups for the whole scoop.

Most multi-bus consoles also have switches to route the group busses to the main stereo bus. Group outputs then become true *subgroups*, and the group faders become *submasters*. A submaster fader simultaneously adjusts the level of all the sources assigned to that subgroup, retaining the balance that you've established for those sources.

If you assign the background vocal tracks to a pair of subgroups which are in turn assigned to the main stereo mix, You can balance, pan and EQ the tracks, then adjust the background vocal level in the mix using the subgroup faders rather than a whole handful of individual channel faders.

Auxiliaries

AUX busses are typically used to create mixes on AUX SENDs (outputs) that don't go directly to tape or speakers, but are still important to the mixing process. Common applications are to send signals to an effect processor, or to create a custom monitor mix for a performer.

Tape Monitoring in Multitrack Recording

OK, we know you're probably hip to the latest hard disk technology and hope you never see rusty plastic ribbon again, but the term "tape" is just so comfortable—you can record on anything you want using your Mackie—honest.

The thing that most distinguishes a multitrack recording console from other mixers is its tape monitoring facilities. With a simple stereo mixer, you mix a group of microphones or other sound sources so that they blend well, perhaps add some EQ and reverb, and send the mix to a stereo (2-track) recorder or PA system. What you mix is what you get.

In multitrack recording, instruments and voices are recorded on their own tracks, often one at a time, as the song moves through the production process. Not until all the tracks have been recorded do we "print" the final stereo mix.

When recording new tracks, you'll need to hear the previously recorded tracks so that you can make sure the new parts fit—you're mixing, at least roughly, while you're recording. A multitrack recording console needs to be able to do both jobs at once.

It's desirable to record each track as "hot" as possible in order to achieve the best signal-to-noise ratio when using an analog recorder or preserve low level resolution when recording digitally. This means that tracks will all be recorded at approximately equal volume even though that's not how you want them to sound in the final mix.

While laying the tracks, however, you must to be able to construct a mix that's musical enough so as not to distract the talent, the producer, or the engineer.

A multitrack recording console has a special section dedicated to monitoring during the tracking phase of the project. At all times, you can conveniently hear a mix (or, using AUX busses, more than one mix) of the multitrack recorder outputs. This allows us to record at optimum levels while monitoring tracks at any level we choose, temporarily boosting a track, muting a track, or, when overdubbing, listening to what we're sending to the recorder.

Multitrack consoles have a set of dedicated TAPE RETURN inputs, one for each recorder track. These are routed to the monitor section when tracking, but are also often available as additional inputs during mixdown.

There are two different architectures used in the design of a multitrack console, both related to the form that the monitor mixing section takes. These are called "split" (or "side-by-side") and "in-line" monitoring. Split monitoring is sometimes known as "British style" (not to be confused with "British EQ") because it evolved from the first multitrack consoles designed by British studios. In-line monitoring is sometimes called "American style" because it was developed by an early US console manufacturer.

The Mackie 8-bus is in an in-line design. While Mackie doesn't make a split console, in the Hookups section, we'll show you how to set up a 4-bus mixer for split multitrack monitoring.

A split console has a distinct group of controls dedicated to multitrack monitoring during the tracking phase. The monitoring section is clearly recognizable as a separate mixer with, at minimum, level and pan controls for each track, sometimes minimal EQ and an AUX send or two, but rarely the full set of channel strip controls.

While a split console may have 36 or more input channels, its monitor section is almost always smaller, typically 16 or 24 inputs, corresponding to the number of tracks on the recorder with which it was designed to work. A 48-channel monitor section on a split console, while not unheard of, is pretty unwieldy.

Expansion to more tracks, unless you planned ahead often involves console replacement when the recorder is upgraded. (Remember when you said “Eight tracks is all I’ll ever need.”?)

Today, most multitrack consoles are the in-line style—they’re less expensive to build and usually will live through a couple of recorder upgrades before you outgrow it.

The in-line design has two big advantages over the split console:

First, without the separate group of monitor controls, it can be physically smaller for the same number of inputs and outputs. Second, since every input section has a set of recorder monitor mixing controls, you are able to monitor and mix as many recorder tracks as you have input channels.

An in-line console, like a split console, provides a separate mix (often called Mix-B as it is on the Mackie 8-Bus) for the recorder returns. Unlike the split console though, all of the controls and circuitry associated with the monitor section are located within the channel strip.

Regardless of the architecture, the monitor mix end up going to the main left-right buses. That’s what you normally listen to in the control room when tracking and it also feeds the stereo mixdown recorder—handy for recording rough mixes while working.

With today’s modular digital multitrack recorders and easily expandable audio workstations, adding more tracks isn’t the huge investment that it used to be. It’s often as simple as borrowing another recorder from a band member and hooking it up. With a split console, even if you have enough channels to mix all the extra tracks you hauled in for the project, you might have to kludge some additional monitor inputs when tracking. With an in-line console, if you have enough inputs to mix your tracks, you’ll have enough inputs to monitor them.

Now that you’ve had the ten dollar tour, let’s move on to specifics about Mackie mixers. In the next section, we’ll look at all the input and output connections, then follow with details on what the controls do.

Chapter 2: Inputs, Outputs, and Related Controls

Mackie E-Z Interfacing

Concerned about levels, balancing, impedance, polarity, or other interface goblins? Don't be. On your Mackie mixer, you can connect almost anything almost anywhere, with nary a care. Here's why:

- Every main input and output is balanced.
- Every 1/4" input and output jack also accepts unbalanced connections.
- Every input is designed to accept virtually any output impedance.
- The main left and right mix outputs can deliver +28 dBu into a 600 ohm load. (that's hot!)
- All the other outputs can deliver +22 dBu into a 600 ohm load (that's plenty).
- All the outputs are in phase with the inputs.

All Mackie mixers have both microphone and line level inputs. Some channels have both, some channels have only line level inputs. Consult your mixer's quick start guide (or just look at the connector panel) for the exact arrangement. Here's the rear panel of the 1604VLZ-PRO:

A microphone typically requires 20 to 60 dB of gain in order to bring its signal up to the mixer's operating level. A line level signal requires little if any gain, and may need to be brought *down* in level to match the mixer. So, while there's a wide adjustment range for both mic and line inputs, the mic inputs have higher gain.

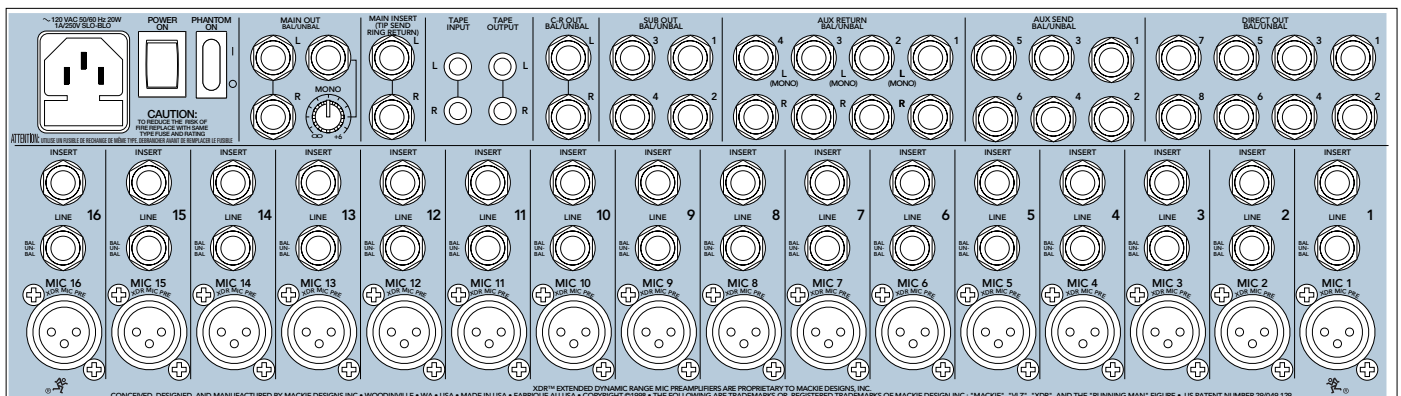


Don't forget the **Level-Setting Procedure**. Mic and line inputs are happiest when running at the optimum levels. This procedure will get you there.

Check the Tips section and Glossary for more information about terms we'll be using here like operating level, impedance, balanced, unbalanced, XLR, TRS, what's hot and what's not.

Mic Inputs

Mackie mixers include phantom-powered, balanced microphone inputs just like the big studio mega-consoles, for exactly the same reason: A fully balanced circuit is excellent at rejecting hum and noise. You can plug in almost any kind of mic that has a standard male XLR-type connector.



Mic? Line?—What's The Difference?



Microphones produce a relatively low voltage, generally in the range of 1 to 250 millivolts, depending on the sensitivity of the mic and how loud the sound is. Voltages in the 0.15 to 3 volt range are considered to be line level. These aren't rules or standards, just ways of talking about two widely different operating voltage ranges, both of which are common in audio.

Professional ribbon, dynamic, and condenser mics will all sound great through these inputs. The Mackie mic preamps will handle almost any mic you can toss at them without overloading.

XLR Input Connectors—Mic Wiring

Nearly all professional mics are designed to connect to a low impedance input and employ balanced wiring terminating in an XLR-style plug. You'll find an illustration and wiring diagram in the Connectors section. We use the wiring convention:

Pin 1 = shield

Pin 2 = positive (+ or hot)

Pin 3 = negative (- or cold)



Even though microphones produce alternating current (ac), they have a definite polarity. Be sure that you don't reverse the wires between pins 2 and 3.

Did you pick up a cool retro mic at a yard sale that has a 1/4" phone plug instead of an XLR? Chances are it's a high impedance mic and it will require a matching transformer in order to sound its best with your mixer. Plug-in transformers available from several sources will not only match the impedance and signal level, but will adapt the plug too.

Phantom Power

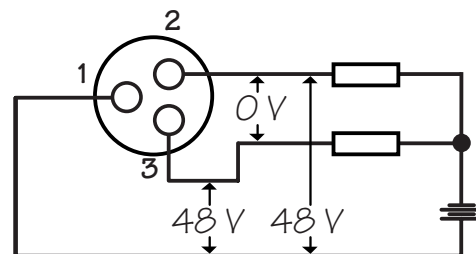
Most condenser mics require phantom power, a system whereby the mixer supplies a dc voltage to the mic's electronics through the same wires that carry audio.

The phantom power on all Mackie mixers is globally controlled by the PHANTOM switch on the rear panel (one switch for each group of 8 mics on the 8-Bus). A PHANTOM LED on the front panel will remind you that phantom power is turned on.

"Phantom" owes its name to the ability to disappear when it's not needed, for instance when connecting dynamic mics (Shure SM57/SM58, for instance) that don't require external power.

The universal standard for phantom power is 48 volts. This voltage is supplied by all Mackie mixers with the exception of the PPM series, which provide 15 volts. Some microphones will operate just fine on lower voltages while others require 48 volts. Check your mics' specs.

Here's what the phantom powering circuit looks like:



There's 48 volts between pins 1 and 2 and between pins 1 and 3. Since pins 2 and 3 are both at the same dc potential, the voltage difference between them is zero. That's why an unpowered mic doesn't "see" the voltage.



Dynamic and ribbon mics don't like to see dc voltage applied to their sensitive insides. Normally it won't be, but a shorted or miswired mic cable can cause the full voltage to appear across the sensitive microphone element. Be sure that your cables are in good condition before applying phantom power.

Also, unless you know for certain that it is safe to do so, never plug single-ended (unbalanced) microphones, instruments or electronic devices into the MIC input jacks if the phantom power is on.



Practice safe phantom powering! Plug in your mics with phantom power switched off! The XDR

preamps used in many Mackie mixers have extra protection against damage from "hot plugging", but your mics don't. Get into the habit of plugging mic cables in with the phantom power off, then turning it on.

Line Inputs

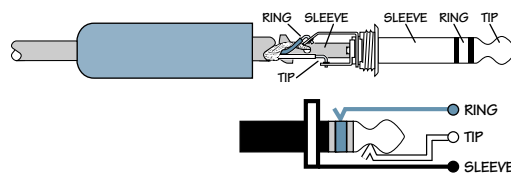
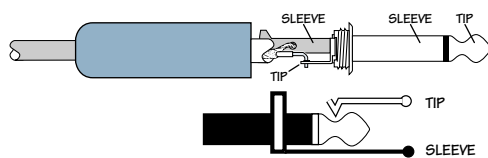
Line inputs on Mackie mixers are on 1/4" TRS (tip-ring-sleeve—the parts of the mating plug) jacks. All Mackie line inputs are balanced and present a moderately high impedance load, around 10 k Ω , depending on the model and the particular input. As we've seen from our block diagram tour, LINE IN jacks on channels with both line and mic inputs share circuitry (but not phantom power) with the mic preamps.

You can use these inputs for virtually any line level signal. With proper adjustment of the TRIM control, they'll accept signals over a range of approximately -45 to +16 dBu.

Some Mackie mixers are equipped with stereo line inputs. These have two line level input jacks which share a common set of controls. To learn how signals are routed from these inputs consult the block diagram or quick start guide for your mixer, or check the Controls section of this manual.

Always be sure to perform the **Level-Setting Procedure** when connecting to LINE inputs which have TRIM controls, and check the level and adjust it externally on the stereo inputs without TRIMs.

To connect balanced sources to the line level inputs, use a 1/4" tip-ring-sleeve (TRS) plug, the type found on some stereo headphones:

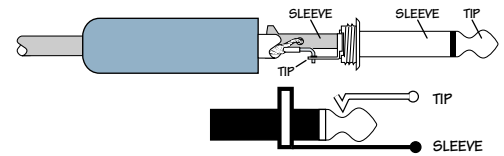


To connect unbalanced line level sources, use a cable with 1/4" mono (TS) plugs. In music store lingo, that's an "instrument cable" or "quarter inch patch cord". When a TS plug is inserted into a TRS jack, the Ring (low signal) and Sleeve (ground) get connected together.

See the Tips section for more information on balanced and unbalanced connections.

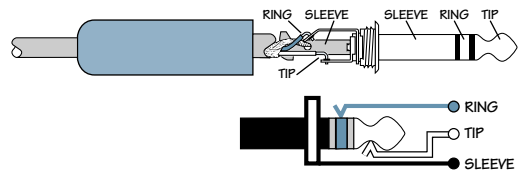
Line In (For Mono Channels)

These inputs can accept 1/4" TRS balanced and 1/4" TS unbalanced plugs from any line-level instrument, effects device, or tape player. The line inputs share circuitry (but not phantom power) with the mic preamps, and can be driven by virtually any line-level signal.



1/4" TS (Tip-Sleeve) unbalanced wiring:
Tip = hot (+)

Sleeve = shield



1/4" TRS (Tip-Ring-Sleeve)
balanced wiring:
Tip = hot (+)

Ring = cold (-)

Sleeve = shield

Line In (For Stereo Channels)

These inputs can accept 1/4" TRS balanced and 1/4" TS unbalanced plugs from any line-level instrument, effects device, or tape player.

When connecting a stereo device (two cords), use both the left (mono) input and the right input.

When connecting a mono device (just one cord), always use the left (mono) input and plug nothing into the right input. A trick called "jack normalling" causes the signal to appear on both sides.

The TRIM Control



The TRIM control adjusts the input sensitivity of the MIC or LINE IN jack to match the input signal to the operating range of the mixer. On all Mackie mixing consoles, the TRIM control is located at the top of the channel strip. That makes it easy to find since TRIM is the first control the signal encounters on its way through the console.

Whenever you connect a new input or the input source changes (you move the mic from the dulcimer to the kick drum), you should perform the **Level-Setting Procedure**. That procedure is "how to use the TRIM control."

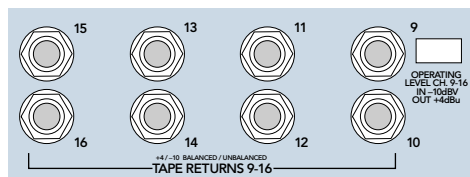
Connecting Electronic Instruments



Synthesizers, guitar processors, other electronic instruments, and computer sound cards work fine when connected to the LINE inputs. The pickups on electric guitars and basses, even though the plug fits, are a different animal, however. Pickups require a higher input impedance than the Mackie LINE IN provides. Plugging a guitar into a LINE IN jack will probably make some sound, but the guitar will lose punch and treble. The proper way to connect an instrument pickup directly to a Mackie mixer is to use a Direct Box (or DI = Direct Injection) between the guitar and a MIC input.

Tape Returns (8-Bus Only)

The TAPE RETURN jacks on the rear panel of the 8-Bus console are for connecting the outputs of a multitrack recorder.



These are balanced 1/4" TRS jacks which will also accommodate unbalanced connections. They're grouped in sets of eight, with each group having an OPERATING LEVEL switch to properly match the gain of these inputs to pro (+4 dBu) or semi-pro (-10 dBV) recorders.

TAPE RETURN jacks can be routed to either the channel (for mixing) or Mix-B (during tracking). Their fate is determined by the FLIP and MIX-B SOURCE switches.

Tape Inputs

These unbalanced RCA TAPE INPUT jacks are designed to work with semi-pro as well as pro recorders. They provide an unbalanced output at 0 dBu. Connect your 2-track tape recorder's outputs here, using standard hi-fi RCA cables.

In recording applications, these jacks make playback of your mixes through the Control Room monitors convenient. You'll be able to listen to a mix, then rewind and try another pass without repatching or disturbing the mixer levels. In sound reinforce-

ment applications, you can use these jacks to feed intermission music from a tape or CD player without tying up a mixer channel.

If your mixdown recorder has balanced XLR outputs at the "pro" operating level of +4 dBu, in addition to adapting the connector type, you'll need to attenuate the signal level by approximately 10 dB to avoid overloading these inputs. If the recorder has an output level control, just turn it down. Otherwise, you'll need to purchase or build an in-line attenuator. Alternate, you can connect the recorder outputs through an XLR-TRS cable to a channel line input.

Control and routing of the TAPE inputs varies among models. See the Control Room Source controls in the Master section for particulars.

Auxiliary Return Inputs

Auxiliary Returns are line level inputs designed for connecting the outputs of effects devices. They can also be used to connect extra line level audio sources. They're balanced 1/4" TRS jacks and will accept signals from just about any pro or semi-pro effects device on the market.

Typically Auxiliary Returns from effects are routed to the main L/R busses, but on certain models, they can also be easily routed to AUX sends or subgroup busses for more flexibility.

Talkback Microphone Input

Mackie SR series consoles are equipped with a connector for a dedicated *talkback* microphone. A mic plugged into this XLR jack can be routed to the main mix, allowing you to speak to the audience ("And now, let's give a warm welcome to *The SLOUGHTONES—YEAH!!*") or via AUX Sends 1-2 to stage monitors or headphones ("Hey Flash! Your D string is flat!").

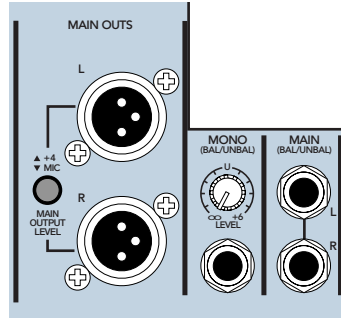
The talkback mic has its own volume control and routing switches located in the Master area of the console. The TALKBACK MIC jack does not provide phantom power, so you can't use a condenser mic for talkback.

The 8-Bus console is also equipped for talkback, but we've built the mic into the console. Just push the button and talk. In addition to communication with the play-

ers, the Talkback mic on the 8-Bus console can be sent to the bus outputs to *slate* the tape. (“Purple Tears, take seventeen”)

Don’t feel left out if your console isn’t equipped with dedicated talkback circuitry. You can accomplish the same thing by plugging a mic into a spare input channel and routing it to wherever you want to talk.

Main Outputs



When we speak of the *main outputs*, we’re talking about outputs from the left and right (stereo) busses. These outputs are controlled

by the MAIN MIX fader. This is where the fully mixed audio enters the real world.

The XLR MAIN OUTPUTS are differential, providing an extra 6 dB of output level, up to +28 dBu. Main outputs are duplicated on impedance balanced 1/4” TRS jacks which provide up to +22 dBu output level and work equally well when connected to either balanced or unbalanced inputs. When connected to a balanced input using a TRS plug, they appear to the input as a balanced output with all the benefits of a balanced connection. When connected to an unbalanced input, they appear as unbalanced outputs at the same output level as the balanced TRS outputs.

The PPM series mixers, since they’re intended primarily to use with their own amplifiers, have only 1/4” unbalanced jacks for the main mix outputs.

Main Output Level Switch

Some compact mixer models are equipped with a switch which attenuates the XLR MAIN OUTPUTS by 30 dB, bringing them down to approximately the level of a microphone. If you’re submixing a group of instruments on stage (say a bunch of keyboards), by switching the output to MIC level, you can send this submix back to the main mixer just as if it was another microphone. It’s also handy for plugging into the one mic input you get with the hotel meeting room.

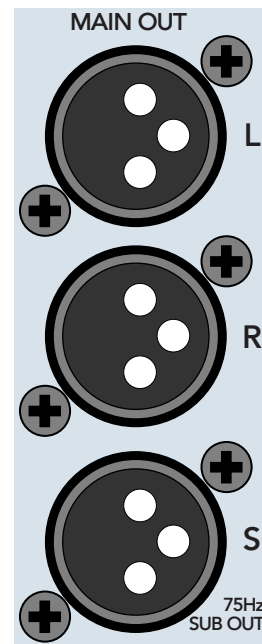
Mono Output

Several Mackie mixers have a dedicated Mono output for those requirements that seem to pop up now and then, demanding a monaural mix of painstakingly-created stereo panorama. The last thing you want to do is start twirling all your carefully-placed PAN settings to one side. What to do? Stick a cable in the MONO output, hand the other end to Mr. Mono, and you’re done. He’s got his mono mix and you’ve still got your stereo mix. The MONO output is nothing more than an equal mix of the left and right MAIN MIX outputs.

Mono Level Control

So, Mr. Mono comes running back, screaming about the level being so loud that his camcorder is melting. Just reach for this knob and turn it down a bit. Just the thing for sending mono signals to mic inputs like camcorders, telephone interface boxes, even answering machines.

Subwoofer (75 Hz) Output



The CFX series of mixers is equipped with a unique feature—a mono output followed by a 3rd order 75 Hz low-pass filter. It sums the left and right main outputs and removes everything but the deepest bass. Connect this output to a high powered subwoofer and the music police will be right over. The level of the subwoofer output is fixed relative to the

main outputs, so be sure that your subwoofer amplifier (or powered sub) has an adjustable input level for balancing the bass level.

Don’t confuse this subwoofer output with its similarly named neighbors, the SUB (subgroup) outputs.

Tape Outputs

These unbalanced RCA jacks tap the MAIN MIX outputs to make simultaneous record-

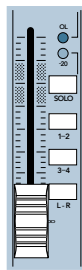
ing and sound reinforcement more convenient. They're also handy for interfacing consumer recorders or semi-pro computer sound cards.

Direct Outputs

These 1/4" jacks deliver the signal from nearly the tail end of the channel path; post-TRIM, post-EQ, post-LOW CUT, post-fader and post-MUTE. Not all models have direct outputs, some have direct outputs on only the lower numbered channels, and only the 8-Bus console has a DIRECT output on every channel.

Direct outputs are useful for sending the channel straight to a track of a multitrack recorder. A direct output can also be used to send a single channel's signal to an effect processor without using up an Auxiliary send.

Bus/Subgroup Outputs



In 4- and 8-Bus consoles, the output of the PAN control can be assigned to subgroup busses. The ASSIGN switches direct the channel output to pairs of busses. The subgroup buss outputs appear on the balanced 1/4" TRS SUB OUT jacks.

The subgroup outputs can be used to feed a mix of several channels (for example six mics on a drum kit) to a single pair of recorder tracks.

Bus Output Levels

The subgroup BUS output level on all of the Mackie mixers is nominally 0 dBu. The output level of the busses is controlled by the bus faders.

On the 8-Bus console. The bus output level is switch-selectable between +4 dBu and -10 dBV, allowing you to properly match pro or semi-pro recorders.

Double (and Triple) Bussing



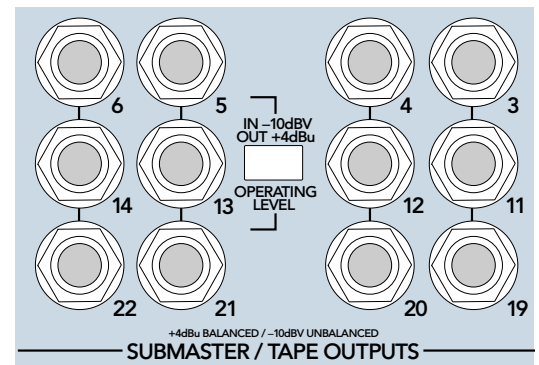
The bus outputs on the 1642VLZ-PRO, SR and 8-bus mixers have duplicate jacks to make connection to a multitrack recorder easier. Even though the SR and 1642 are four-bus mixers with four subgroups, they have eight

SUB OUT jacks for feeding eight recorder tracks, thanks to a trick called Double Bussing.

SUB OUTs 1 and 5 carry the same signal, as do 2 and 6, 3 and 7, and 4 and 8. By connecting those eight jacks to the eight inputs of your recorder, you can route any input channel to any track.

To record onto Track 1, assign the input channel to Bus 1 by pressing the 1-2 ASSIGN button and turning the PAN control fully left. On the recorder, put Track 1 into the Record mode, being sure that Track 5 is in the safe or play mode. To record on Track 5, use the same bus assignment on the mixer, but put Track 5 in Record. To record on Track 2 or 6, ASSIGN to 1-2 and pan fully right.

The 8-Bus console extends this trick by tripling the outputs of each of the eight busses to accommodate a 24-track recorder.



On most Mackie mixers, subgroup busses can be routed to the main outputs so that all channels assigned to that subgroup can be controlled by a single fader, retaining the balance and pan position set by the channel controls. You can individually mic each of the background singers, assign their channels to a single subgroup or pair, and use the subgroup fader(s) to bring the background vocals up in the mix when they're needed.

Not all models have the same SUB outputs, level controls, and routing capabilities. Input channels on the CFX series cannot be routed directly to the MAIN outputs, but must first be routed to a pair of subgroup busses, which are in turn assigned to the MAIN outputs by switches in the mixer's Subgroup section. The PPM series has no SUB outputs since it has no subgroups.

ALT 3-4 Outputs

The dual-purpose MUTE/ALT 3-4 switch on the 1202/1404VLZ-PRO mixers is a variation on the subgroup concept. When Greg Mackie was designing our first product, he wanted to include a mute switch for each channel. Mute switches do just what they say—turn off the signal by sending it into oblivion. “Gee, what a waste,” Greg reasoned. “Why not have the mute button route the signal somewhere else useful—like an ALTERNATE stereo bus?”

So MUTE/ALT 3-4 really serves two purposes—muting (handy during a mixdown session or live show), and signal routing. For multitrack recording and live mixing, it serves as an extra stereo bus.

One difference between the ALT 3-4 outputs and BUS OUTs 3 and 4 on the larger consoles is that there’s no output level control for those busses—they run at unity gain. Another difference is that it’s either/or routing—a channel can’t be assigned to both the MAIN and ALT outputs simultaneously. But like the true 4-bus consoles, the ALT busses can be routed to the MAIN outputs, serving as a submaster pair.

To MUTE channels in the main mix, leave the ALT 3-4 outputs disconnected. Pressing the MUTE/ALT 3-4 button switches the channel signal from the MAIN to the ALT 3-4 busses, sending it off to never-never land, disconnecting it from the MAIN MIX, and effectively muting its input.

To use ALT 3-4 as output busses, just connect the ALT 3-4 outputs to whatever destination you desire. Pressing the MUTE button will remove the channel signal from the MAIN bus and route it to the ALT 3-4 outputs.

The ALT 3-4 outputs are a good way to feed a multitrack recorder when you only need to record one or two tracks at a time or a good way to feed a stereo sound card in a computer.

PHONES Output

The stereo signal at this output jack is the main mix, but it is not affected by the position of the MAIN MIX faders.

The PHONES control knob allows you to set the levels in your headphones as desired, without disturbing the main mix level.

Note: Be very careful because the PHONES jack can drive any standard headphones to very loud levels. Please see the “Safety Instructions” on page 2 for information on hearing protection.

1/4" TRS stereo wiring:

Tip = left, ring = right, sleeve = shield

Walkperson-type phones can also be used with an appropriate adapter.

MAIN Mix Faders (DFXs)

These two faders control the final level of the signals sent to the MAIN outputs.

The TAPE and PHONE outputs also receive the main mix, but are not affected by these faders.

All active mono and stereo channels that are not turned fully down will appear in the main mix. Other signals feeding these faders include: AUX 1 RETURN, AUX 2 RETURN including the EMAC internal effects, and the CD/TAPE inputs.

The faders, set fully up, provide 10 dB of gain. A “U” unity gain point is just below that. When set fully down, the main mix is effectively muted. These are the faders to pull down at the end of the song when you want The Great Fade-Out.

AUX Send Outputs

The Auxiliary Send section is like a bunch of mono mixers, one for each Auxiliary output. The AUX controls send a copy of the channel signal to the AUX bus where the signals are mixed and sent to the AUX output. On some of our mixers, these controls are labeled MONITOR and EFX (effects) since those are common uses.

Pre and Post Sends

Auxiliary Sends are designated as PRE or POST. Pre or post what? POST always means post-fader, PRE is always pre-fader. The output level of a PRE send is independent of the position of the channel fader, while the level of a POST send follows the channel fader. Both come after the LOW CUT filter (if any) but on some models the PRE send is taken before the equalizer section while on others it comes after the EQ. It may come either before or after the MUTE switch. Here’s a summary for all of

the Mackie mixers of where the Pre-fader AUX sends are taken:

Table I: Source for PRE Sends		
Model	EQ	Mute
1202VLZ-PRO	Post	Pre
1402VLZ-PRO	Post	Pre
1604VLZ-PRO	Pre	Pre
1642VLZ-PRO	Pre	Pre
CFX	Post	Post
SR	Post	Post
8-Bus	Switched	Post
PPM	Pre	NA

That's quite an assortment. The variations reflect both the most likely application and what was in favor at the time the mixer was designed. We've included some modification instructions for moving the source of the sends to suit your taste. Here are some common preferences:

Stage monitors are almost always fed from a pre-fader send so that the engineer can create a mix for the musicians on stage and leave it alone once they're happy. Then he can mix the house according to his good taste. (Turn the drummer up really loud in his monitor and he'll think he's that loud in the house)

In larger sound reinforcement systems, pre-EQ sends for the monitors are nearly always preferred since there's usually an outboard equalizer between the console and the monitor amplifier. If the monitors are fed after the channel equalizer, an EQ boost to bring out an instrument in the house might cause a blast of feedback on stage. Also, musicians don't like to hear you messing with "their tone".

Studio headphone cue feeds are similar to stage monitors only there's no risk of feedback so fidelity can come first. If an instrument sounds better in the control room with EQ, it will probably sound better to the musician too, so post-EQ monitor sends are usually preferred. A pre-fader send is a good idea, though, since a musician playing an

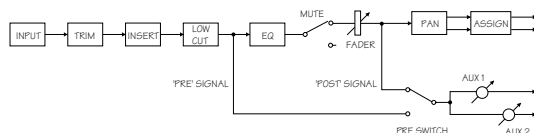
overdub may want to hear a mix that helps him focus on his part, while the producer will want to hear a well balanced mix in the control room.

Effects sends are almost always post-fader and post-EQ. Generally you want to send a signal to the effect processor in proportion to its level in the mix. When you turn something down in a mix, you want the effect level to drop along with it. Sometimes, though, you may want the "dry" signal to fade out, leaving only the effected version. Use a Pre-fader send to achieve this.

Having the pre-fader sends fed after the MUTE switch is very handy for live sound applications. When you mute a channel, it's muted everywhere.

PRE/POST Switches

Some AUX sends are hard wired one way or the other and labeled as such on the panel. Some models have switches which allow you to change the feed point of the Send from the front panel.

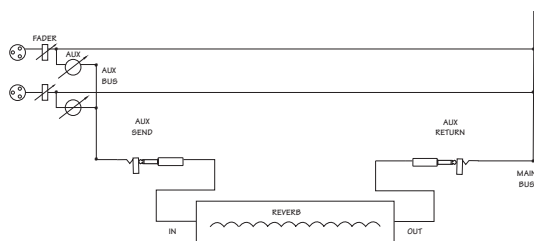


AUX Send Shift

Some Mackie consoles have six AUX SEND busses but only four AUX knobs. The SHIFT button selects whether their associated knobs send the channel signal to AUX outputs 3 and 4 or outputs 6 and 6 on a channel-by-channel basis.

Using AUXes with Effects

Here's the signal flow from the AUX send, through the AUX output, to a reverb unit, and back to the mix through the AUX Return jack.



This is sometimes called a “parallel” effect connection since the path of processed signal is in parallel with the dry signal path.

Built-in Effects

The CFX, DFX, and PPM series mixers have a built in digital effect processor (EMAC). The input of the EMAC is normally connected to the EFX bus (EFX 2 in the CFX), and the output is normally returned to the MAIN bus. The EFX send and return jacks on these mixers have normalizing switch contacts—inserting a plug disconnects the normal send/return loop through the EMAC so that you can use them with an outboard processor.

With some clever cable juggling you can even use the EMAC processor as an outboard effect for another mixer.

Effects: Serial or Parallel?



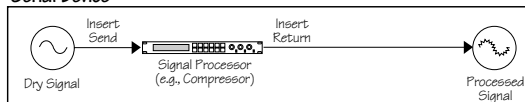
Effects devices are used either in serial or in parallel:

Serial means that the entire signal is routed through the effects device. Examples include: preamps, compressor/limiters,

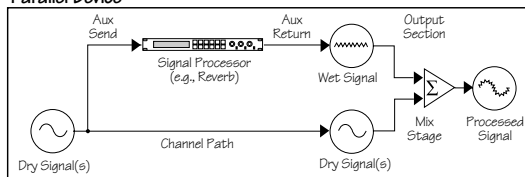
graphic equalizers. Connections are typically made via the channel INSERT jacks.

Parallel means that a portion of the signal is tapped off to the effects device, processed, and returned, to be mixed with the original “dry” signals. Multiple signals (via multiple mixer channels) can all make use of the same parallel effects device. Examples include: reverb, delay, chorus. Connections are typically made via AUX sends and AUX returns.

Serial Device



Parallel Device



EFX Foot Switch and Bypass

We've included some features on the PPM, DFX, and CFX mixers that will be welcomed by the musician or band working without a sound engineer. The Grand Canyon reverb might have sounded great on that ballad you just finished, but it really sounds dumb on your speaking voice when you're introducing the next song. It's handy to be able to quickly kill the effect processor and then turn it back on when it's time for the next song.

The EFX BYPASS switch on the front panel of the PPM and CFX mixers mutes the signal to the effect processor when you're in easy reach of the mixer. By plugging a *normally open, latching (push-on-push-off)* foot switch into the EFX FOOT SWITCH jack of the CFX, DFX, and PPM mixers, you can mute the effect return without leaving your performing position. If the mixer is in sight, the lighted EFX BYPASS LED will remind you of why you're no longer in the Grand Canyon.



If you're using an outboard effect processor on the PPM instead of the EMAC, that will also be bypassed when either the front panel or foot switch is activated. On the CFX and DFX, however, the bypass switch comes ahead of the EFX RETURN 2 jacks, so an external processor coming in to the mix at

that point will not be affected by the bypass foot switch.



Plugging in a foot switch disables the front panel EFX BYPASS switch on the PPF and CFX mixers.

Control Room Output

When recording, it's convenient to have an output with its own volume control that's dedicated to control room listening. Control Room outputs are on 1/4" balanced TRS jacks (unbalanced on the 8-Bus) and are fed from the Control Room Level control. This allows you to crank the playback level so the drummer can hear it, or turn it down to answer the telephone, all without disturbing the MASTER level which you've set to send the proper level to the mixdown recorder.

The source for the Control Room output is selected by switches in the Master section of the console. The Main left/right outputs are always available to the control room monitor, other choices, depending on the mixer, are the subgroups, ALT 3-4 bus, playback from a stereo recorder, Mix B, or another external source such as a CD player.

Headphone Output

The Headphone output is another copy of the Control Room output (the 8-Bus offers several other choices) followed by a low-powered amplifier capable of driving a set of headphones to a comfortable, no, a *pretty darn loud* level.



WARNING: When we say the headphone amp is loud, we're not kidding. It can cause permanent hearing damage. Even intermediate levels may be painfully loud with some earphones. BE CAREFUL!

On some models, the headphones and control room share the same level control, on others, there's a separate control for headphone level.

Some models have two headphone jacks. The two jacks on the 1642VLZ-PRO are fed from the same point and both follow the same Phones level control. The 8-Bus console has two completely independent headphone outputs, each having its own level

control and source selector for two separate mixes to its headphone jacks.

Studio Output (8-Bus)

The Studio Output normally feeds an amplifier and speakers (or powered speakers) in the studio so that the musicians can hear a playback without coming into the control room. These carry the same signal as the Control Room outputs but have an independent level control so that you can adjust the playback volume to suit the listeners.

The 8-Bus TALKBACK microphone can be routed to the Studio output so that you can talk to the players through the studio monitor speakers.



WARNING: When you have mics live in the studio, they'll also be live in the studio monitors if the Studio output control is turned up. In the extreme case, you'll get feedback, but you don't want to record the speaker outputs either. When tracking, be sure to turn the Studio Level control down between playbacks.

The Studio Output jacks can be used with a second amplifier and speakers in the control room so you can compare the mix on your high grade monitors with the boom-box speakers the rest of the world listens to. To switch between the two, turn one level control down and the other one up.

Mix B Output (8-Bus)

The MIX-B jacks are 1/4" unbalanced outputs which carry the Mix B signal. We'll cover Mix B in depth later on, but briefly, it's an independent stereo mix that's usually used for monitoring the multitrack recorder returns when tracking.

Channel Inserts

The channel INSERT jacks allow you to break the channel signal path right after the mic preamp stage and its associated TRIM control (the low-cut filter too on some mixers) and *insert* a device such as a compressor, equalizer, or de-esser into the chain. You can also insert a signal processor such as a delay, flanger, or reverb unit in the channel path to process the source coming into that channel. This is sometimes called a "serial"

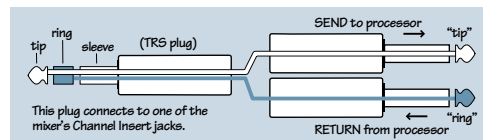
effect hookup since it's in series with the signal path.

Since most people who need only a small mixer don't have racks full of processing gadgets, our smallest mixers don't have Inserts on every channel. The larger ones do.

The INSERT jack is a 1/4" TRS type, but in this instance the two jack contacts are used as an output (*send*) from the preamp and an input (*return*) back into the remainder of the channel strip. Switch contacts built into the jack provide a "normal" connection from send to return when there's no plug inserted. When you push a plug all the way in, this connection is broken and you have access to the channel insert points through the jack.

The tip of the plug connects to the SEND and the ring connects to the RETURN. Since we're using only one contact of the jack for each connection, connections to and from the INSERT are *unbalanced*. The SEND is a low impedance line level output which will drive any pro or semi-pro processing device. The RETURN is a moderate impedance line level input that can be driven by almost any device.

You can buy an *Insert Cable* off the rack at your local music store, or build one. Here's how it's wired:

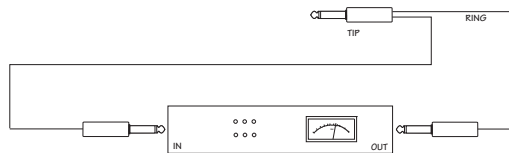


Tip = send (output to effects device)

Ring = return (input from effects device)

Sleeve = common ground (connect shield to all three sleeves)

Here's how to hook an outboard signal processor in line with the channel path (a "serial" connection):



Insert Jack Tricks—Direct Outputs and Inputs

Besides being used for adding outboard signal processors to the channel signal path,

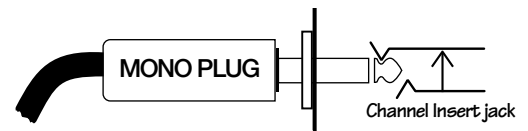
the INSERT jacks can be used as pre-EQ *direct outputs*.

It's a fact—Mackie mic preamps have become so famous that people buy our mixers just to have the preamps in their arsenal. To use a preamp by itself, with nothing else in the signal path to the recorder, use the INSERT SEND to connect it directly to the recorder's input.

If you insert a plug into the jack gently, you'll feel two "clicks", the first as the plug tip passes the *ring* contact and the second when it settles into the *tip* contact. If you use a "mono" (TS) plug and insert it only as far as the first click, the ring contact will connect to the plug, sending the preamp out to whatever's on the other end of the cable.

But waitacottinpickinminnit! The ring is the *return*, isn't it?

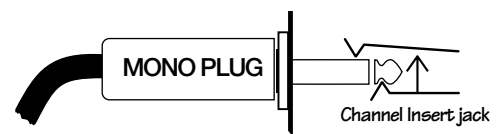
Yes, but with the plug inserted only half way, the normalling switch contacts in the jack don't open—send and return are still connected. Since the normal connection is still intact, the signal passes through the channel path as well as coming out the INSERT jack as a direct output. If you're using the console for sound reinforcement and recording each input separately to a multitrack recorder, this is the way to feed the recorder inputs.



Direct out with no signal interruption to master.
Insert only to first "click."

Here's what an Insert looks like inside the mixer when used as a Direct Output:

NOTE: If you want to keep the preamp signal out of the mix (you will if you're monitoring the returns from the recorder), push the plug *all the way in*. That will open the normal contacts.

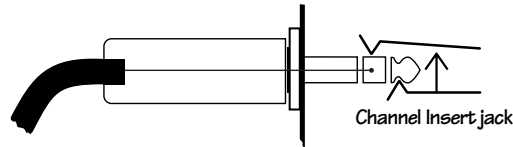


Direct out with signal interruption to master.
Insert all the way in to the second "click."

If you want to use an external mic preamp with your mixer (boo-hoo!), or perhaps a Voice Processor (combination preamp, compressor, equalizer, de-esser and auto-

matic word rememberer) you can completely bypass the fabulous world famous Mackie mic preamp by bringing the signal from your outboard preamp into the mixer through the INSERT Return.

To do this, you'll need a custom cable with a stereo (TRS) plug wired with the hot lead connected to the *RING*, the shield connected to the sleeve, and nothing connected to the tip.



Plugging this all the way into the jack will break the normalling contacts, replacing the Mackie preamp with the output from the outboard unit.



With this connection, you're bypassing the TRIM control, so you'll need to use the Output Level control on your external preamp when using the **Level-Setting Procedure**. It's still important to check the channel level even when using an outboard unit as the front end.

Utility Output (CFX Series)

The stereo signal from the UTILITY OUT 1/4" TRS jacks is the same as the MAIN output, but with its own level control, allowing you to set the level to an auxiliary set of speakers or a recorder without disturbing the main mix level.

These outputs are handy for driving "fill" speakers in areas of a room that aren't adequately covered by the main speakers. Or you might use them to feed speakers in the lobby, or the Green Room. (Regardless of the color, every concert venue has a Green Room—it's where the musicians and hang-ers-on hang out)

UTILITY outputs are taken after the main fader, so adjustment of the MAIN level control *will* affect the UTILITY output level.

PPM Series Outputs

The PPM mixers have two power amplifiers built in. A flexible arrangement of switches and jacks allow you to use those two ampli-

fiers for any combination of main and monitor speakers.

SPEAKER Out

These 1/4" TS jacks carry the outputs of the power amplifiers.

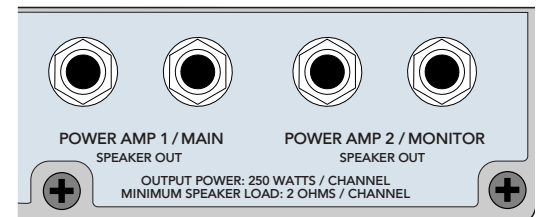


There's real power behind these jacks—125 to 600 watts depending on the model, so don't connect them to anything other than passive speakers—not active speakers, not a recorder, not another mixer. **SPEAKERS ONLY!**

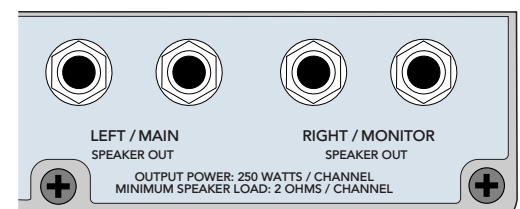


Caution: *Don't use standard guitar cables to connect your speakers. Guitar cords are designed for line level signals and low currents, not the high currents and voltages of speaker levels. Use heavy duty speaker cables, at least 16 gage. The heavier (lower gage number) the better. Thin speaker cables equal thin sound!*

There are two SPEAKER OUT jacks for each amplifier. The jacks are wired in parallel.



Mono Version Speaker Connectors (408M)



Stereo Version Speaker Connectors (408S)

- Two 8 ohm speakers in parallel equals 4 ohms.
- Two 4 ohm speakers in parallel equals 2 ohms.

The general formula for impedance in parallel is:

$$Z \equiv \frac{1}{\frac{1}{Z_1} + \frac{1}{Z_2} + \frac{1}{Z_3}} \text{ etc}$$

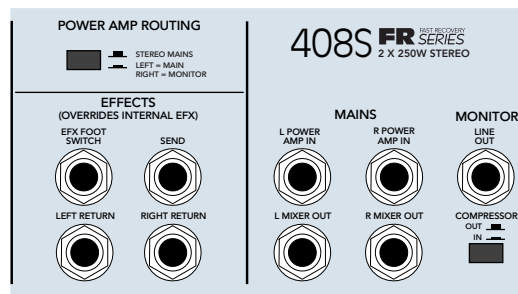
In electronics, Z is the symbol for imped-

ance. We don't have a clue as to what word Z stands for. Tell us and win a T-shirt.

Can you plug speakers with different impedances into the two jacks same amplifier? Sure, as long as the total impedance isn't lower than 2 ohms. Chances are, however, that the efficiency of the speakers will be different enough so that they'll produce noticeably different volumes. This may or may not be what you want.

Note that the stereo and mono PPM mixers have different labels on the speaker connectors. They also have different options for routing MAIN and MONITOR mixes to those two amplifiers.

There's a convenient push button to select the most common configurations. There are also patching jacks which allow you to rearrange things to your liking or add an external power amplifier for more flexibility. See the PPM Applications for some suggestions



On the stereo mixers, the choices for Power Amp routing are:

POWER AMP ROUTING—In

AMP 1—MAIN Mono (left + right)

AMP 2—Monitor

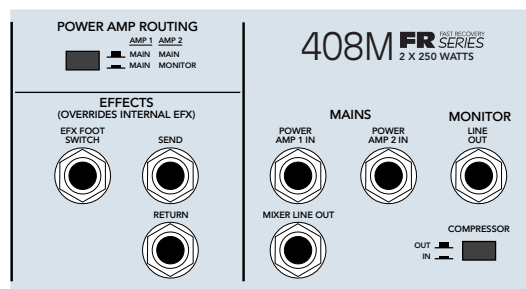
Or

POWER AMP ROUTING—Out

AMP 1—MAIN Left Channel

AMP 2—MAIN Right Channel

On the mono mixers,



the choices are:

POWER AMP ROUTING—In

AMP 1—MAIN

AMP 2—Monitor

Or

POWER AMP ROUTING—Out

AMP 1—MAIN

AMP 2—MAIN

For both the mono and stereo models, when the switch is in the *Out* position, both power amplifiers are dedicated to the MAINs. If you want to connect a stage monitor, you'll either need to use a powered speaker or an external power amplifier connected to the MONITOR LINE OUT jack.

The speaker outputs are affected by the MAIN MASTER level control and the MAIN Graphic Equalizer.

MONITOR Line Out

This is a 1/4" unbalanced jack with a line level signal suitable for driving powered speakers or power amplifiers. The signal is fed pre-fader and pre-EQ from the mixer's monitor bus. The MONITOR output has its own low-cut (rumble) filter, MONITOR MASTER volume, control and MONITOR graphic equalizer.

MIXER LINE OUT (Mono)

L or R MIXER OUT (Stereo)

These are 1/4" unbalanced line level jacks that provide the main mixer output(s). The jack is fed from the MAIN mixer bus (left and right busses in the stereo version), following the MAIN MASTER control and MAIN graphic equalizer, and just ahead of the POWER AMP IN jack(s).

You can connect the MIXER OUT to an external power amplifier for more power and more speakers. Plugging into these jacks does not interrupt the signal flow to the PPM's built-in power amplifiers.

POWER AMP IN

This is a 1/4" jack that accepts an unbalanced line-level signal. Whatever signal you plug into these jacks is combined with the internal mixer signals and routed to the power amplifiers. You could connect the LINE OUT jacks from another mixer (even another PPM) to the POWER AMP IN jacks to increase the number of channels available.

You could connect the outputs from the opening band's mixing board to these inputs. This way you don't have to set up two sets of speakers—the other band can use their mixer together with your PPM, using its amplifiers and your speakers.

On mixers later than Serial Number *****, these jacks are wired in the “normalled through” configuration allowing you to use an *Insert Cable* for patching in an outboard graphic equalizer, limiter, or crossover network. On earlier mixers, a modification is available to convert the “input only” jack to a normalled through configuration. Contact Mackie Tech Support for details if you wish to modify your own mixer.

Powering Up! Power Supplies and Accessory Power

Compact Mixers

Of course you have to be able to turn your mixer on—but you don't want to turn it off inadvertently, either. To keep such accidents from happening, we hide the POWER switch on the rear panel of all of the Mackie compact mixers.

At Mackie, “wall wart” is a four letter word. All of our compact mixers have internal power supplies. A standard IEC connector and power cord connects the mixer to an ac outlet. You can plug the ac linecord into any ac outlet with a voltage between 100 and 240 Vac, without having to worry about setting any switches or using a step-up or step-down voltage transformer. The power cord is an industry-standard item, so if you leave it behind, you won't have to look too far for a replacement. Most computers use the same power cord, and you can buy one at most computer or office supply stores as well as your local Radio Shack or music store.

All Mackie mixers will operate equally well from 50 or 60 Hz power, but they're designed for only a single ac line voltage, depending on the country where you bought it. If you travel world-wide, you'll need a step-up or step-down transformer in countries that have a different line voltage than what's specified for your mixer.

There's a power-on indicator on the front panel. Check your Quick Start guide for its location if it's not obvious.

Power Switch

Push the side or top of the switch (labeled “ON”) to turn the mixer on; you should see the POWER LED glow in confirmation on the top right of the mixer. To turn the mixer off, push the switch the other way.

As a general rule, turn the mixer on first, before any amplifiers or powered speakers, and at the end of a show, turn it off last. This will prevent any turn-on or turn-off thumps being heard in your speakers.

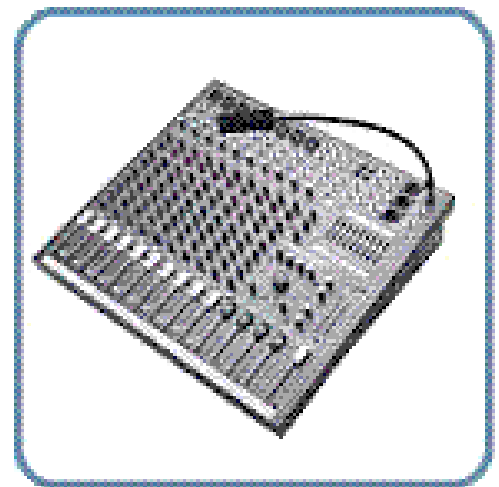
8-Bus Console

The 8-Bus console has an outboard power supply with the main power switch on its front panel. The switch is also the power-on indicator.

Connect the power supply to the mixer using the special multiconductor cable provided, then connect the ac power to the power supply. An optional 15 foot extender cable, part number 040-118-00 is available from the Mackie parts department.

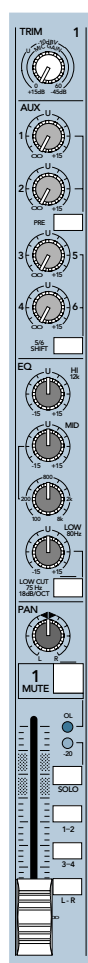
BNC Lamp Socket

Those decidedly non-audio-looking BNC connectors on the top panel of several Mackie mixers are for mounting and powering gooseneck lamps so you can find the knobs in the dark. The BNCs are wired with the center post at +12 Vdc.



We recommend LittleLite lamps 12" gooseneck #12G or #12G-HI (a high-intensity version) for the smaller consoles and the 18G-HI with an 18" gooseneck for the larger mixers. Buy them from your pro audio dealer or order them on line from www.mackiegear.com.

Chapter 3: The Channel Strip



This is where most of the work is done, so let's take a stroll down the strip. We'll use the 1604VLZ-Pr as an example since it's typical of most mixers, and along the way we'll point out differences you'll find with other models.

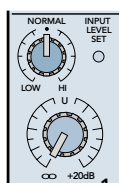
TRIM

This control sets the sensitivity of the channel input stage to match the output level of the source connected to the channel mic or line input. A hot microphone on a kick drum will require a lower TRIM setting than a ribbon mic six feet from a classical guitar. The **Level-Setting Procedure** will guide you to the correct setting of the TRIM control.

Unless you change the input source or you've dialed in large EQ boosts on the channel, once you've set the TRIM properly, you can usually leave it alone throughout the session.

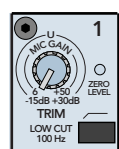
The TRIM control is located at the top of the channel strip and is identical in function and appearance on all the Mackie mixers with exception of the PPM series.

PPM Input Level



On the PPM, the INPUT LEVEL control serves the same function as the TRIM. It's located just above the VOLUME control.

Zero Level LED (CFX Only)



The CFX mixers have a Zero Level LED next to the TRIM control. It indicates the channel signal level after the preamp and low-cut filter, but before the equalizer. This handy indicator glows when it receives a signal at or above 0 dBu. It's not as accurate for setting channel gain as using the **Level-Setting Procedure**, but it will clue you that an input is in danger of clipping on peaks when the band starts crankin' on stage.

If the LED is on solid as opposed to flickering, turn the TRIM down. If it never flickers, you can safely turn the TRIM up.

"U" Like Unity Gain

Mackie mixers have a "U" symbol on almost every level control. U stands for "unity gain", meaning no change in signal level when the signal goes through the control or the stage it controls. Once you have performed the **Level-Setting Procedure**, even though TRIM may not end up set to unity gain, you can set every control at "U" and your signals will travel through the mixer at a safe, distortion-free level.

What's more, all Mackie level controls are labeled in decibels (dB), so you'll know what you're doing level-wise when you change a control setting.

Of course many of your settings will deviate from the Unity position—that's what mixing is all about—but you'll find that, for a balanced sounding mix, they won't be too far off Unity. The unity gain point is a good benchmark to assure that you're not getting too close to overload or too far down into the noise floor of your system.

Channel Fader

The fader is almost the last control in a channel's signal path, but it's probably the one you'll use the most. Electrically, it comes after the EQ section and before the PAN control. It's located at the bottom of the channel strip where it's handy.

On the PPM series, the VOLUME knob is the equivalent of the Fader. On the 1202VLZ-PRO it's the knob marked GAIN

The Unity Gain mark (U) on the fader, known in engineering circles as "design center", is located about three-quarters of the way up. On most Mackie mixers, pushing the fader all the way up provides an additional 10 dB of gain, should you need to boost a section of a song. If you find that the overall level is too quiet or too loud with a fader near Unity, that's an indication that the channel input level or your EQ settings have changed substantially since you initially set the TRIM. Get yourself back into the normal operating range by performing the **Level-Setting Procedure** again.

MUTE Switch

The MUTE switch (on all models except the PPM series) is nestled close to the PAN control. Engaging it turns the channel signal off in the Main outputs. It also removes the signal from the subgroup busses, except in the case of the 1202 and 1402VLZ-PRO models where it doubles as the subgroup assignment switch (see ALT 3-4).

Depending on the model, the MUTE switch may or may not mute the signal to the AUX busses. See the Auxiliary Sends section for a who's where.

The SOLO PFL signal is tapped off the channel ahead of the MUTE switch, so you can audition a muted channel in the headphones before bringing it up in the mix.

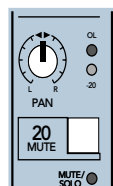
STATUS LEDs—Mute, Solo, Overload, and Signal Present

These LED indicators located in the vicinity of the PAN and MUTE controls (their arrangement and presence varies among models) give you some information about what's going on with the signals.

MUTE

This indicator lights when the channel is muted.

OL (Overload)



This LED monitors the signal level just ahead of the channel fader (after the mic preamp and EQ section) and lights when the peak signal exceeds the maximum level that the channel can pass without distortion. If you see this light blinking, it's time to back off on the TRIM control. (that ol' **Level Setting Procedure** again)

-20 (Signal Present)

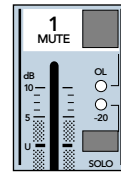
This LED monitors the signal at the output of the mic preamp and illuminates when the signal level at that point is at least -20 dBu. It gives you a clue that there's actually something coming into the channel. By watching the blinking of the LED, you might even be able to guess what it is. For instance a kick drum will cause the LED to

pulse in time with the music (hopefully!) whereas a synth pad will cause it to light more evenly.

SOLO

This LED illuminates when the channel SOLO button is pressed. It works in conjunction with the RUDE SOLO light in the Master section to call your attention to channels that are in SOLO. It's a good clue as to why you're not hearing what you think you should be hearing in the headphones.

Shared LEDs



On some mixers, we've pressed a couple of the indicators into serving double duty. This saves space, but requires some explanation.

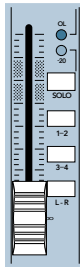
Notice the lines connecting the OL LED and MUTE button, and the -20 LED and SOLO button. When the channel isn't muted, the OL indicator flashes to indicate a dangerously high channel signal level. When it lights brightly and continuously, it indicates that the channel is Muted. With a little practice you can visually tell the difference between a guitar amp turned up to 11 and a muted channel.

Similarly, the -20 (signal present) LED blinks when the SOLO button is pressed. The SOLO indication is brighter than the signal present indication and blinks at a steady rate so you can tell one indication from the other. If the -20 LED is on steady and the RUDE SOLO light isn't on, it's indicating a signal present.

Bus ASSIGN

Mackie makes 8-bus, 4-bus, 2-bus (stereo) and mono mixers. The bus ASSIGN switches, working in conjunction with the channel's PAN control, determine to which bus the channel signal will go. This is one place where the channel strips on the different models diverge. We'll take a look at each of the variations.

4-Bus Consoles



Alongside each 1604-VLZ PRO channel fader you'll find four buttons, labeled SOLO, 1-2, 3-4 and L-R. The latter three are collectively referred to as channel assignment switches. 1, 3 and L are the left half of these bus pairs, and 2, 4 and R are the right half. Used in conjunction with the channel's PAN knob, these switches determine the destination of a channel's signal:

The 1604-VLZ PRO, 1642-VLZ PRO, and SR series are what we call "true 4-bus" mixers. Each channel can be assigned or unassigned to any of the subgroups without affecting the other subgroups or settings within the channel, and each subgroup has its own master fader and dedicated output. In fact, since there are 4 subgroups *and* the MAIN L-R MIX, it's actually a 6-bus mixer. We could have named it the 1606-VLZ. Darn!

With the PAN knob set to the center position, the left and right or odd- and even-numbered busses receive equal signal levels. To feed only one side or the other, just turn the PAN knob accordingly.

If you're doing a 2-track mixdown, simply engage the L-R switch on each channel that you want to include in the mix, and those channels will be sent to the MAIN L/R outputs. If you want to create a subgroup of certain channels, engage either the 1-2 or 3-4 switches *instead of the L-R*, and they'll be sent to the appropriate subgroup faders. From there, the subgroups can be sent back to the MAIN MIX, allowing you to use the subgroup faders as a master control for those grouped channels.

If your recorder's inputs are connected to the mixer's bus 1-4 outputs, when recording new tracks or bouncing existing ones, you'll also use the 1-2 and 3-4 switches, but without assigning the submasters to the L-R bus. You may find it useful for monitoring and rehearsal, to assign channels with "live" inputs to L-R as well as a subgroup so you can hear what's going on in the studio.

However, if you're printing tracks via the DIRECT OUT jacks, all the channel assignment switches should be disengaged (up) other than those channels that you're using to monitor your recording. See the Recording Applications section for further details.

Bus assignment isn't mutually exclusive. You can assign a channel to both L-R and 1-2, feeding the PA system from the MAIN L-R busses and a radio broadcast or stereo recorder from busses 1-2. That gives you independent control over the two mix levels, and lets you take certain channels out of the radio mix (for example, the obnoxious MC). When sending your mixer's outputs to multiple destinations, using the bus outputs gives you a safety net, too. Since the busses are electrically isolated from one another, a shorted cable to the broadcast feed won't affect your PA mix—the show will go on.

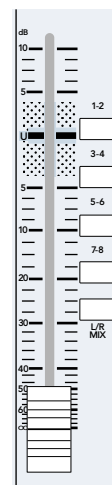
CFX Series 4-Bus



The ASSIGN setup on the CFX series is a bit different. There are four subgroup busses and the MAIN left and right busses, but channels cannot be assigned directly to the L-R busses as they can in the 1604 family, they have to go to a subgroup first.

For stereo mixing applications, typically ASSIGN 1-2 will be engaged on all channels destined for the main mix. By assigning SUB 1 (Left) and 2 (Right) to feed the main mix using the switches next to the SUB faders in the Master section, the channel ASSIGN 1-2 switches effectively become the L-R ASSIGN switches.

Assigning a group of channels, for example, all the drum mics, to 3-4 allows you to create a drum submix. By then assigning subs 3-4 to the main mix, you can adjust the level of the drums separate from the rest of the band.



If you're doing a mono PA mix (typical for small clubs), you don't need to assign subgroups in pairs. By pressing the ASSIGN 3-4 button on all the background vocal channels and turning their PAN controls all the way to the left, you've assigned those vocals to Subgroup 3 and you can control their level (or turn them off completely when they're not singing) by using the SUB 3 fader. Press the ASSIGN 3-4 button on the drum channels, pan fully right, and you have a sub-

mix of drums on BUS 4 to bring into your mix. For mono, you can have up to four subs. If you're doing a stereo mix, you can

have two stereo subs, or one stereo and two mono subs.

8-Bus Assign

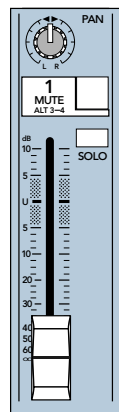
The 8-Bus ASSIGN switches work just the same as those on the 1604 family of consoles only there are more switches since there are eight busses rather than four. Each channel can be assigned directly to the MAIN L-R bus or any of four pairs of subgroup busses.

The 8-Bus has three output jacks for each of the eight subgroup busses so they can be used conveniently to feed up to 24 recorder tracks. (see Double Bussing)

PPM Bus Assignments

This is too easy. There are no bus assignment switches on the PPM series. Since these mixers are either stereo or mono, all channels are permanently assigned to the MAIN busses. On the stereo PPM models, the PAN control assigns the channel signal to Left, Right, or anywhere in between. There's no PAN on the mono versions. Isn't that simple?

The ALT 3-4 Bus (1202/1402VLZ-PRO)



The 1202VLZ-PRO and 1402VLZ-PRO (and the original CR1604 too) are also 4-bus mixers, but there are *only* four busses, not six. Channels can be assigned to either the MAIN mix (L-R) or to the ALT 3-4 busses.

The single bus assignment switch is labeled MUTE ALT 3-4. It doubles as a channel mute switch if you're not using the ALT 3-4 bus in the mix.

Here are two examples of how the ALT 3-4 bus can be used:

Recorder Feed from ALT 3-4

When doing multitrack recording, use the ALT 3-4 outputs to feed your multitrack. Using Y-cables, you can mult the ALT 3-4 outputs so that one output feeds multiple tracks. Connect ALT OUT LEFT to tracks 1, 3, 5 and 7, and ALT OUT RIGHT to tracks 2, 4, 6 and 8. Now, tracks that are in Record or Input modes will hear the ALT 3-

4 signals, and tracks in Playback or Safe modes will ignore them.

ALT 3-4 as a Subgroup—the Assign to Main Mix Switch

When doing live sound or mixdown, it's often handy to have subgroups, and with the ALT 3-4 bus, you can have them. Simply assign channels you want to subgroup, for instance background vocals, to the ALT 3-4 bus. Now, go over to the Master section CR SOURCE control group and press the ALT 3-4 and ASSIGN TO MAIN MIX switches. In this configuration, the CTRL ROOM/SUBMIX fader becomes the sub-master for the channels assigned to ALT 3-4 and that fader will control the level of the subgrouped channels in the MAIN mix.



Note that the meters read the *input* selected to go to the main mix. In this configuration, they no longer read the MAIN MIX level. As you raise the CTRL ROOM/SUBMIX fader, the subgrouped channels will come up in the MAIN MIX, though the meter reading will not change. Also, you won't be able to monitor the main mix in the headphones.

Another way to accomplish the same thing is to assign the channels to the ALT 3-4 mix, then patch out of the ALT OUTs back into an unused stereo channel.



If you do that, don't ever engage the MUTE/ALT 3-4 switch on that stereo channel, or you'll have every dog in the neighborhood howling along with your feedback loop.

ALT 3-4 as AFL (1202VLZ-PRO)

The 1202VLZ-PRO isn't equipped with a PFL/AFL SOLO MODE switch, but you can use the ALT 3-4 as an AFL (After Fader Listen). Just select ALT 3-4 as the Control Room SOURCE (Master section). When you engage a channel's MUTE/ALT 3-4 switch you'll get that channel, all by itself, in the CONTROL ROOM and PHONES.

MUTE/ALT 3-4 is one of those clever concepts that can bewilder a newcomer, so take your time and play around with the controls. Once you've got it down, you'll probably think of a hundred uses for it!

PAN

(Note from the author: You have no idea how difficult it's been up to this point to be formal and scientific and write "pan control" when all my life, I, and every engineer I know has called it a "pan pot". That's going to change RIGHT HERE! Ahhh, I feel better now.)

The PAN pot splits the channel signal into two branches and adjusts the amount of the channel signal sent to one branch or the other. Those two branches go to the L-R main mix or *odd-even* subgroup bus pairs. Turning the PAN pot changes the *phantom image*, the apparent position where the sound appears between the stereo speakers.

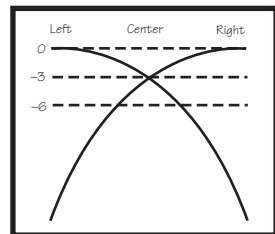
With the PAN knob turned fully left, the channel signal feeds only the Left MAIN or odd-numbered subgroup busses, depending on which ASSIGN switches are pressed. Turning the PAN knob fully right feeds the channel signal to only the Right MAIN or even-numbered assigned subgroup busses. In the center, the signal is sent to both Left and Right MAIN busses (or odd and even subgroups) at equal levels.

If you have a mono PPM mixer, don't look too hard trying to find the PAN control. There's no need for one since the channel is always assigned to the single MAIN bus.

Constant Power!!!



Mackie mixers employ a design called Constant Power panning. As you move the phantom image by turning the PAN knob from left to right, the volume of the panned sound will remain constant throughout the move.



In order to accomplish this, we build the PAN pots so that the signal level drops by 3 dB from its full left or right level when the knob reaches its center position. When the two -3 dB signals combine, they sum back to the original signal level.

It follows from this that when you pan a source from the center to full left or right, the left or right level will be 3 dB higher than it was in the center.

PAN on Stereo Inputs

Some Mackie mixers are equipped with stereo input channels that can accept either one or two inputs. When a stereo channel is used with a single (mono) input, the PAN pot works as described above.

With a stereo source (two inputs), the PAN pot on a stereo channel works just like the *Balance* control on your stereo receiver. When it's centered, the Left bus gets the full Left input signal and the Right bus gets the full Right input signal. As you rotate the PAN knob to the left, the signal going to the Right bus drops in level. At full left rotation, the Right signal is completely off and only the Left input is fed to the Left bus. Vice versa when you turn PAN fully right.

Most of the time when using stereo inputs, L-R panning is established at the source—a CD, a multi-timbral synth, sampler, or drum machine. You'll probably leave stereo channel PAN pots centered most of the time other than for small tweaks.

EQUALIZATION (EQ or eee-QUE)

Usually we expect our electronics to have flat frequency response—all frequencies amplified equally. Sometimes, though, we want to tailor the frequency response of a channel to compensate for acoustics, alter the response of a microphone, modify the sound of an instrument or voice, or to help balance frequencies to get a track to "sit" better in the mix. The Tips section contains a more detailed discussion of the use of equalization, but since we're talking about channel strip controls, we'll introduce the knobs and concepts here.

Mackie mixers vary in the number of EQ controls and their functions. Consult your quick start guide for particulars on your mixer. We'll cover them all in general here.

EQ Parameters

You may have heard the term *parametric* equalization. This refers to the parameters that affect frequency response. Those are:

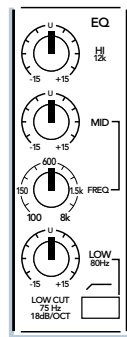
- Amplitude—the amount of boost or cut at or around a specific frequency
- Frequency—the frequency that's being boosted or cut

- **Bandwidth**—the width of the bump or dip in the frequency response curve resulting from an EQ boost or cut. Some or all of those parameters are adjustable (others are fixed), depending on the model of your mixer.

EQ Bands

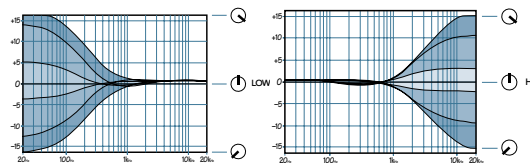
The audio spectrum covers the range of 20 Hz to 20 kHz. We've divided that into three or four equalizer frequency ranges or *bands* so that we can have a manageable number of controls. The number of bands varies depending on the model and whether it's a mono or stereo channel.

Here's a set of controls for a four-band equalizer. Each of the four controls cuts or boosts at its indicated frequency.



Of course the frequency that you're boosting or cutting with the EQ controls matters. We've chosen these fixed EQ bands carefully so their frequency range and bandwidth will be useful for solving problems and enhancing your mix musically.

The high and low frequency EQ bands have a *shelving* response. As you move below the nominal frequency of a low frequency EQ or above the nominal frequency of a high frequency EQ, the frequency response doesn't just keep rising (with a boost) or dropping (with a cut), it levels off, sort of like a shelf.



On all Mackie mixers, the high and low frequency EQ bands are fixed at 80 Hz and 12 kHz, with shelving response and boost/cut range of ± 15 dB.

The low frequency EQ band emphasizes the punch of a bass drum or bass guitar, fattens synth patches, or some puts a boom into serious bass male singers. The high frequency EQ band adds sizzle to cymbals and can add an overall sense of transparency or edge to keyboards, vocals, guitar, and bacon frying. Turn it down to reduce sibilance or to hide tape hiss.

The EQ frequencies are nominal values, not a sharp cutoff points. Notice that on both the high and low bands, there's some boost or cut action as far away from the nominal frequencies as 1 kHz. This gentle slope up to the EQ frequency is part of what makes the EQ sound musical.

Stereo Graphic EQ (DFXs only)

This equalizer is used to shape the frequency spectrum of the main mix or monitor mix. It is the last thing in the signal chain before the MAIN MIX faders and the MAIN OUT jacks.

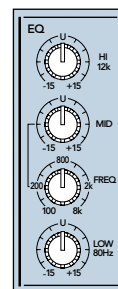
Each EQ slider controls a different section of the audio frequency range. Moving a slider up or down will boost or cut the output levels of signals around the slider's center frequency. By setting all the EQ sliders to zero (center) you'll effectively remove it from the signal path.

Mid-band EQ

The frequency response curve of the mid-band EQ has a *peak/dip* shape. The 1202 and 1402VLZ-PRO and PPM series have a single MID EQ band centered at 2.5 kHz. You'll find that this frequency adds presence to a voice and improves the definition of most instruments.

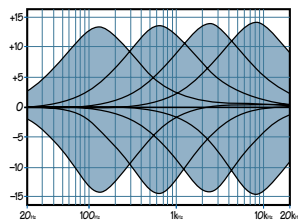
Stereo channels of the 1642VLZ-PRO, SR, and CFX mixers have two MID EQ bands, one centered around 800 Hz (400 Hz on the SR series) and another around 3 kHz. 400 Hz is a good frequency to cut when working with a 'nasal' vocalist, 800 Hz adds some body to a voice, and 3 kHz is a good 'vocal presence' frequency.

Sweep EQ



On some models, the frequency at which boosting or cutting takes place can be adjusted. This is what's called "Sweep EQ", since you can sweep through a range of frequencies and fine tune the equalizer to work exactly where you need it. The mid-band section of the sweep

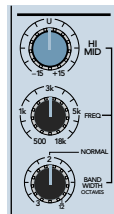
EQ shown here has two controls, one adjusts the frequency and the other adjusts the amount of boost or cut at that frequency.



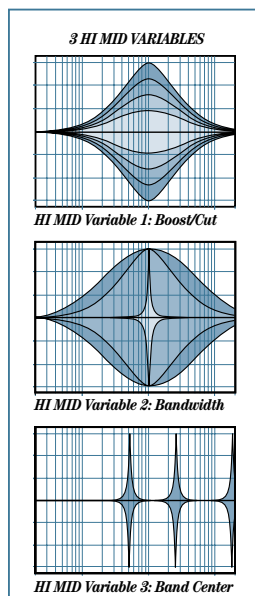
The mid-band EQ on the 1604VLZ-PRO, and mono channels of the 1642VLZ-PRO, SR, and CFX series mixers is

sweepable over the range of 100 Hz to 8 kHz. Here's what the MID EQ curves look like at several frequencies. Note the peak shape of the curve.

Bandwidth

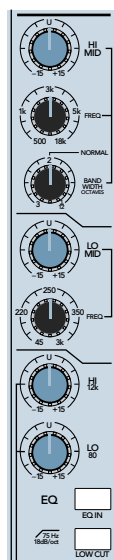


The 8-Bus console has two sweepable MID EQ bands, one covering the range of 45 Hz to 3 kHz, the other covering 500 Hz to 18 kHz.



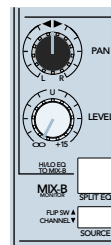
The HI-MID EQ section has a third control marked in octaves which adjusts the width of the peak. The top graph shows the boost/cut action, the center graph shows wide (3 octaves) and narrow (1/12 octave) bandwidths. The bottom graph shows the narrow band setting at three different frequencies.

EQ Split (8-Bus Only)



Note the arrangement of the EQ controls on the 8-Bus. Unlike the other mixers which arrange the EQ frequencies high to low from top to bottom, the HF and LF EQ controls on the 8-Bus are located below the MID EQ group.

The HF and LF shelving EQ bands can be removed from their normal position in the channel path and switched into the MIX-B path to provide EQ on this totally separate mix.



The SPLIT EQ button that does this neat trick is located in the MIX-B section of the channel strip, just below the MIX B LEVEL and PAN pots.

This is a very cool feature. Typically, when you're tracking a multitrack project you record flat (without EQ) since you're not sure just how things will fit together spectrum-wise once you get to the mixdown phase. Sometimes, though, it's nice to have a little EQ available on the monitor mix that you're listening to while you're tracking.

If you monitor using MIX-B (we'll get into that later), by pressing the SPLIT EQ button, you can twiddle the HF and LF EQ controls without affecting the multitrack recording. Be sure to keep notes in case you find some settings that work well. You might want to use them when you do the final mix.

EQ IN/Bypass (8-Bus only)

When the IN button is pressed, the EQ is in the channel path. Release the button and the EQ section is completely bypassed. If you're not EQ-ing the recorded tracks, bypassing the EQ will give you a cleaner signal path.

If you've switched the HF and LF shelving EQ to the MIX B path, the Bypass switch only bypasses the section that remains in the channel path, so you can still use the HF and LF controls on MIX B.

MAIN/AUX 1 EQ Switch (DFXs only)

This switch allows you to add EQ either to your main mix, or to your monitors.

In the UP (MAIN) position, the EQ only affects the main mix, phone or tape outs.

In the DOWN (AUX 1) position, the EQ only affects the AUX SEND 1/MON output; with no effect on the main mix, phone or tape outs.

Bypass EQ Switch (DFXs only)

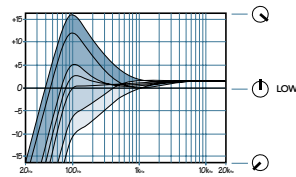
Use this switch to quickly compare the effect of your EQ settings.

In the UP position, the EQ section is in line with the signal path, (either main mix or monitor, depending on the position of the MAIN/AUX 1 switch).

In the DOWN position, the EQ section is bypassed and has no effect on any outputs.

Low Cut Filter

All Mackie mixers with the exception of the PPM series are equipped with a sharp low-cut filter on mic input channels. This filter attenuates frequencies below 75 Hz (100 Hz on the CFX) and it has a steeper slope than the LF EQ. Once it starts cutting, it cuts fast. It's designed to stop rumble dead in its tracks. It will help tame plosives on vocal mics, and may help to reduce wind noise and traffic rumble when you're working outdoors.



Since the LOW CUT filter is close to the working frequency of the LF equalizer, the two interact. This graph shows what

happens to the low frequency response as you adjust the LF EQ with the LOW CUT filter engaged. Combining a LF EQ boost with the LOW CUT filter creates a peak around the “whomp” frequency of a kick drum. Be careful when using a lot of LF boost though. Powerful lows can sap power from amplifiers, and you can blow a speaker before you can say “Oops! Wrong button!”

Other EQs

The CFX, DFX, PPM, and SR mixers have some EQ built into the Master sections. The CFX, DFX, and PPM mixers have Graphic equalizers and the SR has a high frequency EQ which we call “Air”. Since we're still working our way through the channel input section of the mixer, we'll cover those later, but we didn't want you to think we forgot.

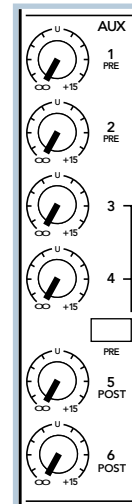
AUXiliary Sends

Here's a Channel Strip view of the AUX controls (disguised as EFX and MON controls on the PPM, DFX, and CFX mixers). If you skipped over the explanation of PRE and POST, go back to it if you get lost here.

The AUX Send controls are arranged pretty much the same on all Mackie mixers. Models vary in the number of AUX busses, so they'll have a different number of controls and a different mix of pre- and post-fader

choices. AUX Sends adjust the volume of the channel signal sent to the AUX bus and ultimately appearing at the AUX OUT.

Let's look at the AUX section of the SR series:



On the SR mixers, AUX 1 and 2 are always Pre-fader, AUX 5 and 6 are always Post-fader, and AUX 3 and 4 give you a choice. The PRE button switches AUX 3 and 4 between pre- and post fader.

In a sound reinforcement application where you might need several monitor mixes, you'd probably choose to set AUX 3 and 4 to PRE. For mixing a recording where you'll be using several effects processors fed from the AUX busses, Post (the PRE button disengaged) would be the usual preferred choice.

AUX Shifts

The 1604VLZ-PRO and 8-Bus have six Aux busses, but only four knobs.



The top two knobs always control the signal sent to the AUX 1 and AUX 2 busses, either pre- or post-fader as you've selected, but the other two buttons are shared among four busses. With the SHIFT button in the *up* position, they control the signal sent to the AUX 3 and AUX 4 busses. Press the SHIFT button *down* and those same two knobs now send the channel signal to AUX busses 5 and 6. Twice the sends in half the space!

AUX SOURCE (8-Bus Only)

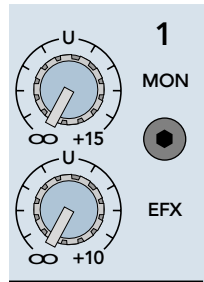


The 8-Bus console has a SOURCE button associated with the lower two AUX controls. This switch liberates AUX 3-6 from the main channel strip world that AUX Sends 1 and 2 are stuck in, allowing them to tap into a different source—the channel's MIX-B signal, either pre- or post-fader. This can be a very handy thing.

If you're using MIX-B for monitoring the recorder returns during tracking, these AUXs can be used for additional head-

phone cue feeds or to put reverb in the monitor without sending it to a track. If you're using the MIX-B section as additional inputs during mixdown (from MIDI synths, for example), pressing the SOURCE button turns AUX 3-6 into effects sends for those inputs.

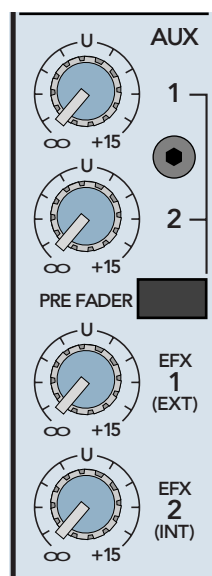
EFX and MON Sends—AUXs in Disguise



Since we've anticipated what you'll use AUX sends for most of the time on the CFX, DFX, and PPM mixers, someone got the bright idea to label them accordingly.

Since monitor mixes are usually pre-fader, on our PPM series, you'll find a MON control. This is a pre-fader Auxiliary send.

The EFX knob is a post-fader Auxiliary send that's normally connected internally to the built-in EMAC effect processor. Turning up the EFX knob sends the channel signal to the EMAC processor or to an outboard effect processor that you've plugged into the EFX SEND jack,



The CFX series has a split personality. Equally at home in the studio or sound reinforcement world, it has both AUX and EFX labels on the controls that send the channel signal to the Auxiliary busses. AUX 1 and 2 are selectable (as a pair) to be pre- or post-fader sends. EFX 1 and 2 are post-fader sends which feed the EFX 1 and 2 output jacks.

EFX 2 (INT) is also normalled to the internal EMAC processor. Unless you've interrupted this connection by plugging something into the EFX 2 output jack, the EFX 2 knob controls the signal going into the EMAC. If you want to use an outboard processor as well as the EMAC, connect your processor to the EFX 1 output.

On the DFX series mixers, AUX 1/MON AUX 1 is designed for feeding a separate

monitor mix to the musicians on stage, via the AUX 1 SEND output jack.

The signal is tapped off before the channel EQ section, and before the channel fader. Only the channel GAIN, MUTE and LOW CUT will affect the output to your monitors.

AUX 2 is designed for feeding the EMAC internal effects processor and the inputs of parallel effects devices.

It is post-fader, so any changes to the channel controls will affect the AUX 2 signal. In addition to feeding the inputs of the EMAC, it also feeds the AUX 2 SEND output.

Using Stereo Effects



While we're talking about effects and sends, here's a word about outboard stereo processors. RTFM!

(Read The *Furnished* Manual). It's pretty common for effect processors such as reverb and chorus to have stereo outputs. Bring those into the mix for a nice spacious stereo effect. But what about their inputs?

Many effect units, particularly older ones, have separate Left and Right inputs, but they're mixed to mono before they hit the circuitry that produces the effect. If your processor unit works like this, there's no need to waste two AUX sends to feed it—one will do nicely, and you'll still get the stereo effect by bringing its Left and Right outputs back into the mix in stereo.

There's a good reason, though, why they have two inputs. Nearly every processor has a control to mix some of the input (dry) signal with the processed (wet) signal so that the output isn't 100% wet. This is an important feature if you're using the processor "in line", patching it through INSERT jacks or between units in a signal chain. Having two inputs preserves the Left/Right relationship of the dry signal that gets mixed with the wet signal before it reaches the processor's output jacks.

If you're feeding an outboard effect unit from an AUX output and bringing its output back into the mix through an Aux Return or a channel line input, be sure that the signal you're returning to the console is *100% wet*—no point in bringing in another copy of a signal that's already in the mix.

Also, unless it's a true stereo effect device with a separate processing engine for each

channel, it will sound the same whether you feed it one signal or two, so Save Your Sends.

Unique 8-Bus Channel Strip Switches

The 8-Bus console has a couple of switches on the channel strip that aren't found on other consoles. Here's what they do:

MIC/LINE Input Selector

The MIC/LINE switch is located all the way up top, just below the Mic jack. It selects whether the Mic jack or the LINE IN jack is the input to the channel strip.

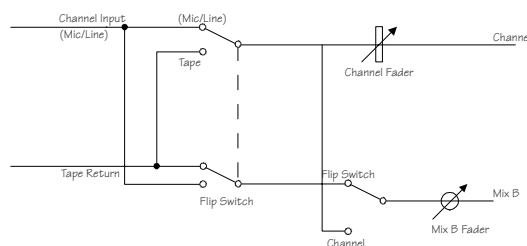
On other Mackie mixers, the Line and Mic inputs are both active all the time, and we assume that you'll have one or the other source connected to the channel, but not both. In the recording studio, however, you might use that channel with a mic when recording and with a synth when mixing. With this switch, you can have them both connected all the time.

FLIP Switch—MIC/LINE-TAPE



This button is located just below the TRIM control. It interchanges the input sources to the channel fader and the MIX-B level control (coming right up!).

As its label indicates, when the FLIP button is in the *up* position, the MIC/LINE input (after the preamp stage) goes to the channel fader and the TAPE RETURN input goes to MIX B. This is the normal mode for tracking and overdubbing, with the main faders controlling the signal going to the recorder and the Mix B rotary pots controlling your monitor mix.

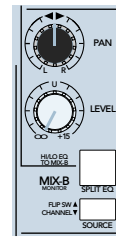


In the *down* position, the TAPE RETURN signal is fed to the channel fader. This is the normal routing for mixdown so you can control the mix of the recorded tracks using the main faders.

To recap—When FLIP is *up*, whatever is going into the MIC or LINE input (as selected by the MIC/LINE switch) feeds the channel, and the TAPE RETURNS feed the MIX-B section.

When FLIP is *down*, the channel is fed from the TAPE RETURNS and MIX-B is fed from the MIC/LINE input. The two sets of inputs are flipped. Get it? Now, about that MIX-B:

MIX-B/MONITOR



Each channel strip has a dual signal path with extremely flexible switching. This allows either the MIC/LINE or TAPE RETURN inputs to be routed through either the channel fader path or a separate stereo mixer with separate EQ and monitoring. That's the mysterious Mix-B!

This handy set of controls is how *in-line monitoring* (a standard feature in modern large multitrack recording consoles) is implemented on the Mackie 8-Bus console. We're not claiming that it's anything new, we've just added some extra features for more flexibility. MIX-B/Monitor routing options can get a bit complicated, so pay attention.

MIX-B is like another stereo mixer, independent of the eight-plus-two recording buses we've talked about so far. There are three input sources available to MIX-B: MIC/LINE, TAPE Return (as determined by the FLIP switch) and the channel strip pre-fader output (via the SOURCE switch). This can become a post-fader channel output (via the source switch) by modifying the channels. See the Modifications section for details on this surgery.

MIX-B SOURCE Switch

When the Mix-B SOURCE switch is up, MIX-B receives its input from the FLIP switch. Remember, the FLIP switch exchanges the routing of the MIC/LINE and TAPE RETURN inputs between the Channel and MIX-B.

With the SOURCE switch *up* (to select the FLIP switch) **and** the FLIP switch is *up* (*unflipped*), the MIX-B section gets its input from the TAPE RETURNS and functions as a recorder monitor mixer. This allows you to hear a mix of the outputs of your

multitrack recorder as you record and overdub. By using the Auto-Monitor switching feature common to most multitrack recorders, you'll also hear the signals being fed to the recorder. This is the most common use of the MIX-B section, during tracking and overdubbing.

With the MIX-B SOURCE switch *up* (to select the FLIP switch), and FLIP *down* (the normal destinations of the TAPE and MIC/LINE inputs flipped), LINE INPUTS (or MIC if it's selected) become inputs to the MIX-B mixer, where they can become "virtual tracks" from MIDI synths or additional effect returns during a mixdown. Simply plug the additional signal into the MIC or LINE jack for the channel and use the MIX-B level and pan controls to place it in the stereo mix.

Although they are normally independent mixes, a button in the Output section (MIX-B TO L/R MIX—coming up later) can add the output of the MIX-B buses to the L/R Mix buses. Voila! Double your mix inputs!

With the MIX-B SOURCE switch *down* (in the CHANNEL position), MIX-B taps its signal from the channel strip, just before the channel fader. MIX-B is separately panable, EQ-able and can be used as an alternate stereo mix, a stereo auxiliary send, a "mix-minus" mixer, a quadraphonic or surround feed, you name it. Mix B can also borrow AUX sends 3-6 to enhance the MIX-B Mix.

Chapter 4: Master Section

This is the portion of the mixer where everything comes together. Here's where you'll find the master level controls for the main and all the auxiliary mixes, auxiliary return controls, monitor source selection and level controls, meters, and other features that vary from model to model. While channel strips look pretty much alike on all of our mixers, there are a lot of differences in their Master sections.

We'll start with the controls that are common to all mixers and then branch out to the unique features to be found on the different models.

MAIN L/R MIX Fader

This is the dual fader or rotary knob that adjusts the level of the MAIN LEFT and RIGHT outputs together. The fader is fed from the MAIN L/R busses, the fader output is amplified, and ends up at the MAIN output jacks. If the mixer is equipped with MAIN INSERT jacks, the fader follows them in the chain.

The Unity Gain point on the Master is approximately 3/4 of the way up on a slide fader and 12 o'clock on a rotary knob. Fully up gives 10 dB of boost available above Unity gain, and fully down shuts the outputs completely off. This is the fader to pull down gracefully when you want to do that Great Fadeout.

One of the differences among models is whether or not the signal to the RCA TAPE output jacks is controlled by the MAIN fader. On the CFX and PPM series, the TAPE OUT jacks are ahead of the MAIN fader (so you won't hear your fades on tape), but on all the other models, the MAIN fader also controls the TAPE outputs.

Meters

On all Mackie mixers, unless you've pressed a SOLO button, the meters read the level of the LEFT and RIGHT OUTPUTs. On mixers which have a set of CONTROL ROOM SOURCE switches, the meters indicate the level of whatever source is selected (one choice, naturally, is MAIN MIX).

If no CR source is selected and no channel is in SOLO, the meters will just sit there—that's not such a bad thing though since it will give you a clue as to why you're not hearing anything in the headphones or control room speakers.

Metering and Common Sense

There are two very important points on the meter scale. The top LED (Clip level), and the 0 LED near the middle of the scale.



The CLIP LED illuminates when the main output level reaches +22 or +28 dBu, depending on the model and the output jack. ***This is the end of the line.*** The mixer cannot put out any more voltage than this, but it will distort while trying to do so. It's guaranteed.



Bear in mind, too, that there's a pretty good chance that whatever's connected to those outputs will be breathing hard at this level. If your amplifier clips at +20 dBu, your sound system will distort before the meters hit +22. Good level management involves knowing the capabilities of every part of the system and setting levels and gains so that nothing clips before the last link in the chain. It does no good to feed a distorted signal to a nice clean amplifier. See Gain Structure in the Tips section for more information and some operating hints that will keep you out of trouble.



The other important point on the meter scale is the 0 LED. It's the target for setting TRIMs using the **Level Setting Procedure** and it also indicates the nominal output level of the mixer.

When any SOLO switch is pressed, the meter is automatically switched to the Solo bus so that it displays the signal level going into the channel fader (or out of the fader if the SOLO MODE switch is in the AFL position). This includes any boost added by the equalizer, which can be significant.



The SR, 1402, 1604, and 1642VLZ-PRO models have two solo modes, PFL and AFL, selectable with the SOLO MODE switch. More about these later, but just a reminder here that you *must* be in the PFL mode in order to use the meters for setting the TRIMs accurately.

8-Bus Meters

The 8-Bus console has a pair of MAIN meters just like the other models, displaying the MAIN and SOLO levels, but there's an additional set of eight meters. Located directly above the Bus faders, these indicate the output level of each of the eight subgroups. The calibration of these meters follows the setting of the output level switches (-10 dBV or +4 dBu). 0 on these meters represents the selected output level.

The reference level for 0 VU on the MAIN/SOLO meters on the 8-Bus console varies depending on whether you're SOLOing a channel or monitoring the outputs. When in SOLO, 0 on the meters represents 0.775 volts (0 dBu), the nominal internal operating level of the console. If you have just one channel assigned to the L/R bus with the faders set so the MAIN meters are reading 0, then you solo that channel, the meters will jump up by 4 dB. Don't panic, it's supposed to work that way. When not in SOLO, the MAIN meters reference 0 VU to +4 dBu just like the submaster bus meters.

The +22 dBu LEDs at the top of the meters indicate clipping. You should keep levels adjusted so that they never come on.

An accessory meter bridge is available for the 8-Bus console which allows you to meter either all the tape return inputs or the post-fader channel outputs (the same point as the DIRECT OUTPUT).



When a Mackie compact mixer's meter (all models but the 8-Bus) reads 0 VU, the output level at the 1/4" main output jacks is 0 dBu (0.775 V) and +6 dBu at the XLR jacks. For this reason, your mixer's meter and the meter on a recorder connected to the MAIN outputs may not agree. Your recorder probably has a nominal input level of -10 dBV or +4 dBu, which is the voltage level required to make *its* meters read 0. There's nothing wrong with either unit. You have to understand what the meters are telling you. Trust them, but trust your ears too. Don't become a meter slave and try to fix something that sounds OK but just doesn't look right on the meters. Things can get even more confounding when you watch analog and digital meters together. There's more info on metering in the Tips section.

Subgroup Faders

As you might expect, these faders control the levels of the signals sent to the SUB Outputs. When a subgroup fader is turned up, any channel assigned to that subgroup that is not muted and doesn't have its fader pulled completely down will appear at the SUB OUT controlled by that fader.

The SR and 8-Bus series consoles are equipped with Pre-Fader INSERT jacks on the submaster path, allowing you to insert an effect processor, compressor, or equalizer in line with the subgroup and control the processed signal with the subgroup fader.

On consoles with "double bussing", that subgroup fader controls the level of all outputs for its associated bus.

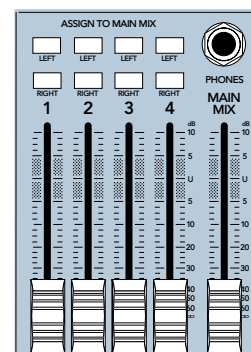
Assigning Subgroups to the Main Mix

One popular use of the subgroup faders is as masters for a group of channels on their way to the MAIN MIX. Let's say you've got a horn section hogging seven channels and you're going to want to fade the horns at a different point or rate than the rest of the band. You don't want to try that with seven hands or seven fingers, so you've unassigned all the horn channels from L-R and assigned them to Subgroups 1-2, panning them the way you like them.

Fear not. Even though you assign subgroups to the MAIN mix, they're still available as separate outputs should you want to use them for recording.

There are different approaches to getting SUB 1 and SUB 2 into the MAIN L-R mix, depending on the model mixer you have.

ASSIGN to MAIN MIX



Here's the subgroup section from a 1604VLZ-PRO. The 1642 and CFX series are similar. The LEFT and RIGHT buttons above the subgroup faders assign the fader output to the MAIN Left and MAIN

Right outputs respectively.



If you're not clear on the concept of subgroup busses and how to use the PAN pots to assign channel signals to busses, review the Bus ASSIGN section of this book. The PAN pots and BUS ASSIGN switches are keys to understanding this important concept.

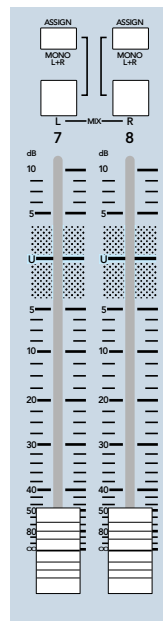
If you've set up a stereo submix (as in the above horn section example), assigning SUB 1 to the LEFT MAIN bus and SUB 2 to the RIGHT will preserve the left-right positioning that you've established with the channel PAN pots.

Pressing both the LEFT and RIGHT buttons on the same subgroup assigns the subgroup to both the Left and Right MAIN outputs equally, putting that subgroup mix in the center of the stereo field. If you're doing a mono mix (a small club PA for example) you could simply assign everything in a submix to a single subgroup and save a fader. For example, assign all the background vocals to Sub 3 by pressing the 3-4 ASSIGN buttons on each background vocal channel and panning all of those channels to the left.

Pressing the LEFT and RIGHT Assign to MAIN MIX buttons above the SUB 3 fader will put the background vocals into both channels of the mix equally and allow you to control the overall background level with the SUB 3 fader.

If you assign a subgroup to both LEFT and RIGHT as in this example, a few dB of attenuation is automatically added to keep the apparent loudness constant, just like our PAN pots do.

8-Bus Subgroup Assign– MONO L+R



The 8-Bus console has a variation on this theme. In addition to the L(ef) and R(ight) MIX assignment buttons above the Subgroup faders, there's one labeled ASSIGN MONO L+R.

The MONO L+R switch works in conjunction with the L and R assignment buttons and functions ONLY when the L or R button below it is also pressed. In the illustration above, pressing L MIX and L+R above it assigns the SUB 7 signal equally to the L and R MAIN busses, putting its signal in the center.

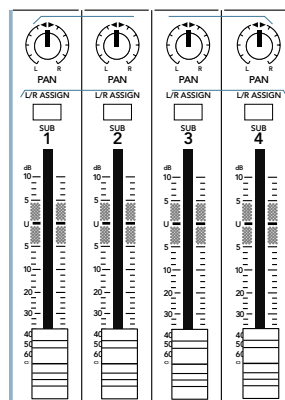


Notice, though that odd numbered busses have only a L MIX assign button, while even numbered busses have only a R MIX button. If you're using stereo subgroups, they work, like the ASSIGN buttons in the channel strip, in pairs, odd numbered subgroups of a pair is assignable to the LEFT and even numbered subgroups assignable to the RIGHT.

For more flexibility in panning of mono submixes on the 8-Bus console, patch a subgroup output into a vacant channel, assign the channel to L-R, and use the channel's PAN pot to fling that drum kit around the room.

Subgroup PAN and L/R ASSIGN (SR Series Only)

The SR series consoles provide still more flexibility in routing the subgroup outputs by replacing the LEFT and RIGHT buttons with a single ASSIGN button and an associated PAN pot.



Pressing the L/R ASSIGN switch above a subgroup fader assigns the fader to the PAN pot above it. The outputs of the PAN pot go to the L-R MAIN mix, allowing you to

pan a whole subgrouped section to the desired position in the mix.

If you have several channels of keyboards and you want them all to appear, for example, a bit left of center in the stereo mix. Assign all the keyboards to SUB 1 (ASSIGN 1-2, PAN fully left), then engage the L/R ASSIGN switch for Subgroup 1 and set its PAN pot to about 10 o'clock.

If you have a subgroup that you want to spread across the stereo field in your mix—for example a large choir or string section, assign the inputs to a pair of subgroups, pan accordingly, and then assign that pair of subgroups to the MAIN outputs, with the SUB pan pots on the odd numbered SUB turned fully left and on the even numbered group turned fully right. If the stereo field is too wide, you can narrow it by turning the pan pots on the SUBs toward the center.



There's no law that says a stereo mix must be exactly centered between the left and right channels. If you want to have the string section in stereo but left of center, set the SUB pan pots for the string channels at, say 7 o'clock and 11 o'clock.

Subgroup SOLO

The 8-Bus and SR consoles are equipped with SOLO buttons for the subgroups to allow you to hear the subgroup signal in the headphones and control room outputs as well as check its level on the meters.

On the SR series, the Subgroup SOLO is normally pre-fader. If you want to solo a subgroup post-fader, you must have the SOLO MODE button pressed (IN-PLACE AFL) *and* have the subgroup's L/R ASSIGN switch engaged. This will allow you audition the subgroup as it appears in the mix.

On the 8-Bus console, the SOLO button above the subgroup fader sends the *post-fader* subgroup signal to the solo bus. If you have not pressed a subgroup's MONO L+R switch, SOLO sends the odd-numbered subgroups to the left speaker and the even numbered subgroups to the right, whether or not the L or R buttons have been pressed. But if you've pressed MONO L+R, the subgroup SOLO will appear in both speakers, right smack in the middle.

SOLO Lights and Knobs

As you've probably surmised by now, SOLO can be a very useful function. It's handy for checking the presence and quality of individual inputs (and outputs on some models) when setting up, recording, and mixing.



Here's a reminder that you can press as many SOLO buttons as you want. All the selected channels will be mixed on the Solo bus and come out the headphone and control room outputs. So, you can have a duet or a trio as well as a solo. This is handy if you have a "section" playing and want to check their balance.



SOLO is intended for more than just "soloing". It's THE way to set levels for best noise and headroom performance. You're probably tired of reading it by now, but here's just one more reminder about the

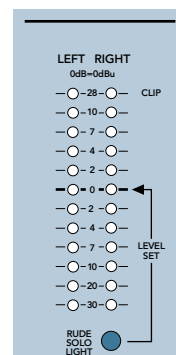
Level-Setting Procedure.

There are a few extra goodies in the Solo section of some of the Mackie consoles which affect how SOLO functions, so stand by.

RUDE SOLO LED

This is a Mackie trademark. Every one of our mixers (with exception of the PPM series, which has no Solo functions at all) has a RUDE SOLO light. It's the biggest, brightest, reddest light on the mixer and it flashes when *ANY* SOLO button is pressed. If you can't hear anything because you've accidentally pressed a SOLO button on a channel, this light will rudely remind you to look over the buttons and lights to find the problem.

LEVEL SET LED



To quote Step 4 of the **Level-Setting Procedure**: . . . "Some of our mixers have a SOLO MODE switch in the output section. If applicable, select the LEVEL SET (PFL) mode—The LEVEL SET LED will light."

The LEVEL SET LED comes on when the SOLO MODE switch is in the PFL/Level Set position *and* a SOLO button is pressed. The

arrow connecting the RUDE SOLO, Level Set, and 0 VU LED (your target for level setting) will remind you of the association. The arrow tying the Level Set LED to the 0 VU LED is there to remind you of the Level-Setting Procedure target level.

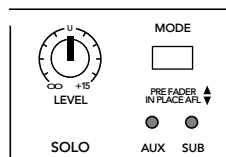
You must have the PFL/LEVEL SET mode selected in order to accurately read and set the channel signal level. If you try to set the TRIM level in AFL Mode, the meter will follow the fader, and that's not the level you want to set. So, what's this PFL/AFL business? Read on.

SOLO MODE (PFL/AFL)

Pre-Fader-Listen (PFL) and After-Fader-Listen (AFL) are old British studio terms that came over on the Mayflower and stuck around. When we (in the US) say "Solo", we usually mean PFL—hearing the signal before it hits the fader and pan pot. To others, "Solo" means AFL, not only after the fader, but after the pan pot, too. This allows you to hear both the relative level and position in the stereo field. Because AFL comes after the pan pot, it's often called "solo-in-place" to distinguish it from "just-plain-old" solo.

For setup, troubleshooting, and cueing in a live mix (psssst . . . don't forget TRIM setting, too), PFL is the way to go. You can hear the signal when nobody else can. You can solo a channel with its fader down and see what's going to come up in the mix when you push up that fader.

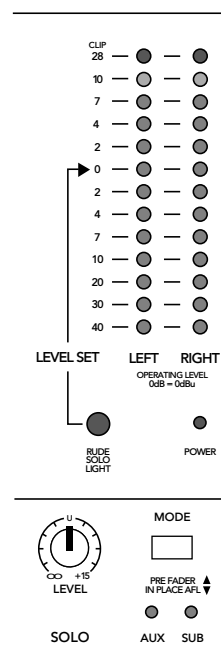
For mixing, it's sometimes helpful to hear the signal as it appears in the mix—that's where you might use AFL.



The SOLO MODE switch in the Master section of some Mackie mixers selects whether the SOLO buttons function as PFL or

AFL. On mixers without this switch (1202 and CFX), solos are pre-fader. On the 8-Bus, they're post-fader, but there's a modification to change them to post-fader. (See the Modifications section in the Appendix).

Which mode is normal? That's a function of the mixer's primary application and what flavor coffee Greg had on the morning he picked the labels for the switches.



On the SR series mixers, which are slanted toward sound reinforcement applications, the normal (button up) mode is PFL, and pressing the Solo MODE button switches to AFL. On the other models, the normal (button up) mode is AFL, and you must press the button for PFL SOLO and Level Setting. On some mixers, we even remind you by labeling the PFL Solo Mode LEVEL SET.

SOLO LEVEL

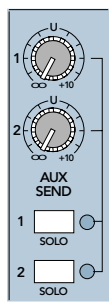
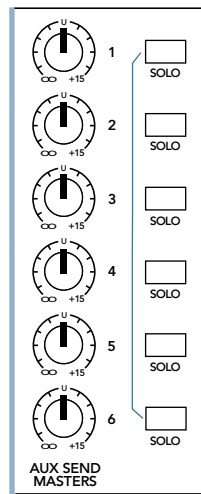
It's handy to have a separate level control for the SOLO signal so you can hear the soloed channel at a comfortable level without changing the overall monitor level. Most Mackie mixers have a SOLO Level control located near the Solo MODE switch. This adjusts the level of the solo signal fed to the phones and control room outputs, but it doesn't change the level going to the meter. You wouldn't want that anyway since you want to read the actual signal level.

SUB and AUX SOLO LEDs

The SR series consoles have a couple of extra SOLO LEDs along with the RUDE SOLO light to help you locate a stray SOLO button. They're located just below the Solo MODE switch. The AUX LED blinks whenever the SOLO switch for any AUX SEND MASTER or GLOBAL AUX RETURN SOLO is engaged. The SUB LED blinks whenever a subgroup solo is engaged.

AUX Send Masters

The AUXiliary section is a mixer within the mixer (several of them, actually) and the AUX Send Master controls are the masters for those mixers. This is one of those areas where you'll find substantial differences among models as to how many controls are provided.



Not all models have a Master for every AUX bus. That's an SR AUX Master section on the left and a 1604VLZ-PRO section on the right. Those without Master controls

(AUX 3-6 on the 1604) send the AUX mix to their respective AUX OUT jacks at Unity Gain, relying on the individual channel AUX send levels and the input level adjustment of the effect device or monitor amplifier receiving the AUX signal for overall level adjustment. The Quick Start Guide, the block diagram, or your mixer's control panel is the place to look to see just what your mixer has.

You'll probably want a master level knob handy to adjust the overall volume of AUX mixes used for stage monitors or studio headphone cues. If your mixer doesn't have a Master for every AUX Send, reserve those unity gain AUXs for effects that don't need level tweaking once they're set.

Just a reminder—AUX Send signals can be tapped off either before or after the channel fader, as selected by the PRE/POST switches or determined by controls which are permanently assigned pre- or post-fader. As a rule of thumb, PRE-fader sends are used for monitor feeds, POST-fader sends are used for effects.

SR and 8-Bus AUX Masters

The SR and 8-Bus consoles have individual Master level controls and SOLO switches for each AUX bus. The Unity Gain setting is straight up, with up to 15 dB of boost available. Of course they go all the way off. ("Our knobs go to zero"—Mike Rivers, retired Mackie tech writer)

VLZ-PRO AUX Masters

The VLZ-PRO series of mixers have Masters for some, but not all the AUX busses.

The 1202VLZ-PRO has a Master only for AUX 1, but there's a switch below it which selects whether AUX 1 is pre- or post-fader.

For a stage monitor, select PRE and you'll have a handy knob to turn when the lead singer cups his hand to his ear, frowns, and points down at the monitor speaker (the universal signal for "I can't hear myself. TURN IT UP!!!!").

Other VLZ-PRO mixers have Masters for AUX 1 and 2 which can be switched PRE or POST at the channel strip.

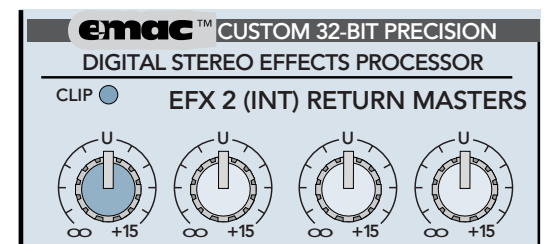
CFX, DFX, and PPM AUX Masters

The CFX, DFX, and PPM series mixers with their built-in EMAC effect processor treat the AUX Masters a bit differently. Like the SR and 8-Bus, there's a Master for each AUX bus, but they don't look quite the same on the panel as the other mixers.



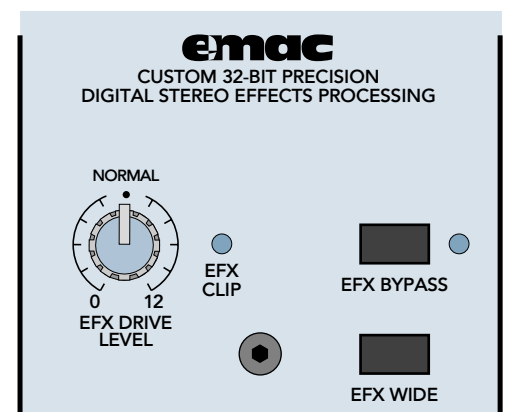
On the CFX series, Masters for AUX 1, AUX 2 and EFX 1(EXT) are grouped next to the Graphic Equalizer. These control the level of their respective OUT jacks.

Since the EFX 2 (INT) bus is internally normalised to the EMAC processor, its Master, labeled EFX 2 SEND, is located along with the EMAC controls.



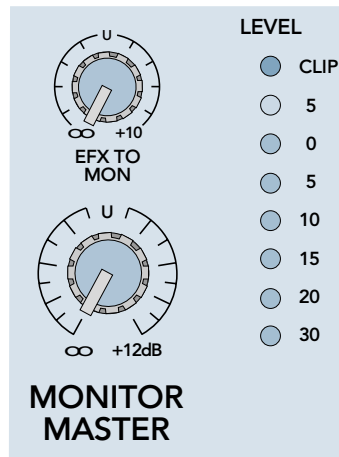
If you're using an outboard effect processor or using the EFX SEND 2 output for an additional monitor feed, the EFX 2 control will adjust its level.

On the PPM series, the Master for the EFX bus is located in the EMAC section and is labeled EFX DRIVE LEVEL.



Normally this controls the master level to the EMAC processor (the EFX Clip light next to it will tell you when you're driving

it too hot), but if you plug an outboard processor into the EFX SEND jack, the level to that processor will be controlled by EFX DRIVE.



The MONITOR MASTER on the PPM is located between the upper graphic equalizer and meter. Normally it controls the level to the internal power amplifier that's

assigned to the monitors, but it *also* adjusts the level at the MONITOR LINE OUT jack.

On the DFX mixers, the AUX 1/MON SEND allows you to control the level of the signal going out to your monitors.

Any signals from each channel's AUX 1/MON control (and the EFFECTS TO MONITOR control), are summed together, then sent through this master control.

Turned fully up, it provides 6 dB of gain, "U" is unity gain, and fully down is off.

When the talent wants a louder monitor mix, this is the control to crank up—watch out for feedback!

The AUX 2/EFX SEND effects send signals are derived by each channel's AUX 2/EFX control, summed together, then sent through this master control. It allows you to simultaneously adjust the signal levels out to the AUX 2/EFX SEND output, and to the EMAC effects processor.

Turned fully up, it provides 6 dB of gain, "U" is unity gain, and fully down is off.

Use the individual channel AUX 2/EFX controls to adjust the amount of each channel's signal you want to go to the AUX 2/EFX SEND or EMAC. Then use this master control to make sure the EMAC's LEVEL SET LED never lights more than occasionally.

AUX Master SOLO

Some models have SOLO switches for the AUX masters. The 1604 and 1642VLZ-PRO have SOLOs for AUX 1 and 2 only, while

the SR series and 8-Bus have SOLOs for all the Auxiliary sends. When using the AUXs to feed stage monitors, soloing them in the headphones is a good way to check what the musicians on stage are hearing from the monitor speakers.

The AUX SOLOs are pre-fader (OK, so the rotary knob isn't exactly a fader) on all but the 8-Bus where they're post-fader. Since the 8-Bus was designed primarily for recording applications, the post solo will be most useful for checking the level going to an outboard effect processor. As studio effects are often installed in a rack out of easy reach and view of the engineer, it's handy to be able to check their input level right from the console. Calibrate the setting of the input level control on the processor so that that 0 VU on the processor's meter (or around -16 if it's a digital processor with 0 dB at the top of the meter scale) corresponds to 0 VU on the console meter and you can always check your processor send level right from the console.

There are a few extra reminders, in addition to the RUDE SOLO light, to help you figure out just what the heck you've soloed. On the VLZs, there's an LED next to each AUX SOLO button. On the SR, there's an AUX SOLO light near the SOLO LEVEL control, and on the 8-Bus, you'll find a single Solo LED below the group of AUX SEND Masters.

AUX RETURNS

All Mackie Auxiliary Returns come in pairs since most effect devices and auxiliary line level sources have stereo outputs. (the PPM series is the exception—stay tuned)

Signals from the AUX inputs are returned, through level controls, to the MAIN bus where they're mixed with the channel signals. On some models, Aux Returns can be routed to subgroup busses. There are also some neat tricks for adding effects to monitors at the press of a button.

AUX RETURN Level Controls

Each pair of stereo AUX RETURNS is controlled with a single knob. With all Mackie mixers, unless you have another button pressed (which we'll get to real soon), signals passing through these controls will proceed directly to the MAIN MIX bus.

When using the AUX RETURNS as inputs from a parallel-connected effect processor driven from an AUX Output, the RETURN Level controls can generally remain at their Unity Gain (center) position. You'll usually get the right level into the mix with the effect processor's output level set for whatever that manufacturer calls unity gain. Mackie AUX RETURN controls can provide up to 20 dB of gain to accommodate effects with lower output levels such as those designed to work in line with electric guitars and of course you can turn the level of the effect down by going below the Unity gain point.

Yes, you *can* use a favorite stomp box as an effect with your Mackie mixer. You'll probably need to back down the level of the AUX MASTER that's feeding it, since units with low level outputs are generally designed to work with lower level inputs.



The nominal output level of effect processors covers a pretty wide range depending on their intended application and era of manufacture. Old effect processors can be pretty cool! Some of them can be pretty darned noisy, though. With proper gain management you can minimize the amount of noise creeping into your mix from these devices.

Using Noisy Effect Units

While this deviates from the **Level-Setting Procedure** that we keep pushing, sometimes your ears can be a better guide than the meters. If your effect is too noisy at Unity return gain, try this:

First get a pretty good mix going without the effect. If the effect unit has an Output Level control, turn it up about 3/4 full. With the multitrack recorder stopped (or the band, if you're working live), raise the AUX RETURN level until you just start to hear noise from the effect, then back it down just a touch. Now, with the music playing, raise the SEND level to the effect until you hear as much of the effect as you want in the mix. Be careful not to overdrive it. Listen for distortion.

PPM Series AUX RETURN Level Controls

The PPM series mixers have AUX RETURN level controls too, but like the Sends, they're not quite so obvious—their names have been changed to protect the

innocent. Normally the EMAC processor is the source for the effect return input, so the EFX TO MAIN and EFX TO MONITOR controls serve as the AUX RETURN Level controls to the main and monitor busses.

Mono PPM mixers have only a single EFFECTS RETURN jack. The stereo models have two, but unlike the other Mackie mixers, they aren't normalled to accommodate a mono source. EFFECTS LEFT RETURN goes to the LEFT MAIN bus and RIGHT RETURN goes to the RIGHT MAIN bus. Plugging into the RETURN jacks disconnects the output of the EMAC processor, but the level controls are still active.

To use a mono effect processor and send its output to both channels, you'll need a Y-Cable to split its output to the two jacks. Don't confuse this with the *Insert* cable we described previously, which also has one plug on one end and two plugs on the other but is wired differently.

The EFX TO MAIN knob serves as the AUX Return level control. The EFX TO MONITOR control sends a mono sum of the left and right returns to the Monitor bus.



It's possible to use an external processor and the EMAC at the same time. If the outboard processor will be used on only a single channel, it can of course be patched in line through the channel INSERT jacks, but there's more.

The EFFECTS SEND output is a TRS jack wired just like the INSERT jacks, so you can use it to patch a "serial connected" processor in line ahead of the EMAC unit. The outboard processor will receive the EFX mix, and its processed output will become the input to the EMAC.

Another way to skin this cat is to take the "parallel" route. Plug a TS plug halfway into the EFFECTS SEND jack (to the first click) and you now have a direct output from the Effects bus with which to feed the outboard processor while still leaving the normal feed to the EMAC intact. Bring the outputs of the outboard effect processor into the mix through a pair of mono or a stereo line level channel inputs and you can blend in the desired amount of the outboard effect.

Remember that the EFX control on the channel to which you're returning the effect processor is still active, so you can send the processor's output back into the processor.

This often results in an undesirable feedback loop, but at times, particularly with delays, it can be cool. Don't be afraid to experiment—but only at times when the audience won't hear your bad mistakes.



Don't be tempted to plug an outboard effect processor's output into the EFFECTS RETURN jacks if you also want to use the EMAC. Doing this will disconnect the EMAC output from the mix.

While you won't have much control, if you're running a stereo PPM in the mono/monitor mode it's possible to plug a mono effect output into one of the Effects Return jacks and have both one channel of the EMAC and the outboard effect going to the main mix. Just in case you were wondering.

CFX AUX RETURN Level Controls

On the CFX series mixers, you'll find EFX 1 RETURN just to the left of the meter. It works identically to the conventional AUX RETURN Level controls, sending the Return signal directly to the Main mix bus.

The CFX mixers are similar to the PPM series, with the built-in EMAC processor normally connected to the EFX 2 send, and returning the processed signal to the main and monitor busses through the EFX TO MAIN and EFX TO MONITOR (AUX 1 and AUX 2) controls.

Plugging into the EFX 2 RETURN jacks disconnects EMAC processor output signal and replaces it with whatever you've connected to those RETURN jacks. The EFX TO MAIN MIX control on the EMAC panel becomes the Return Level for the EFX 2 signals. EFX TO MONITOR controls send the return signal to the AUX 1 and AUX 2 busses.

Like the PPM (and *unlike* the other compact mixers), the STEREO EFX RETURN jacks aren't normalised to accommodate a mono return signal, they're strictly stereo. If you want to connect a mono source to the EFX RETURNS, you'll need a Y-cable to split its output to the two jacks.

Unlike the PPM, the RETURN 2 jacks aren't wired as inserts, so you can't patch a processor in line with the EMAC outputs, nor can you use the EFX SEND 2 as a direct output from the EFX 2 mix. If you want to use a second processor along with

the EMAC, you must do it through the EFX 1 connections.

The EFX BYPASS switch and foot switch will kill the output of the EMAC, but not the output of a processor plugged into the EFX RETURN 2 jacks.

DFX AUX RETURN Level Controls

AUX 1 RETURN Inputs

Patch the outputs of external parallel effects devices to these inputs.

Signals coming in are adjusted using the AUX 1 RETURN fader and MUTE.

When connecting a mono device (just one cord), always use the left (mono) input and plug nothing into the right input. The signal will appear on both sides.

Any signals going into the AUX 1 RETURN jacks from an external processor pass through the MUTE switch and then to this stereo fader. This is like a channel fader, except it is adding the signal from an external device to the main mix, rather than adding a channel signal. Turned fully up, it provides 10 dB of gain, "U" is unity gain, and fully down is off.

AUX 2 RETURN Inputs

Patch the outputs of external parallel effects devices to these inputs.

Signals coming in are adjusted using the AUX 2 RETURN fader and MUTE.

When connecting a mono device (just one cord), always use the left (mono) input and plug nothing into the right input. The signal will appear on both sides.

Note: The AUX 2 RETURN signal is combined with the signal from the internal effects processor.

The EMAC output is added to any signals going into the AUX 2 RETURN jacks from an external effects processor. The final effects signal is then sent to this MUTE switch and stereo fader. It acts like a channel fader, except it is adding the final effects signal to the main mix, rather than adding a channel signal.

Turned fully up, it provides 10 dB of gain, "U" is unity gain, and fully down is off.

Press MUTE to quickly compare the difference between adding, or not adding effects. The MUTE and fader affect the level of the

effects going to the main mix only, and not of any effects going to the monitors.

Effects To Monitor

Typically, this control is used to add effects to the stage monitors. Any effects sent to the monitors are the same as in the main mix, except for the level, and they are summed mono. The effects level sent to the monitor mix is not affected by the AUX 2/ EFX RETURN fader, so the monitors and main mix effect levels can be set independently.

Turned fully up, this control provides 6 dB of gain, and “U” is unity gain. Turn it fully down for no effects sent to your monitors.

This control is similar to each channel’s AUX 1/MON (21) control, except it adds the effects (summed mono) to the monitor mix, rather than adding a channel signal.

EFFECTS TO MONITORS

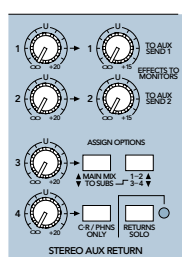


We’re about to get into a very cool feature here, but in order to keep from getting confused, you need to understand something that’s not intuitive. It would seem logical that AUX SEND 1 and AUX RETURN 1 go hand-in-hand—send out a signal on 1, and bring its return back on 1, right? Well, in reality there’s no connection between the two. They’re just names, like two people named Fred.

You can connect the output of an effect processor to any RETURN regardless of which SEND is feeding the processor. It’s easier to dig this if you remember that a Return signal can come from *any* line level source (like a CD player or a drum machine), not just a reverb unit fed from a SEND on the mixer.

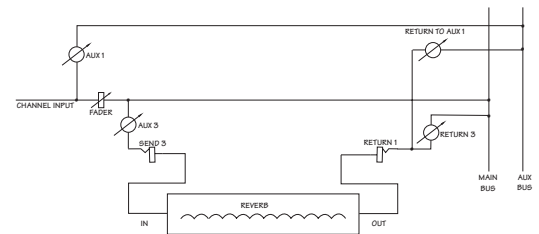
VLZ-PRO and SR Mixers

If you want to put some reverb or delay in the stage monitors or headphones, this knob’s for you. Here’s the 1604VLZ-PRO AUX RETURN section as an example:



The TO AUX SEND 1 and 2 controls are like AUX 1 and AUX 2 Send controls on the main channel strip. The difference is that instead of tapping a signal off the channel input, they tap off the AUX RETURN 1 and RETURN 2 jacks.

Typically, you’d feed a reverb unit from a post-fader send (3-6). By connecting the reverb outputs to RETURN 1, you can add reverb to the main mix using the AUX RETURN 1 Level control *plus* add the same reverb to a monitor fed from the AUX SEND 1 jack. Same for RETURN 2 going to SEND 2. Here’s a simplified block diagram of the signal flow:



The EFFECTS TO MONITORS controls are fed ahead of the AUX RETURN controls, so if the singer in the studio wants to drown his ego in reverb but the producer in the control room wants to hear how well he’s articulating the words, you can send just a tad of reverb to the control room mix and give the singer as much reverb as he wants.

Since the RETURNS are stereo but the SENDS are mono, the left and right returns are summed to mono before hitting the EFFECTS TO MONITORS controls.



Some chorus and flange effects achieve a wide stereo spaciousness by throwing one channel completely out of phase from the other channel. This can sound cool in stereo, but when the two channels are added together and become a mono source, the effect can practically disappear because one channel will nearly cancel the other. If you’re hearing a nice rich and spacious chorus in the stereo mix and the person on the other end of the monitor chain isn’t hearing it, this may be what’s happening. One compromise solution is to use only a single output from the chorus unit as the effect return. It won’t be as effective an effect in stereo, however.



You can create a feedback loop if you use AUX SEND 1 to an effect processor which you return to AUX RETURN 1, and then send that to the monitor with the EFX to MONITOR switch with nothing connected to RETURN 2. It’s best to try to avoid this connection. Just return the effect processor you want to share with the monitor to RETURN 2 and you’ll be safe.

CFX, DFX, and PPM Mixers

The CFX, DFX, and PPM mixers also have the ability to send an effect to the monitors. On these mixers, the EFX TO MONITOR controls are fed from the EMAC or an out-board processor plugged into the EFX RETURN 2 (EFFECT RETURN on the PPM) jacks. The two EFX TO MONITOR controls on the CFX send the same effect (from the EMAC or the RETURN 2 jacks) to AUX 1 and AUX 2 through independent controls so you can give the singer a lot of reverb and the drummer just a little. The DFX and PPM have a single EFX TO MON control.

Other AUX RETURN Routings

Since subgroup busses can carry their own independent mixes, doesn't it make sense that those mixes should have access to effects, too? How about sending an effect return to only the headphones or control room monitor without adding it to the MAIN mix? Can do.

ASSIGN OPTIONS—Main Mix/To Subs (I604 and I642VLZ-PRO)

Don't make the mistake of reading this button as "Main mix to subs." While there may be an occasion to send the main mix to the SUB OUTs, that's not what this switch does—it tells RETURN 3 where to go. Its two positions are MAIN MIX (*up*) and TO SUBS (*down*). Make more sense?

With this switch *up*, AUX RETURN 3 behaves like all the other RETURNS—it delivers a stereo signal, controlled by its Level knob, to the MAIN mix. When you engage this switch, the RETURN signals are removed from the MAIN busses and are sent to the 1-2/3-4 switch right next door. This switch selects which subgroup pair receives the RETURN 3 signal.

With this switch in the MAIN MIX (*up*) position, the 1-2/3-4 switch does absolutely nothing. It's only functional in the TO SUBS (*down*) position where, with the 1-2/3-4 switch *up*, RETURN 3 will go to subgroups 1 and 2. Push the button *down* and RETURN 3 will go to subgroups 3 and 4.

Let's say you have a drum submix on subgroups 1-2 that's assigned to the MAIN

mix. You have a reverb for the drums returning to the mixer on RETURN 3.

You could get the drum reverb into the mix simply by letting RETURN 3 go directly to the MAIN mix (MAIN MIX/TO SUBS switch *up*), but there's a better way. Instead, switch Return 3 TO SUBS and set the 1-2 / 3-4 switch *up*. Now the reverb return will be blended into the drum mix on SUBS 1-2. As you ride the SUB 1-2 faders, the reverb level will follow.

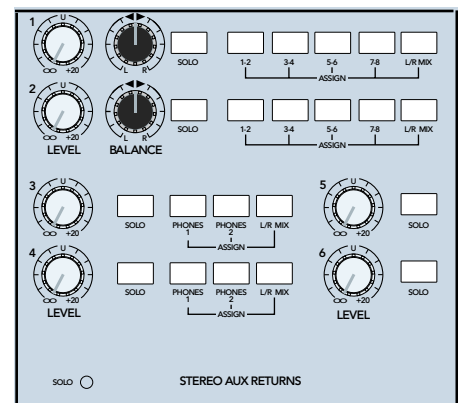
Why do we want to do that? Because if you had simply sent the reverb directly to the MAIN mix and did a drum fadeout using the SUB faders, the reverb wouldn't know anything about that fade and would just go on sending "wet" drums to the mix as long as the drums were playing. While that might be a cool effect sometimes, normally it's not what you want. And that's why we threw in those switches.

ASSIGN TO SUB (SR consoles)

By the time we got around to designing this console, we recognized the confusion with a switch labeled MAIN MIX TO SUBS so we more clearly renamed it ASSIGN TO SUBS. Its companion switch is labeled SUB, with positions 1-2 (*up*) and 3-4 (*down*). These two switches serve the same function as those described in the previous paragraph only they work on RETURN 4 rather than 3. Same train, different time.

8-Bus

The 8-Bus console has some very flexible AUX RETURN routing options.



RETURNS 1 and 2 can be assigned to the Main L/R MIX and/or any of the four pairs of subgroup busses by pressing the button for that bus pair.

RETURNS 1 and 2 also have a BALANCE control which can be used to adjust the

position of a stereo return in the mix. If a stereo effect is used for overall ambience, the BALANCE control will normally be set to the center, distributing it equally to both busses. If an effect is being used as an integral part of a sound, you can use the BALANCE control to move the apparent position of the effect return to coincide with the panned position of the source—wailing sax on the left, sax-drenching reverb on the left, sax reverb stays out of the way of the articulately tinkly mandolin on the right.

RETURNS 3 and 4 can be assigned to the PHONES 1 or 2 buses in addition to the L/R MIX by pressing the appropriate ASSIGN button(s). This is similar to assigning a return to an AUX monitor bus in other mixers.

RETURNS SOLO

You can solo the channel inputs, so it's logical to be able to solo the returns, too. You can on some models.

8-Bus Returns Solo

Each of the RETURNS on the 8-Bus console has its own SOLO button. Pressing it sends the RETURN signal to the headphone and control room outputs. Like all solos on the 8-bus, these are post-fader, so the solo level will change as you adjust the Return Level pot.

Global Returns Solo

The 1604 and 1642VLZ-PRO and the SR series mixers have a single Returns SOLO button. On the VLZs, it's called RETURNS SOLO, while on the CR it's labeled GLOBAL AUX RETURNS SOLO.

All of the return signals are summed and sent to this switch, so when you have more than one RETURN signal, what you hear can be confusing. The effect returns solo function follows the PFL/AFL SOLO MODE switch, so by selecting AFL and turning the Return Level control down for all but the return you want to hear, you can solo a single return signal.

RETURNS to CR/PHONES ONLY

Remember, the default for all the STEREO AUX RETURNS is to feed them directly into the MAIN MIX. You've learned about diverting one of the AUX RETURNS to the subgroup busses, but the 1604 and 1642VLZ-PRO provide another option for routing RETURN 4.

Engaging the C-R/ PHNS switch removes AUX RETURN 4's stereo signal from the MAIN MIX and sends it directly to the CTL ROOM/PHONES monitor bus.

Let's say you're doing a direct-to-stereo recording of a live performance, and the drummer wants to play to a click track in the headphones which you've plugged into the PHONES jack. If the click track went directly into the MAIN MIX, then the recorder and audience would hear it—not very cool. By gum, this is the switch for you. If you connect the metronome to RET 4 and engage the CR/PHONES ONLY switch, the click will only go to the phones or the Control Room monitor speakers, not the MAIN L/R outputs.

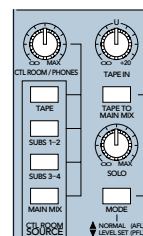
Similarly, this switch can be used to add reverb to the headphone and control room mix when tracking, while still allowing you to record dry and process later.

CONTROL ROOM SOURCE SWITCHES (Matrix)

Normally you want to hear the MAIN MIX in the control room, but sometimes you want to hear something else—tape playback of that song you just mixed, or a subgroup that you're sending to the recorder. We've provided a handy set of pushbuttons to select your monitor source.

1604/1642VLZ-PRO CR Source Matrix

Here's the CTRL ROOM SOURCE section from the 1604VLZ-PRO:



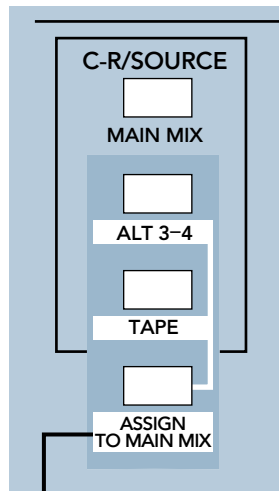
The four SOURCE buttons send the MAIN MIX, SUBS 1-2 or 3-4, or the TAPE IN to the CONTROL ROOM OUT jacks and also to the headphones. To play back the mix from your stereo mix-

down recorder (which is connected to the TAPE IN jacks), de-select MAIN MIX and select TAPE. Normally you'll hear the playback at the same level as the mix if the TAPE IN control is set to its Unity gain position. If you need to match the playback level to the mix level (and you should do so, so you aren't thrown off by the difference in levels), use the TAPE IN control.

Tape to Main Mix (I604/I642VLZ-PRO)

The TAPE TO MAIN MIX switch sends the TAPE input, through its Level control to the MAIN outputs. Use this switch when you want to play intermission music from a tape deck or CD player without taking up a pair of channels or AUX return.

I202/I402VLZ-PRO CR Source Matrix



The 1202 and 1402VLZ-PRO mixers are similar, with ALT 3-4 replacing the two SUBs. Pressing the ALT 3-4 button in the CR/SOURCE group routes any channels assigned to the ALT 3-4 bus (MUTE switch pressed) through the CTL ROOM/SUBMIX fader to

the headphone jack and Control Room output. Note that with this button (or the TAPE button below it) pressed, the meters read the ALT/TAPE level, not the MAIN MIX level.



These buttons aren't interlocking, so you can press more than one at once. If you're using the ALT 3-4 outputs to feed a sound card for recording on a computer, you can combine the sound card playback returned to channels assigned to the MAIN mix with the ALT 3-4 output that you're recording. This is one way of monitoring what you're singing or playing while overdubbing to previously recorded tracks. See the Hookups section for details.

ASSIGN to MAIN MIX (I202/I402VLZ-PRO)

The ASSIGN TO MAIN MIX button is associated with the ALT 3-4 and TAPE buttons immediately above it, and the CTL ROOM/SUBMIX fader below it. The lines connecting them will remind you of the connection.

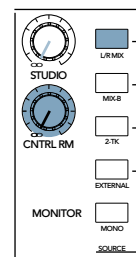
When both the ALT 3-4 and ASSIGN TO MAIN MIX buttons are pressed, the CTL ROOM/SUBMIX control becomes a master for channels assigned to the ALT 3-4 bus. Raising the SUBMIX fader brings the submix of those channels into the MAIN mix.

Pressing the TAPE and ASSIGN TO MAIN MIX buttons sends a tape or CD connected to the TAPE IN jacks to the MAIN outputs. Use this in a PA application for playing recorded music during a break, or for copying a tape to another recorder connected to the MAIN outputs.



Feedback Alert! If you have a recorder's inputs connected to the TAPE or MAIN OUTPUTs and its outputs connected to the TAPE IN jacks, if the ASSIGN TO MAIN MIX button is pressed while the recorder is in the Record or Record Ready mode, you'll send its output back to its input, creating a feedback path. It's pretty obvious with a hardware recorder, but if your recorder is a computer with a sound card, you should check the "input monitor" settings carefully.

8-Bus Control Room/Studio MONITOR Switch Matrix



On the 8-Bus monitor source selection matrix, the TAPE IN source is labeled 2-TK (2-Track). IN addition, there's an EXTERNAL button which could come from a second stereo recorder, cassette, or CD player. There's no separate level control for the 2-TK or EXTERNAL sources, so you may have to re-adjust the CNTRL RM MONITOR Level control for a comfortable playback level if your recorders don't have playback volume controls. It's right handy.



The MONO switch in this group combines the left and right inputs to the monitor section, allowing you to hear your mix in mono. While this

might not seem very important, think of how many people wake up to a clock radio or hear television audio in mono. Sometimes a stereo source or effect can become weak or drop out of the mix in mono due to the phase relationships between the channels. It's a good idea to check any mix in mono, and on the 8-Bus, we've made it very convenient.

The monitor source that you select for the Control Room also goes to the STUDIO OUTPUT jacks, which have their own level control.

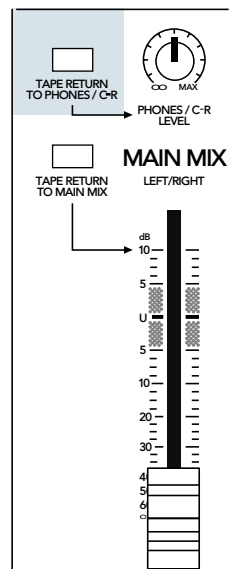
TAPE IN (RETURN) ROUTING

All Mackie mixers have stereo TAPE INPUTS and an associated Level control. Depending on the model of the mixer, that TAPE INPUT can go to several places.

CFX and PPM Mixers

On these mixers, the TAPE IN can go to only one place—the MAIN mix (is this true for the DFX?). Nothing fancy here. The TAPE inputs are active regardless of the setting of the BREAK switch (coming up shortly).

SR Consoles



On the SR series mixers, the TAPE INPUTS don't go anywhere until you've assigned them. The two TAPE RETURN assignment switches are located in the vicinity of the MAIN MIX fader. The TAPE RETURN TO PHONES/CR switch sends the tape to the control room and phones outputs. It's equivalent to the TAPE switch in the CR Matrix of the VLZ mixers. The TAPE

RETURN TO MAIN MIX switch assigns the TAPE INPUTS to the MAIN mix, just like it says. This is how to feed intermission music to the house speakers or bring a tape playback into the mix without eating up a channel.



WARNING: Assigning the TAPE to the MAIN MIX can create a feedback path between TAPE INPUT and TAPE OUTPUT. If you're using the tape deck for playback only (intermission music), this is not a problem, but watch out if you're recording your mixes. Make sure your tape deck is not in *record*, *record-pause* or *input monitor* mode when you engage this switch, or make sure the TAPE IN level knob is fully counterclockwise (off).

CD/TAPE RETURN Fader and MUTE (DFX Mixers)

If you have a CD or Tape player connected to the CD/TAPE inputs, the signals pass through the MUTE switch, through this stereo fader, and then into your main mix. This is like a channel fader, except it is adding the CD/TAPE input to the main mix, instead of adding a channel.

Set this fader to minimum, then play your selection and increase the fader as desired. Assuming the MAIN MIX faders are set, you should hear the music. Press down the VOCAL ELIMINATOR if you like to remove centered vocals, and BREAK if you want to play soothing CDs or tapes while the band is breaking up the bar.

BREAK SWITCH

No, we don't want you to break anything, nor does this switch dial Mackie's toll-free phone number when your mixer breaks (that'll be never). But the band gets a break now and then, and if you're lucky so do you, the sound engineer. Walking away from the mixer in a crowded club with live mics on stage can be embarrassing. What if a drunk (called a *customer* in polite circles) walks up to the stage and starts singing into a mic? Or someone starts fooling with the knobs and pushes the system into feedback?

To avoid such disasters, we've provided a handy BREAK switch on the CFX, DFX, and PPM mixers.

CFX BREAK SWITCH

On the CFX, pressing the BREAK switch disconnects the L/R bus from the final output section of the mixer and switches it safely to the TAPE IN jacks. Pressing the BREAK switch kills all the mics in the MAIN, SUBWOOFER, and UTILITY out-

puts. Start up a CD or tape connected to the TAPE IN jacks (or sometimes silence is nice, too) and you can take that needed pause for the cause.



The BREAK switch doesn't kill the AUX outputs, so your stage monitors will still be live. If you're going to be out of shooting range of the console for very long, turn the AUX MASTERS that feed the monitors down, too. Be sure to note where they were set so you can return them to the proper level when the band starts the next set. You know the first thing they say after "Thank you" will be "More monitors, please", but you might as well start where you last left off.



The BREAK switch also doesn't kill the signal going to the TAPE OUT jacks. If you're recording the show and leave the tape deck running, you'll get a nice recording of the noise during the break. While this might seem like a big waste of tape, if you're thinking of releasing a live album, blending in some club "atmosphere" when editing down the recording is a useful technique to make your listener feel more "there".

PPM BREAK SWITCH

The BREAK switch on the PPM mixers serves the same purpose, but it works a little differently. Pressing the BREAK button mutes Channels 1-6, the MONITOR output, and the EFX.

The switch doesn't affect the TAPE IN or stereo Channels 7 and 8, so you can play intermission music, and also leave a mic live on stage for the MC or the comedian (the one the club booked, you hope).

DFX BREAK SWITCH

Engage the BREAK switch (as in take-a-break) and the entire main microphone mix is switched off.

The CD/TAPE input is still active, so you can play your intermission CDs or tapes through the main outputs. On the DFX•12, line level inputs 9 and 10, 11 and 12 are also still active, and are summed with the CD/TAPE input. During an intermission, you could play keyboards or a second tape deck into these line inputs.

BREAK acts as a master mute switch for your microphones, so it is useful when moving microphones around on stage. It

does not mute the AUX 1 or AUX 2 outputs, so turn down the master AUX SEND knobs if desired.

VOCAL ELIMINATOR Switch (DFX only)

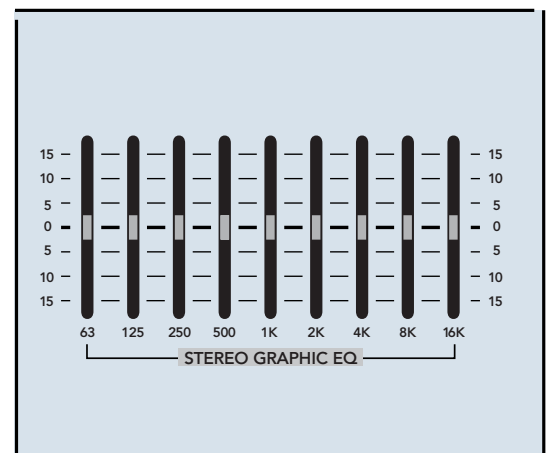
This switch allows you to eliminate the vocals from any CD or Tape selection playing in the CD/TAPE inputs.

In the UP position, the CD or Tape signals are not affected. In the DOWN position, any vocals which are centered in the normal stereo image will be attenuated.

The circuit design uses phase cancellation of the vocal frequency range between the left and right channels. The effect is most noticeable on music material which has the vocals panned dead center.

Master Section Graphic EQ

In sound reinforcement applications, it's fairly common to include some equalization between the mixer output and the speaker system so that the overall mix can be fine tuned to the room acoustics. The PPM and CFX mixers provide graphic equalization on their outputs.



The name "Graphic" comes from the appearance of the equalizer controls. A graphic EQ is a collection of several (nine in this case) fixed-frequency filters with the amount of boost or cut adjusted by vertical sliders. The knobs on those sliders form a graph of the frequency response through the equalizer. The frequencies are spaced at intervals of one octave (bet you didn't know you were going to get a music appreciation lesson in a mixer manual!), giving it the name "Octave-band equalizer". Other graphic equalizers use narrower band filters

spaced at intervals of one-third or one-half octaves.

Graphic equalizers are often used to reduce feedback by reducing the gain at the frequency at which the system is ringing. See the Tips section to learn how (and how not) to do this.

CFX Graphic EQ

The graphic equalizer on the CFX mixers is stereo—each knob controls two filters. Equalizer inputs come from the BREAK switch (either the MAIN L/R mix or a tape player for intermission music) and the equalizer outputs go to the MAIN LEVEL controls.

PPM Graphic EQ

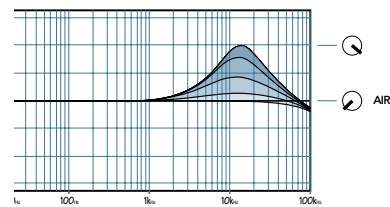
The PPM mixers, both the mono and stereo versions, have two graphic equalizers, one for the MAIN output, the other for the MONITOR output.

The MONITOR EQUALIZER is the last thing in the signal chain prior to the MONITOR LINE OUT jack. If you're operating with the POWER AMP ROUTING switch pressed *in*, with the Right power amplifier feeding the monitor speakers, this EQ will adjust the frequency response in the monitor speakers connected to the RIGHT (stereo PPM) or POWER AMP 2 (mono PPM) jacks. If you're using an outboard monitor amplifier or powered speakers connected to the MONITOR LINE OUT jack, the MONITOR EQUALIZER will tweak it.

The MAIN-STEREO/MONO EQUALIZER operates on the MAIN outputs. It's the last thing in the chain before the MIXER LINE OUT (mono) or L and R MIXER OUT (stereo) jacks. The MAIN equalizer on the stereo PPM models is stereo, so with the AMP ROUTING switch set to STEREO MAINS, both power amplifiers will receive an equalized signal.

AIR Equalizer (SR Only)

AIR is a high frequency peaking equalizer placed at the output of each of the four subgroups. It's centered around 16 kHz and provides a maximum boost of 10 dB. It's boost-only, so turning the AIR knob fully down leaves the frequency response flat.

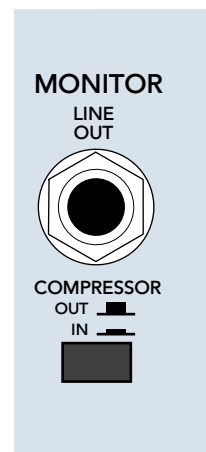


Note that small amounts of AIR boost give a gentle rise in frequency response at the top end of the vocal range.

If a vocal is assigned to a submaster, AIR can give it a little, well . . . air. Be careful with large amounts of AIR boost. It can add a crystalline quality to certain sounds, particularly synthesizers, but if the synth is noisy, a healthy shot of AIR will also boost the hiss.

COMPRESSOR (PPM Only)

The PPM mixers are equipped with a compressor at the input of each of the two power amplifiers.



The COMPRESSOR IN/OUT switch is located just below the MONITOR LINE OUT jack. Pushing this switch activates the compressor that senses the signal level going into the power amplifiers and reduces high level signals to reduce the possibility of driving the amplifier into clipping.

Clipping is nasty distortion which not only sounds bad, but can actually damage your loudspeakers.

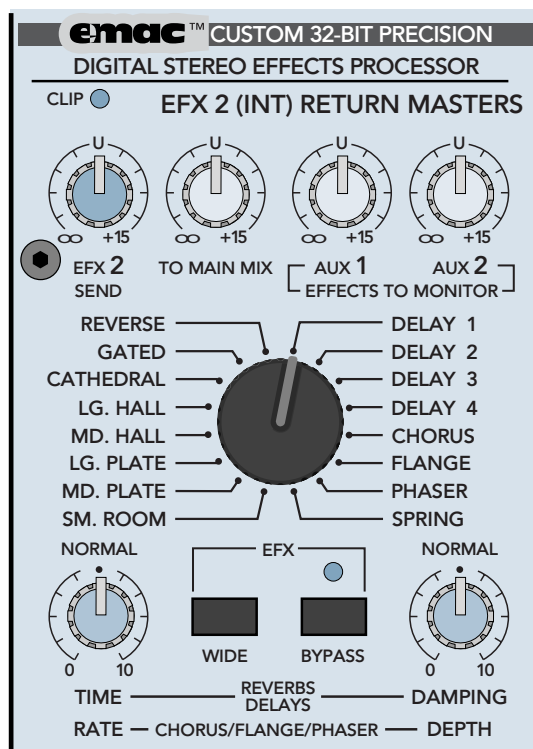
Keeping the compressor engaged is a good precaution but when it's really working hard, it can contribute its own flavor of distortion to your mix. Pay attention to what's coming out of the speakers, and if you hear the compressor in action, back off. The amplifiers aren't going to get any louder, so they might as well play cleanly.

Chapter 5: EMAC Effects Processor

EMAC™ stands for Extended Multiply and Accumulate, which is a proprietary 32-bit digital stereo processor developed by the Mackie Digital Engineering Group. It provides the CFX and PPM mixers with 16 preset digital effect algorithms to enhance your mix. In addition to the presets, there are two parameter controls which adjust parameters appropriate for the type of effect you've selected. You can tweak the preset values to change the sound and make it unique for your particular application.

In this section, we'll describe the effects and how to adjust them. We've already explained how signals get into and out of the EMAC processor, but we'll review those controls in case you jumped in here.

The CFX, DFX, and PPM mixers have the same EMAC functions, the only difference being how the input and output controls are labeled and where they're located. This illustration shows the EMAC section from a CFX mixer. Differences in labels on the PPM and DFX mixers are enclosed in [BRACKETS].



EMAC Level Setting

To get started, set the input and output levels to Unity gain [NORMAL]. Generally you'll be able to leave these controls alone as this will give you the best signal-to-noise

ratio, but they're there if you need them. *Note that when we talk about MAIN MIX here, we're talking about the control in the EMAC group of the CFX mixer, not the Master main mix fader.*

On the CFX, the input level to the EMAC is controlled by EFX 2 SEND. It's like a Master to the EFX 2 controls on the channel strip. The output level controls to MAIN MIX, AUX 1 and AUX 2 are grouped together and labeled EFX 2 (INT) RETURN MASTERS.

On the PPM, the input control is EFX DRIVE LEVEL, and outputs are EFX TO MON and EFX TO MAIN (above the MASTER knobs, next to the meters.

Adjusting the Effect Drive Level

Use the EFX2 [EFX] controls on the individual channels to adjust the amount of each channel's signal you want to go to the EMAC. If you find that you're not getting enough of the effect in the main mix, make sure that the MAIN MIX [EFX TO MAIN] control is turned up at least to unity. It's okay to turn up the EFX DRIVE LEVEL some more if you need to. Just make sure the EFX CLIP LED never lights more than occasionally. Read on to find out why.

EFX CLIP

This indicates when the signal going to the EMAC is 6 dB below clipping. This LED should only light occasionally. If it blinks frequently, you should turn down the input to the EMAC a little. Digital clipping is not a pleasant sound to behold.

EFX BYPASS

Pushing in this button causes the EFX BYPASS indicator to light and mutes the effects signal. It affects both the internal EMAC effects and, on the PPM (but not the CFX) an external effects processor you may have connected to the EFFECTS RETURN jack.

BYPASS LED (DFX and PPM Mixers)

This LED will come on when the EMAC circuit is bypassed by use of an external footswitch. The bypass affects only the internal EMAC effects, not any external effects processor you may have connected to the AUX 2 RETURN jacks.

TIME/RATE PARAMETER

If you have a reverb effect selected, this control adjusts how long the reverberation lasts, with 0 being a short reverb time and 10 being the longest.

If you have a delay effect selected, this adjusts the amount of time between the original signal and the delayed signal, with 0 being a short delay time and 10 being the longest delay time.

If you have a chorus, flange, or phaser effect selected, this control adjusts the rate or speed of the modulation of the effect.

These controls aren't calibrated in physical units like seconds, it's just a relative scale of zero to ten, with zero not necessarily being completely off.

DAMPING/DEPTH PARAMETER

If you have a reverb or delay effect selected, this control adjusts how fast the higher frequencies roll off in the reverberation or delay, with 0 having no roll off and 10 having the most roll off.

If you have a chorus, flange, or phaser effect selected, this control adjusts the depth of the modulation of the effect.

EFX WIDE

Depending on the effect selected, this switch adds more width or depth to the effect. Note that it doesn't work with the DELAY and PHASER effects because they are monophonic.

EFFECT PRESET SELECTOR

Rotate this big detented knob to select the preset effect you want to use. Effects fall under the general category of Reverbs, Delays, and Modulation Effects. Take your choice:

REVERB PROGRAMS

These reverbs are designed to provide a wide variety of reverberation sounds for vocal and instrument applications.

REVERSE

Standard reverse reverb, simulating a tail-first effect increasing to the original note. Decay range adjustable from 35 ms to 515 ms. No pre-delay.

GATED

Standard gated reverb where the reverb tail is cut off sharply after preset decay length. Decay range adjustable from 35 ms to 515 ms. No pre-delay.

CATHEDRAL

Dense, smooth reverb with a very long tail, long pre-delay, and late reflections. Tails are very warm with some additional high-end reflections imitating the stone walls of a cathedral. A very dramatic effect that works well with wind instruments such as flute, slow finger picking on acoustic guitar, and quiet vocal group harmony and choirs. Also works well with keyboards and drums using short decay. Decay range adjustable from 2 seconds to 10 seconds. Pre-delay fixed at 75 ms.

LG. HALL

Dense, smooth reverb with long tail, long pre-delay, and some early reflections. Tails are warm with more apparent high end. Works well with vocals and electric and acoustic guitar. Decay range adjustable from 1 second to 5 seconds. Pre-delay fixed at 75 ms.

MD. HALL

Dense, smooth reverb with normal tail, normal pre-delay, and increased early reflections. Tails are warm with more apparent high end. Works well with vocals and elec-

tric and acoustic guitar. Decay range adjustable from 750 ms to 2.5 seconds. Pre-delay fixed at 65 ms.

LG. PLATE

Good early reflections and no pre-delay. Tails are normal and warm with strong high end for increased presence. Perfect for vocals and snare. Decay range adjustable from 1 second to 5 seconds.

MD. PLATE

Good early reflections and no pre-delay. Tails are short and warm with strong high end for increased presence. Perfect for tight vocals and snare. Decay range adjustable from 750 ms to 2.5 seconds.

SM. ROOM

Reverb featuring very fast and scattered early reflections with a short pre-delay. Tails are very short and warm with normal high-end imitating absorbent wall materials and audience. Good for tight vocal effects. Decay range adjustable from 250 ms to 1 second. Pre-delay fixed at 30 ms.

SPRING

Mimics the vintage 60's-style wet spring reverb effect. Tails are normal with strong high end and a slight waver imitating the slow flutter of a mechanical spring system but without the boinnnng! when you bump the cabinet. Very good with acoustic guitar. Decay range adjustable from 1 second to 5 seconds. No pre-delay.

Reverb Parameter Tweaks—TIME, DAMPING, WIDE Switch

When Reverb effects are selected, the TIME/RATE knob controls the reverberation decay time (length of the tail). The range of values adjusted by the control is scaled to be appropriate for the preset you've selected and is different for each preset. 0 is the shortest time (never actually zero seconds) while 10 is the longest time. See the description of the programs for the time ranges.

The DAMPING/DEPTH knob controls the amount that the simulated reflecting surfaces reflect the high frequencies. The darkest tone (greatest absorption of highs) is at 0 and the brightest tone (greatest reflection of high frequencies) is at 10.

The EFX WIDE switch is very effective at increasing the stereo image of the reverb effect in the stereo versions and enhancing reverb density in mono versions.

DELAY PROGRAMS

There are four delays available. They differ in the amount of delayed signal fed back into the delay processor. With no feedback, you get just straight delay—one repeat of the input after the selected amount of delay time. When the repeat is fed back into the delay processor, you get a repeat of the repeat . . . and a repeat of the repeat of the repeat. You get the idea...idea...idea..idea.

Delay 1 has one repeat after the original note. Delay 2 has two repeats, Delay 3 has three, and Delay 4 has four. These are four fixed amounts of feedback which provide good variety of effects without running the risk of getting into an infinite feedback loop.

The TIME knob adjusts the delay time. The DAMPING knob adjusts the high frequency response of the delayed signal, just like it does for Reverb programs.

DELAY 1

One repeat. Works best for the slapback delay used in country and swing guitar, and for rockabilly and some country vocals. Delay range is adjustable from 5 ms to 524 ms.

DELAY 2

Two repeats. Provides a fuller, more dramatic effect for rock and gospel vocals, acoustic guitar, and wind instruments such as flute. Especially effective for some finger-picking styles. Delay range adjustable from 5 ms to 524 ms.

DELAY 3

Three repeats. An excellent delay for slow, bluesy vocals and melodic flute music. This delay usually works best when the channel EFX send is set at less than halfway. Delay range adjustable from 5 ms to 524 ms.

DELAY 4

Four repeats. This is for very dramatic delay effects, particularly for enhancing long vocal notes and dramatic instrumental note-chopping effects. Be sure to set the

channel EFX send at about halfway. Delay range adjustable from 5 ms to 524 ms.

Delay Parameter Tweaks—TIME, DAMPING

When Delay effects are selected, the TIME/RATE knob controls the time between repeats, with the fastest repeats at the 0 position (5 ms) and the slowest repeats at 10 (524 ms).

Adjust this control to match the tempo of the music for the most dramatic effect. Try delaying for a quarter-note or half-note. Delays of triplet length can be pretty cool, too.

The DAMPING/DEPTH knob controls the high frequency response of the delayed signal, with the darkest tone at 0 and the brightest tone at 10. For an analog tape delay sound, use a darker setting so that each subsequent repeat is darker than the prior one. For a crisp digital delay sound, use a brighter setting so the tone retains its high end on each subsequent repeat.

Since the delay effect is not stereo, it is not affected by the EFX WIDE switch.

MODULATION EFFECTS

These effects include Chorus, Flange and Phaser, and are generally used for enhancement of instrumental music. However, Chorus adds a dramatic effect to vocals as well.

Unlike reverbs and delays where, once parameters are set, they stay put, in a Modulation effect, some parameter is constantly being *modulated*, hence the name.

CHORUS

Provides a soft, ethereal sweeping effect. Perfect for enhancement of electric and acoustic guitar and bass. Also adds a dramatic effect to vocals, particularly group harmonies and choirs. The channel EFX send should be set halfway or higher. The sweep rate is adjustable from 0.5 Hz to 30 Hz. Depth of the effect is adjustable from 0% to 100%.

FLANGE

Creates a strong sweeping effect, particularly effective on rock electric guitar, lead

and rhythm. The channel EFX send should be set halfway or higher. Sweep rate is adjustable from 0.5 Hz to 20 Hz. Depth adjustable from 0% to 100%.

PHASER

This effect is perfect for enhancing strummed acoustic guitar or electric guitar power chords. The PHASER effectively duplicates the popular 70's phase shift effect used for guitar. Sweep rate is adjustable from 0.5 Hz to 35 Hz. Depth adjustable from 50% to 100%.

BAD EFFECTS

The following effect ideas did not quite make the grade:

ELEVATOR

Each musical selection is carefully stripped of its lyrics, passed through a true mono, house band simulator and emerges as a piece even its own songwriter would not recognize or sue you for. The intermingled “bing” at each floor, will have you facing the walls, looking up at the ceiling, and avoiding all eye contact in your own living room.

DAD

When musical passages increase in volume, they reach a certain trigger point (dad-threshold) at which point, irate male vocals containing the words: “turn it down!” can be faintly heard coming from downstairs, even if you don’t have a downstairs, followed by the sound of a broom handle banging on the virtual ceiling. Additional settings including the phrases: “if I have to come up there!” and “you long haired hippy!” complete this sound effect suite.

PUNK

This will turn all of your favorite lyrics into something shouted loudly and out of key, about teenage rebellion from not being all that good at school, not having an easy job that pays good money, or not ever having a girlfriend who was very nice to you.

Modulation Effects Parameter

Tweaks—RATE, DEPTH,

WIDE Switch

When a Modulation effect is selected, the TIME/RATE knob controls the effect Rate, which is the speed of the sweeping effect. Fully counter-clockwise produces the slowest sweeps and fully clockwise produces the fastest.

The DAMPING/DEPTH knob controls the effect depth, which is the strength of the sweeping effect. Fully counter-clockwise produces the lightest sweeping effect and fully clockwise produces the thickest.

The EFX WIDE switch is very effective at dramatically increasing the strength and thickness of a modulation effect. For example, using the EFX WIDE on Chorus mimics a multi-voiced chorus effect.

The following table is a quick reference to the EMAC effects showing the range of parameters and suggested applications. Your mileage definitely will vary, but give these a try for starters.

EMAC SELECTIONS		
Name	Parameters	Typical Applications
<i>Reverbs</i>	<i>Decay/Pre-delay</i>	
REVERSE	35ms-515ms/none	Special effects on instruments or drums
GATED	35ms-515ms/none	Special effects on instruments or drums
CATHEDRAL	2sec-10sec/75ms	Wind instruments; acoustic guitar; choir
LG. HALL	1sec-5sec/75ms	Vocals; electric and acoustic guitar
MD. HALL	750ms-2.5sec/65ms	Vocals; electric and acoustic guitar
LG. PLATE	1sec-5sec/none	Vocals; snare
MD. PLATE	750ms-2.5sec/none	Tight vocals; snare
SM. ROOM	250ms-1sec/30ms	Tight vocals
SPRING	1sec-5sec/none	Acoustic guitar
<i>Delays</i>	<i>Delay/No. of Repeats</i>	
DELAY 1	5ms-524ms/1	Country, swing guitar; rockabilly vocals
DELAY 2	5ms-524ms/2	Rock, gospel vocals; acoustic guitar; wind instruments
DELAY 3	5ms-524ms/3	Slow, bluesy vocals; melodic flute
DELAY 4	5ms-524ms/4	Long vocal notes
<i>Chorus/Flange/Phase</i>	<i>Modulation/Depth</i>	
CHORUS	0.5Hz-30Hz/0%-100%	Electric, acoustic guitars; bass; choir
FLANGE	0.5Hz-20Hz/0%-100%	Rock electric guitar
PHASER	0.5Hz-35Hz/50%-100%	Strummed acoustic guitar, electric guitar power chords

Chapter 6: Mackie Technology

Mic Preamps

Our new XDR mic preamplifiers are truly world class. They're becoming the new standard of high-end mic pres. They have awesome headroom and dynamic range. They're quieter at the operating levels you use in day-to-day tracking applications. They deliver the same frequency response whether your mics are on the end of a ten-foot or a hundred-foot cable. They just plain sound fantastic—with either knock-around dynamic mics or ultra-expensive condensers.

XDR

Room ambiance so detailed you can practically hear the carpet pattern... high frequency resolution that defines cymbals, triangles, and bells down to the molecular level... midrange that's warm, fluid, and comfortable... and tight bass that's intoxicatingly rich in harmonic texture.

Spending \$1000 a channel for an "audio-ophile" mic preamplifier is worth it... because no matter how much you spend on a microphone, its ultimate performance depends on what it's plugged into and how the two interact.

Yes, many high-end outboard mic preamps can effortlessly amplify the slightest sonic nuance. They can create an aural panorama that is breathtakingly realistic, excitingly vivid, and truly 3-dimensional in scope. Although extremely expensive, outboard microphone preamps have filled a need for fidelity that wasn't possible with the "stock" mic preamps built into mixing consoles. Until now.

XDR™: Two Years of Meticulous Research and 512 Highly Subjective Arguments Later

No other company in the world has spent the sheer number of research hours on a single mic preamp design that we just did. Greg Mackie believed it was worth it because a new vastly-improved design could be used on so many different mixer models—and thus could be used by far more people than any expensive esoteric preamp.

We started with blank paper, concerned only with matching or exceeding the performance of \$1000 to \$2000-per-channel esoteric outboard preamps. By the time we were finished, we ended up re-engineering far more than just Mackie's core preamp technology.

Four Critical Criteria—Until Now, You Had Your Choice of Maybe Two at Any One Time

1. A mic preamp should be able to protect itself from destructive external forces.
2. A mic preamp should not be prone to Radio Frequency Interference (RFI).
3. A mic preamp's response should not vary with different mic/mic cable impedance loads.
4. A mic preamp should be accurate, sound good, and have very low noise and wide bandwidth at any gain level.

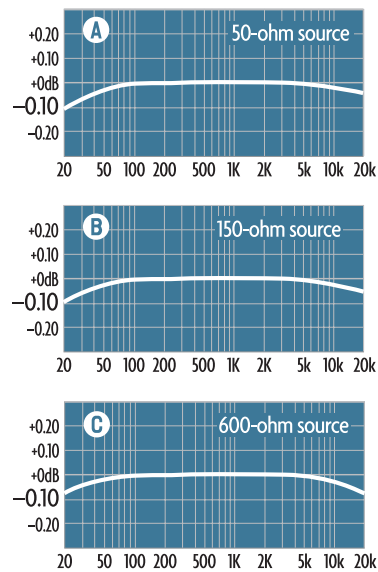
Many high-end outboard mic preamps achieve goals 3 and 4, but lack protection and good RFI shielding. Conventional mixer preamps often meet goals 1 and 2 at the expense of good sound¹. XDR represents major advances in all four critical areas.

Protection Against "Hot Patching" and Short Circuit Damage

Slow degradation (and eventual failure) of mic preamps resulting from hot patching is one of the mixing console manufacturing community's dirty little secrets. If you route a phantom-powered mic through a patch-bay, it's roughly equivalent to shorting out a cable every time you patch it. The mic preamp's input transistors progressively break down (called zenering), while the mixer channel gets noisy and can eventually totally croak.

Mackie engineers weren't interested in any "solution" that even slightly affected the preamp's sound. A lot of research and considerable old-fashioned trial and error listening resulted in the addition of ultra-high-speed, large-geometry input diodes to the front end of our input circuit. They completely protect XDR mic preamplifiers from the consequences of hot-patching and direct short-circuits in cables carrying phan-

tom power. Yet this has no effect on sound quality.



Rejecting Radio Frequency Interference Without Tuning Out Sound Quality



This little bitty thing, discussed below, is the bifilar-wound dc pile transformer that makes the huge difference in XDR sound quality.

A mic preamp has to be able to amplify faint signals lower than 1 microvolt² to as high as 10 volts and have sufficient bandwidth to insure absolute accuracy of reproduction. With a voltage gain of one thousand (60 dB),

any nonlinearity in the preamplifier circuitry could serve as an unwanted radio receiver. (Propeller-heads refer to this process as nonlinear signal rectification.) The more capable of conveying fine sonic details and wide bandwidth a mic preamp is, the more prone it is to picking up the nearest trucker's CB or All Sports Talk Radio station.

We assaulted RFI on three fronts. First, we incorporated custom bifilar-wound dc-pulse transformers with high-permeability cores that reject RFI but don't attenuate the sound. Second, we used high-precision components for critical areas of the XDR preamplifier. Third, we direct-coupled the circuit from input to output and used pole-zero-cancellation constant-current biasing (which also avoids increased intermodulation distortion at high common-mode signal levels).

Bottom line for the nontechnical: you can use our new XDR mic preamps at the end of extremely long mic cables in an RFI-saturated urban environment while talking on

your cell phone and microwaving coffee... without hearing a trace of RFI.

Input Equality with Controlled Interface Input Impedance

A microphone's frequency response (and thus how it sounds) is partly a function of the load presented to it. That load is the impedance characteristic of the mic preamp it's connected to. If a mic preamp isn't designed right, it will actually sound different depending on the impedance of the microphone and the cable load³!

What we've done with the XDR design is create a Controlled Interface Input Impedance system that's able to accept an enormous range of impedances without compromising frequency response. Figures A, B, and C (left) show XDR frequency response with 50-ohm, 150-ohm, and 600-ohm source impedances. Note that the charts are calibrated in TENTHS of a dB. As you can see, at any load, frequency response is virtually unaffected down to just one tenth of a dB at 20 Hz and 20 kHz no matter what the load.

Extended Dynamic Range and More

We call our new design "XDR" because it has over 130 dB of total dynamic range. That means the XDR mic preamps on our VLZ PRO mixers can handle inputs from 24-bit, 192 kHz sampling-rate digital audio workstations without added coloration. It means you can generate 145 dBA sound pressure levels at the diaphragm of a suitable microphone and get a signal through the preamp that doesn't break up.

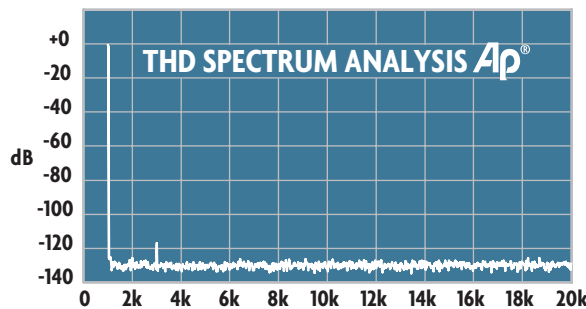
But wide dynamic range is valuable only if the noise floor stays low at normal gain levels.

Ultra-low Noise at "Real World" Gain Settings

Equivalent Input Noise (E.I.N.) is a common mic preamp noise spec. But this oft-touted specification is traditionally measured at maximum gain levels only. Normal operating levels are most often between +20 and +30 dB of gain. Thus many mic preamps that can brag about their -129 dBm E.I.N. at +60 dB generate far worse noise performance at +30 dB where you're going to end up using them much of the time.

Graph D, below, charts E.I.N. versus gain level for two mic preamps that boast

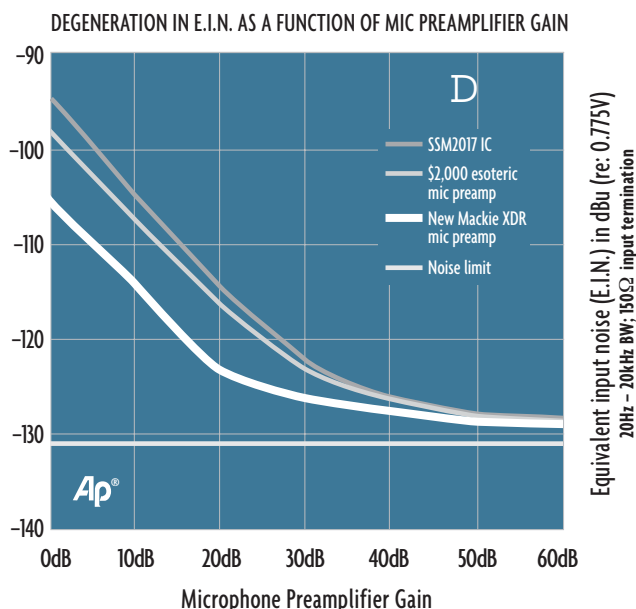
129 dBu at a maximum gain of 60 dB. One is our XDR, the other is the SSM2017 Mic Preamp IC used in many outboard mic preamps and \$20,000 (and up) consoles. As you can see, our XDR design maintains lower noise levels in the critical 0 to +30 dB gain range than the IC or other competitors. In fact, the noise is typically half that of the most expensive units found on this planet. The XDR can handle signal inputs from a full +22 dB line level (thanks to extended dynamic range) down to microvolts without a corresponding increase in noise or distortion.



Other Characteristics of \$1000 Mic Preamps Now Available on Affordable New VLZ PRO Mic/Line Mixers

While specifications are only one measure of how a microphone preamplifier sounds, XDR matches esoteric outboard designs number for number:

0.0007% Total Harmonic Distortion. A THD figure this low has never before been offered on affordable mic/line mixers (and only rarely on \$500,000 mega consoles, for that matter).



Near “dc-to-Light” bandwidth—without RFI side-effects. Not only are XDR mic preamps flat within a tenth of a dB across the bandwidth of any known microphone, but they’re also only 3 dB down at an astonishing 192 kHz, a figure that should satisfy anyone in search of elusive upper harmonics soon available with emerging ultra-high sampling rates.

Super-low intermodulation distortion at high operating levels. Or to put it another way... total independence and stable operation no matter what signal level you send it. This is due to our instrumentation-style balanced differential architecture, linear biasing, and use of dc-coupled pole-zero-cancellation constant current that frees the mic preamp from induced dynamic intermodulation distortion.

Hear XDR and Decide for Yourself

We can sling Audio Precision® graphs around 'til the cows come home and still not convince the truly critical listener. When it comes to mic preamps, the proof is in the listening. Plug a really, really good condenser microphone into an XDR preamp and do some critical auditioning.

Notice the highly-focused high end. The superior ambience retrieval. The substantial, detailed lower bass octaves. Hear textural midrange details you never knew the microphone could reproduce.

Above all, hear this kind of mic preamplifier performance on truly affordable mic/line mixers for the very first time.

Now shipping from Mackie Designs.

SELECTED XDR MIC PREAMP SPECIFICATIONS	
EIN 20 Hz-20 KHz, source Z:	
150 ohms	-129.5 dBu
50 ohms	-131.0 dBu
0 ohm	-134.5 dBu
Frequency Response:	
-1 dB	5 Hz-100 KHz
3 dB	3 Hz-192 KHz
IM Distortion 4 to 1 ratio SMPTE 35 dB gain	0.0008%
Harmonic Distortion 20 Hz-20 kHz 35 dB gain*	0.0007%
Gain	Max +60 dB Min, 0 dB or Unity
Max. Input	+22 dBu

*typical gain structure

1. One really REALLY expensive mega-console (with mic preamps nobody particularly likes) “fights” RFI with a humongous capacitor across two pins of its mic inputs. RFI stays out but high frequency response is seriously affected, resulting in a rather dull sound.
2. Slightly less than the total neural voltage of an elderly Chihuahua but slightly more than that of a typical game show hostess.
3. Cable capacitance can add up, especially on long runs down a snake to a remote truck or front-of-house console.

VLZ Circuitry

An incredibly high percentage of Mackie owners rave about how quiet our mixers are. We accomplish it with premium components and Very Low Impedance (VLZ™) circuitry.

Developed for our acclaimed 8•Bus console series.

VLZ is a unique (and, frankly, expensive) approach to circuit design that until now has been found only in money-is-no-object consoles in the \$200,000-and-up price range. Now you can enjoy its benefits on our 1604-VLZ PRO, 1642-VLZ PRO, 1202-VLZ PRO, 1402-VLZ PRO, SR24•4-VLZ PRO, and SR32•8-VLZ PRO.

VLZ wouldn't be necessary if we did all our recording in outer space or other places where it's extremely cold. Near absolute zero (-273°C), circuit components are noise-free because the atoms that comprise them are completely at rest. However, at over three-hundred degrees hotter (room temperature), all the atoms in circuit components are agitated and constantly crash into each other. That causes little random voltage spikes that create thermal (or white) noise (non-electronic types call it hiss).

Mackie deals a crushing blow to thermal noise by making internal impedances as low as practical, at as many places as possible within the mixer. Very Low Impedance (a.k.a. Ω , a.k.a. Z) is achieved by scaling down resistor values by a factor of three or four—resulting in a corresponding reduction in thermal noise.

That alone isn't the expensive part. What's expensive is the high current necessary to use low impedance. You don't have to be

an electrical engineer to realize that high current requires lotsa power—whether it's current for an amp, a motor, or a mixer circuit.

And that's where our mixers excel, thanks to their robust internal power supplies delivering current levels that are impossible with wimpy external “wall warts.” Major power lets them achieve very low impedance in critical circuit areas. This means astonishingly low noise levels—and as an added bonus, far less crosstalk between mixer channels.

EMAC™

Built-in Digital Effects That Sound Real!

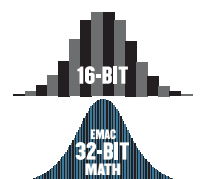
The Story Behind the 32-bit Digital Effects Featured in Our DFX, CFX, & PPM Series Mixers

What's the point of onboard effects if they sound like tin cans on the end of boingy metal springs? We wouldn't offer effects mixers if we didn't think that we could provide serious studio-quality digital effects. And we did. Effects that blow our competition into the ether.

For that, you have our Digital Group to thank. While creating the world's best 32-bit processing digital recording console, they developed a set of core effects based on the same algorithms used in some of the world's most expensive and prestigious outboard signal processors. These effects had to be very good.

The key lies in doubling the bit rate to accommodate far more complex (and realistic) effects processing. To sound authentic, digital effects such as reverberation and delay require incredibly complex arithmetic calculations. This requires both processing power and a wide data path.

Without getting too far into computer-geeky stuff, let's just say that 32-bit processing can crunch far more complex calculations than the 16-bit processing that our competitors' effects mixers use. There's no loss of internal precision (or detail) during processing. The result is stunningly realistic effects that we call EMAC,

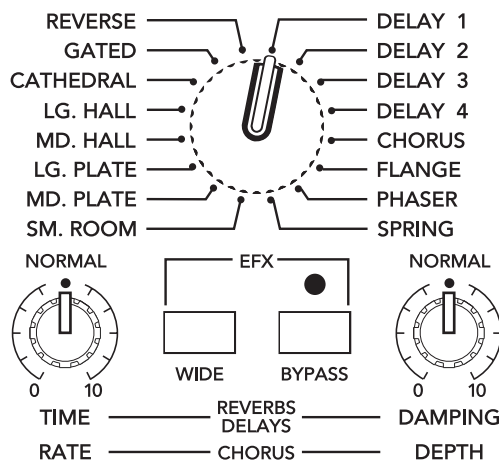


for Enhanced Multiplication and Accumulation.

We also added two parameters to control each effect. They let you vary things like delay time and damping on reverbs, or depth and rate on chorus and flange. In other words, you don't just get 16 different high quality effects, you get an almost infinite number of variations to suit your own tastes. The supercool EFX Wide switch acts like a spatial enhancer on the reverbs, chorus, and flange. It gives you fuller reverbs for small, dead rooms—and delicious, thick chorus and flange sounds for acoustic guitars and background vocals when you need them.

Sure, we can make wild claims about how good these effects are. But the bottom line'll be when you hear them. Compare EMAC™ effects on the DFX, CFX and PPM Series with those in competitors' mixers. You'll be impressed with our technology.

Spin the EMAC™ Effects Dial



Reverse Reverb. that simulates a tail-first effect increasing to the original note. Decay range adjustable from 35ms to 515ms. (No pre-delay.)

Gated reverb. The reverb tail is cut off sharply after the preset decay length (variable from 35ms to 515ms).

Cathedral. A dense, smooth reverb with very long tail, long 75ms pre-delay and late reflections. A dramatic effect that works well with wind instruments, slow finger-picked acoustic guitar, and quiet group vocal harmonies. Decay adjustable from 2 to 10 seconds.

Large Hall. A slightly “smaller” version of Cathedral that sounds great on vocals, electric and acoustic guitar. One to 5-second decay range, 75ms pre-delay.

Medium Hall. The most “compact” of the environment reverbs with normal tail, 65 ms pre-delay and increased early reflections. 750ms to 2.5-second decay range.

Large Plate. Good early reflections and no pre-delay. Tails are normal and warm with strong high-end for increased presence—perfect for vocals and snare drums. One to 5-second delay.

Medium Plate. A “smaller” version of Large Plate with short warm tails and strong high-end that works well with tighter vocals and snare. 750ms to 2.5-second decay.

Small Room. A very fast, scattered reverb with diffused early reflections and a short, 30ms pre-delay. Tails are very short and warm to imitate absorbent wall materials and the presence of an audience. 250ms to 1 second decay range.

Delay 1. One repeat for slapback delay used in country, swing, and rockabilly. Five to 524ms delay range.

Delay 2. Two repeats for a fuller, more dramatic effect on rock and gospel vocals, acoustic guitar, and wind instruments. Five to 524ms delay range.

Delay 3. Three repeats repeats repeats for slow bluesy vocals and melodic wind music. Five to 524ms delay range.

Delay 4. Guess how many repeats this one has? Used for occasional very dramatic effects in long vocal notes and instrumental note chopping. Five to 524ms delay range.

Chorus. A soft, ethereal sweeping effect. Adds dramatic effects to vocals (particularly group harmonies), electric and acoustic guitar, and bass. Chorus rate is adjustable from 0.5 Hz to 30 Hz; depth is adjustable from 0% to 100%.

Flange. That somewhat indescribably sweeping stereo effect that sounds so cool on electric guitar. Somewhere between psychedelic and merely trippy. Rate adjustable from 0.5 Hz to 20 Hz; 0% to 100% depth adjustment.

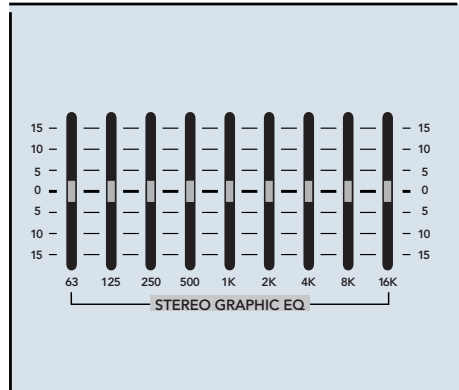
Phaser. The classic early 70's phase shift effect. Sounds great on strummed acoustic guitar or electric guitar power chords. Certifiably psychedelic; appropriately mono. Rate adjustable from 0.5 Hz to 35 Hz; 50% to 100% depth adjustment.

Spring. Mimics that vintage 60's-style, “wet” spring reverb made famous by surf guitarists. Tails are normal with a strong

high end and slight waver to imitate the slow waver of a mechanical spring system. Great on acoustical instruments.

And remember folks, besides Time/Rate and Damping/Depth adjustments, all effects except Delay and Phaser can be further modified with the EFX Wide button found on the CFX and PPM Series mixers. EFX Wide acts like a spatial enhancer for wider reverbs, multi-voice-like chorus, and phat flange.

Distortion-free Graphic Equalizers on Board, Too



If a graphic equalizer is made with cheap-off-the-shelf integrated circuits, it can overload, add noise, and create audible phase distortion - even when the controls are set at zero. And since the graphic EQ is always in the final output signal path of effects and powered mixers, your whole sound can suffer.

Our DFX, CFX, and PPM Series mixers have graphic equalizers made with more discrete parts, including multiple-summing amps and precision infinite-feedback band-pass filters for superior combining properties and linear phase. Yeah, that's a technical mouthful. But the result is a better-sounding graphic equalizer that doesn't screw up the overall sound of the mixer.

FR Series Fast Recovery Power Amplifiers

- 800-, 1400-, and 2600-watt models
- High continuous current output
- Fast Recovery design for lower distortion at clipping
- Built-in Low Cut filters
- Switchable limiters
- Extra system-enhancing features on each model
- XLR and TRS inputs with XLR signal pass-through

- 5-way binding posts and TRS or Speakon® outputs
- T-Design cooling with front inlet and side outlets
- Detented gain control calibrated in volts and dB
- Output level LED ladder, Signal Present, and OL LEDs
- Massive toroidal transformers



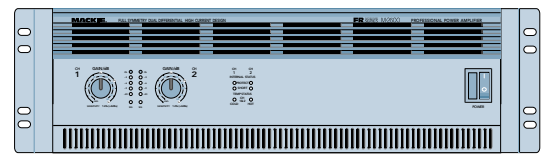
M•800



M•1400



M•1400i

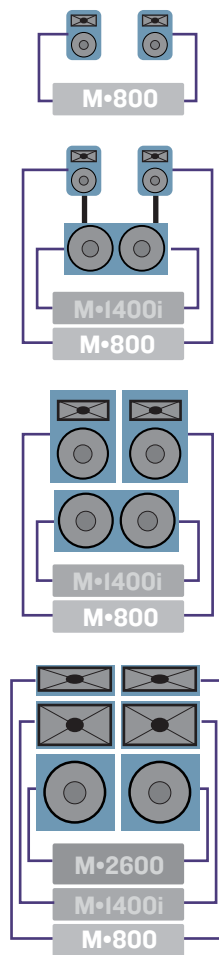


M•2600

Brute Power, Clean Sound, and Unique Features in Three Convenient Sizes

When we introduced our FR Series™ High-Current Power Amplifiers, some folks went, "Huh? Power amps from Mackie?" After all, we've mostly been known for our precision studio-quality mixers.

Four years later, the “Huh?”s have turned to “WOW!”s. FR Series power amps are everywhere! And with good reason. They’ve racked up a strong record of reliability in demanding sound reinforcement applications. And tens of thousands of satisfied owners have come to appreciate the sound quality of the FR Series and its extra system-enhancing features. Now these features have proliferated inside our powered mixers, active sound reinforcement speakers, and our industry-standard HR824 and HR624 near-field studio monitors.



No Snake Oil— Just Solid Engineering

FR Series power amps don’t pull power out of the ozone or switch on and off a gazillion times a second. They’re traditional “lead sled” designs with massive toroidal transformers, huge storage capacitors, state-of-the-art output devices, and cooling systems that outperform all amps in their price range. They deliver enormous amounts of current into extremely low impedances—and they have dynamic power reserves to spare. The reason FR Series amps are so affordable is because of Mackie’s huge investment in automated manufacturing technology—not because we cut any corners.

FR Stands for Fast Recovery

Cutting to the chase: FR power amps sound better than their competition when driven into clipping (overload). Since that’s how 99 out of every hundred PA systems are usually operated, this is a major point¹.

When an amp clips, you start hearing harmonic distortion. So conventional amps use something called negative feedback to help control clipping distortion. The out-

put section of the amp sends a signal back to earlier stages to “throttle the system down.” The trouble is, this “solution” causes its own problem: During clipping, when the amp relies on negative feedback, it electronically “confuses” earlier amp stages, causing them to latch in a clipping mode instead of recovering quickly. Latching causes the amp to sound really awful when pushed too hard—which it inevitably will be.

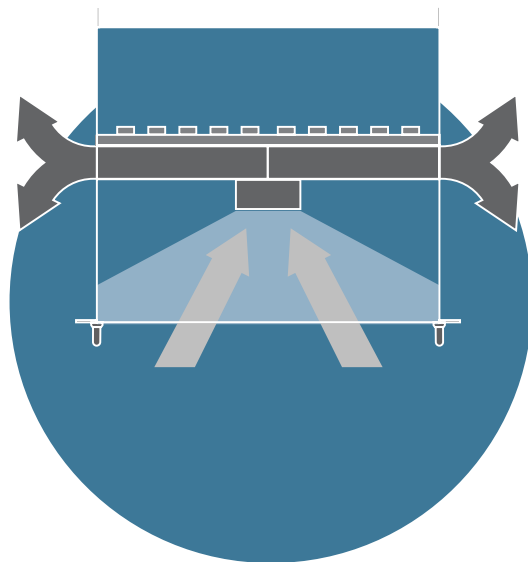
To effectively deal with clipping, an amp must be able to recoup almost instantaneously. That’s what we mean by Fast Recovery. FR Series amps use very sparing amounts of negative feedback. Then we use complementary Baker Clamp circuits on the positive and negative voltage amp stages, which prevent the stages from saturating (and latching) during periods of overdrive. An additional transistor senses when the Baker Clamp is active and then activates the FR Series amplifiers’ internal limiting circuits. The result is no latching, instant recovery from overdriving the amp, and way better sound.

Better Cooling—Better Reliability

The more heat you can conduct away from an amp’s output devices (transistors) the longer they’ll last. The better physical contact that can be made between a transistor and its heat exchanger, the more heat can be pulled away and dissipated in the air.

The worst cooling solution is found in cheapo amps that just have a fan on the front or back. It blows through the inside of the chassis and across the output devices. Not only is this inefficient (leading to premature transistor failure), but it also ends up coating the whole inside of the amp with dust and crud.

Better amps place their output devices on a metal heat exchanger. Fan air is channeled air down a long passage over the cooling fins. Trouble is, the air heats up as it moves down the long channel. The output devices at the end get hot air blown over them. So they tend to fail first.



All four FR Series amplifier models have two short fan tunnels instead of one long one.

We start with a mirror-polished heat exchanger that maximizes thermal transfer. This requires hand polishing the exchanger extrusion, but we think it's worth it to achieve better reliability.

The exchanger itself is transverse-mounted with its air inlet in the middle. Cool room air enters the front of the amp, then travels down two short cooling tunnels and exits on the amplifier's sides. This T-Tunnel design provides a more constant thermal gradient than one long cooling tunnel—in other words, two short air passages keep all transistors closer to the same temperature instead of getting progressively hotter as the air temperature increases down one long exchanger.

The result is enhanced amplifier reliability under demanding conditions like hot clubs or outdoor concerts on sweltering summer days.

Extra Sound-enhancing Features

A big advantage to owning an FR Series amp is that you get important extra features². Features other amp manufacturers make you pay extra for or leave out all together.

Variable low cut filters for tighter bass. All PA cabinets only reproduce bass down to a certain point, called the tuned frequency. Below that point, you get audio sludge and potential woofer damage. FR Series variable low-cut filters let you feed your system only the frequencies it can handle. You can “dial in” any tuned frequency from 10 Hz to 170 Hz. All models also include an infra-

sonic stabilizer circuit that cuts the really low, inaudible frequencies that cause distortion-inducing -visible woofer cone flutter.

Constant Directivity Horn EQ for smoother treble (N/A on M•2600). To improve high frequency reproduction, many sound systems use constant directivity (CD) horns. They improve treble dispersion by distributing high frequencies more evenly. But in doing so, they actually create a frequency “dip” that reduces important frequencies anywhere from 2.5 kHz to 5 kHz. The old way to compensate for this was a special crossover module (hard to find and usually expensive) or a graphic equalizer (nice try, but the “cure” is worse than the problem). The M•800, M•1400, and M•1400i include separate left and right CD horn EQ with variable frequency. This 6 dB/octave EQ's “knee” position is sweepable from 2 kHz to 6 kHz—we extended the high-end boost so you can add “Air” EQ even if you're not using CD horns.

Built-in subwoofer crossovers. The M•1400 and M•1400i feature 2nd order, 12 dB/octave, -Bessel electronic crossovers with two selectable frequencies. The M•2600 features 4th order, 24 dB/octave, Linkwitz-Riley electronic subwoofer crossovers with three selectable frequencies. All have uniform time-delay low-pass filters. Just select the crossover frequency, connect the subwoofer output to another amp—and drive that subwoofer all over the map.

	M-800	M-1400	M-1400i	M-2600
4-ohm bridged	800W	1400W	1400W	2600W
8-ohm bridged	550W	1000W	1000W	1700W
2-ohm load	400•400W	700•700W	700•700W	1300•1300W
4-ohm load	275•275W	500•500W	500•500W	850•850W
Low-cut filters	yes	yes	yes	yes
Limiter circuit	yes	yes	yes	yes
CD Horn EQ	yes	yes	yes	—
Subwoofer crossover	—	63Hz, 125Hz	63Hz, 125Hz	60Hz, 90Hz, 120Hz
XLR pass-through	yes	yes	yes	yes
5-way binding posts	yes	yes	yes	yes
Neutrik® Speakons®	—	yes	—	yes
1/4" TS outputs	yes	—	yes	—
230V & 100V models	yes	yes	—	yes
T-Design cooling	yes	yes	yes	yes
Demand-sensitive fan	yes	yes	yes	yes
Detented level controls	yes	yes	yes	yes
LED output display	yes	yes	yes	yes
Signal present, OL and Channel Status LED	yes	yes	yes	yes
Temp & Short Circuit LED	yes (Temp LEDs only)	yes	yes	yes

Loaded with Extra Stuff

In typical Mackie fashion, we've added all sorts of handy extras to FR Series amplifiers.

All models include both XLR and TRS inputs. Even better, we've added an extra male XLR signal pass-through. Besides letting you "daisy-chain" amps, it comes in very handy during set-up, since you can plug in either "gender" of XLR instead of having to go find another cable.

All four FR amplifiers have integral servo-controlled limiters that you can use to coax more SPLs out of a sound system—or defeat via back panel switch if you don't believe in limits.

All models have elaborate but normally inaudible protection circuits and automatic turn-on delay so you don't stress out fuse boxes or circuit breakers when you power up a whole stack of amps.

Instead of just a single power "blinkie," you get a full-blown stereo LED ladder display that closely monitors power output. Plus signal present LED and thermal overload LED and a lighted rocker switch.

Individual left/right, detented level controls are calibrated in both volts and dB.

Stereo/mono/bridge amp mode select is done with a simple switch instead of a weird wiring configuration.

Finally, all FR Series amps will operate on as little as 63% of their rated voltage. This is a major plus when you have to work at the end of a long, skinny extension cord (which reduces voltage), or during brown-outs.

Collect 'Em—Trade 'Em With Your Friends

Whether you run your whole sound -reinforcement system with a single amplifier or have rack upon rack of them backstage, there's an FR Series amp tailored for your application.

They offer you more system-enhancing features than cheaper amps. They deliver better performance and better reliability than comparably-priced amplifiers.

Finally, the M•800, M•1400, M•1400i, and M•2600 just plain sound good, even when they're cranked up to past clipping. Bass is tight and controlled. Mids are free from fatigues and harshness. And high frequencies are crystalline and detailed.

Hear the FR Series power amplifiers at your Mackie dealer right now. We think you'll agree that a mixer company really can make kick-butt amps.

FR FAST RECOVERY SERIES

BACK PANEL	
1	1/4" TS outputs
2	Neutrik® Speakon® outputs
3	5-way binding post outputs
4	1/4" TRS inputs (balanced/unbalanced)
5	Female XLR inputs (bal./unbal.)
6	Male XLR input or signal pass-thru (balanced/unbal.)
7	CD horn EQ control. 2kHz to 6kHz (N/A on M•2600)
8	CD horn EQ bypass switch (N/A on M•2600)
9	Variable low-cut filter. 5Hz to 170Hz
10	Amp mode switch (stereo/mono/bridged)
11	Limiter defeat
12	Subwoofer frequency switch. 125Hz/63Hz on M•1400 & M•1400i; 120Hz/90Hz/60Hz on M•2600.

Components and Construction

Steel Chassis

Rugged steel main chassis are standard on all Mackie equipment.

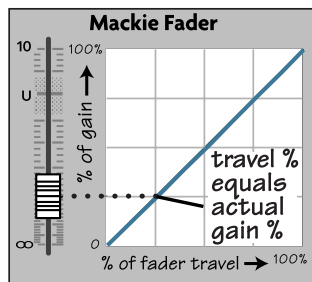
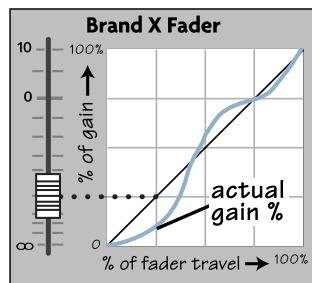
Circuit Boards

We use only military grade epoxy and extensive surface-mounted components on all our circuit boards.

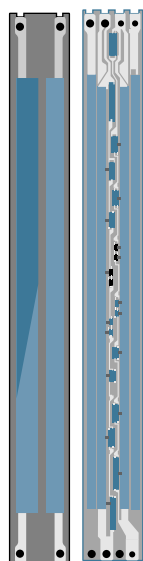
Internal Power Supplies

Power supplies on all Mackie equipment are built in, making them integral to the units.

Log-Taper Faders



Conventional faders can make accurate level control hard... and smooth fades nearly impossible (figure at left). About halfway down, the sound just goes away. It's the fault of the simple, cost-cutting internal construction of D-taper faders (see leftmost figure below).



The faders on all of our mixers feature a precision log-taper design. A complex series of additional resistive elements are screened onto the wiper's path (see rightmost figure).

They produce a precise, logarithmic taper that gives you amazingly smooth, consistent fades that extend the full length of the control (see first figure above).

There's more. The 1604-VLZ PRO, 1642-VLZ PRO, 1402-VLZ PRO, and SR Series console faders use a completely new internal wiper design that incorporates a special long-wearing contact material orig-

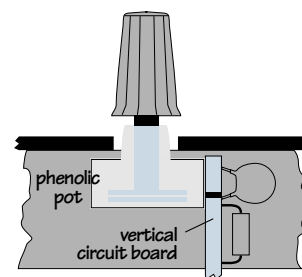
inally developed for weatherproof exterior sensors. You get longer fader life plus improved resistance to dust and other miscreants that inevitably weasel their way past even the tightest lip seal.

At Mackie, we know that faders are the first thing you reach for, so we make sure they are the best. Go move one for yourself.

Sealed Rotary Pots

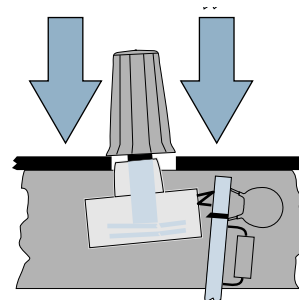
Days after the 1994 Los Angeles earthquake, Mackie Designs mixers were dubbed "audio throw pillows" by San Fernando Valley studio owners who had begun the painful process of cleaning up. Why? Because so many of our mixers had taken the full impact of falling monitor speakers without so much as a broken knob. Several were even buried by collapsing ceilings, only to be unearthed, tested, and declared to be in working order. This is good to know when you're hauling your mixer out into the field where anything and everything can happen to it.

Scrunch! @#%&*\$%! Many other mixers make three mechanical design mistakes that can lead to expensive repair bills. First, they use unsealed potentiometers containing brittle phenolic as a base for the potentiometer's wiper. Next, they mount the pots on vertical circuit boards. Finally, many mixers have pots whose bushings extend through chassis holes (see figure below).



This is a deadly combination when something heavy drops on the mixer. When downward force is

exerted on this kind of knob, all of the shear force is transmitted to the control and circuit board solder connection (see figure below).



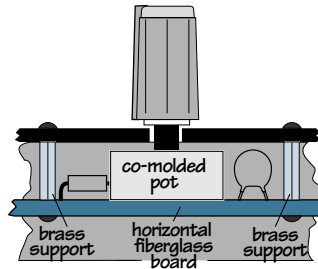
Even if the phenolic material inside the pot doesn't crack, the whole thing can still get yanked off the vertical circuit board or break electrical con-

tact. If a number of knobs get bashed from above at one time, the repair bill can be hefty.

Our mixers head off all three design errors.

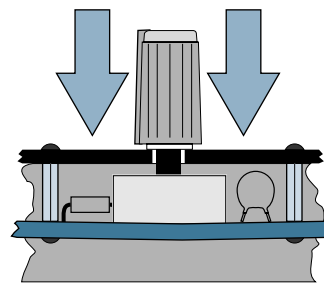
Mackie's first line of defense against external damage is right there, in the knobs. They're designed to "ride" just thousandths of an inch from the metal surface of the mixer chassis (see figure below).

When downward force hits, it's transferred from knob to steel—from knob to control.



Next, we employ a co-molded potentiometer that doesn't use brittle material at critical mechanical points. If the mixer really gets whacked, our pots withstand far more abuse than regular pots.

Finally, to eliminate force transferred to the circuit board, we use a braced horizontal circuit board and shock-absorbing structure (see figure below).



The board is thick fiberglass and is connected at regular points to the chassis by metal standoffs. Ultimately, extremely brutal

knob impact is absorbed by broad pressure on a tough circuit board instead of acute stress at a few solder joints that weren't meant to take physical abuse.

Section 2: Applications

Section 2 Table of Contents

Chapter 7: Suggested Hookups and Applications

At Mackie, we've designed our mixers to fit a wide range of applications. In this section, we'll show you some potential applications for each of the models together with typical hookup diagrams and a "how to do it" description. These knob-by-knob descriptions will be helpful if you're a beginner. Some highlight unique features of specific Mackie mixers. You'll find them worth reading.

Many of the hookups and applications in this section are applicable to any mixer, so if you don't see your application for your mixer, see if there's a similar one for another model. It'll get you started.

Generalities and Substitutions

While you may not have every component on the hookup diagram that most closely matches your application, you can apply these general principles.

Recorders

Our diagrams show a variety of recording devices—stereo and multitrack, analog and digital, tape and disk. Used to be in the old analog days that there was an input and an output for every track. Today we have computer audio interfaces (that's the two-dollar word for "sound card") that have, for example, two inputs and eight outputs. When the budget's tight and you don't need to record more than two tracks at a time, one can substitute nicely for an 8-track recorder as long as you understand the limitations.

When it comes to the two-track mixdown recorder, we scatter DAT and cassette decks around our diagrams with reckless abandon, but there's no reason you can't mix to a computer with a sound card, a stand-alone CD recorder, or analog reel-to-reel.

Similarly, wherever you see a cassette or CD player as a playback source, you can substitute any stereo source.

Operating Levels and Impedance

With the exception of instrument pickups, loudspeakers, and the occasional vintage

signal processor salvaged from a broadcast studio, there's little reason to worry about impedance matching when making connections between units. What deserves some attention, however, is matching of operating levels, and recognizing the impact of differences.

For many years there have been two de facto standards for output voltage and input sensitivity that fall under the broad classification of "pro" and "semi-pro". They usually use different types of connectors, but that's just a physical thing. What's important to understand is that a "pro" input may require more signal voltage than a "semi-pro" output is capable of delivering. Mackie mixers are designed to cover the full range of operating levels, but sometimes you have to turn the knobs off their unity gain positions in order to match up with other equipment. There's nothing wrong with that, it's why we put them there.

Loudspeakers

Traditionally, speakers are passive devices, requiring an external power amplifier. The PPM series mixers have power amplifiers built in, but you'll need power to drive speakers from any of our other mixers.

Active loudspeakers with built-in amplifiers are becoming mighty popular. Mackie makes some of the best, both for studio monitoring and sound reinforcement. It's a great idea—an amplifier and speaker designed and built as an integral unit assures the ideal matching between those two components. The length of wire connecting the two, where most power loss occurs, is absolutely minimized. They also save space and make hookup easier. When you see a hookup diagram showing a speaker with an amplifier between it and the mixer, you can always replace the combination with a powered speaker.

We're often asked if you can use your home stereo system for monitoring and playback in your studio. Sure you can—just connect the mixer output (as shown on the diagram) to a Line Input on your stereo.



Beware, though, that the records you listen to at home usually have very limited dynamic range, whereas your own recordings, at least dur-

ing the tracking process, may not be so well constrained. It's easy to blow a home stereo speaker when playing back a mix at fairly high volume. Don't say we didn't warn you.

Hooking It All Up

The following section contains a wide assortment of applications with enough information to get you started working. They are by no means the only way to accomplish the goal, but by following these guidelines, you'll be able to hear something, start working, and explore other possibilities and variations.

Unless you do something that's physically difficult like connect a microphone to a power amplifier output or phantom power a synthesizer output, or connect your mixer to the wrong power line voltage, you won't blow anything up. The worst that will happen is that you won't hear what you expected.

Don't be afraid to experiment, and until you get very comfortable with all the gozintas and gozoutas, take notes and draw your own diagrams of what's working for you so that you can always get back to normal after you've made a change. Engineering is all about adapting and interconnecting your equipment to get a certain task accomplished. You have a good start with the mixer you've purchased. Use it intelligently and it won't let you down.

Now get out those cables and start plugging things in.

Chapter 8: 8-Track Recording Applications– I604VLZ-PRO, I642VLZ-PRO

These true 4-Bus mixers are well suited for 8-track recording applications. They're compact enough to fit on a desktop and flexible enough to provide real pro studio features. We'll describe a multitrack recording setup with a I604VLZ-PRO and point out the few differences when using a I642VLZ-PRO in this application.

Multitrack Split Monitoring

This application illustrates the concept of Split Monitoring. The lower numbered channels of the mixer (1-8) are used to send signals to the recorder, while the higher numbered channels (9-16) are dedicated to monitoring the outputs of the multitrack recorder. Input channels 1-8 are assigned as needed to get the input signal to the desired track on the recorder. The outputs of the "monitoring" channels 9-16 are assigned to the MAIN L-R busses.

Recorded tracks, as well as anything routed to the recorder when the recorder is in the Input Monitor mode (which sends its inputs straight out its outputs) will be heard when you push up the faders on Channels 9-16.

Since the recorder returns are always assigned to the MAIN mix, you can always hear what you're recording and what you've previously recorded.



When a microphone channel is live (assigned to a tape track and not muted) *and* you're monitoring that tape track on the control room speakers, you can get feedback. When recording from a mic in the control room, kill the signal to the CR speakers by turning the CR volume down or switching the amplifier off.

Recorder Inputs

In this hookup, recorder inputs for Tracks 1-4 are fed from SUB OUT 1-4, Tracks 5-8 are fed from DIRECT OUT 5-8, the channels to which the keyboard and drum machine are connected. When you're recording, Channels 1-8 are assigned either to sub-groups or nothing, depending on what you're recording at the time. You'll only

assign them to L-R if you're rehearsing and what to hear something without the recorder being turned on.

The mixdown (Cassette/DAT) recorder is connected to the RCA TAPE OUT jacks which deliver the MAIN L/R mix at a level of 0 dBV. If you have a mixdown recorder that operates at "pro" levels, connect it instead to the balanced or balanced./unbalanced MAIN stereo outputs. It's handy to have the mixdown recorder outputs connected to the TAPE IN jacks, however, even if an adapter is required. That way, you can listen to your mix by simply selecting TAPE as the Control Room Source.

An equalizer and compressor are plugged into the MAIN INSERT jacks. This allows you to do some pre-mastering, touching up the final frequency balance and reducing the dynamic range so that all the elements of your mix can be heard even in a noisy car. We suggest that you only connect this final compressor and/or EQ when you're ready to do your final mix. Their action could be misleading while you're tracking.

Effects

We're feeding a "parallel path" reverb from AUX SEND 4 and returning its output to the mix in stereo on STEREO AUX RETURN 2. A digital delay is fed from AUX SEND 3 and returns on AUX RETURN 1. If you have a mono delay, connecting its output to the left RETURN channel centers the delayed signal in the stereo mix.

What's wrong with using SENDs 1 and 2? Nothing, and when you have a rack full of effects at your disposal, there are times when you'll do so, but there's a method to this madness.

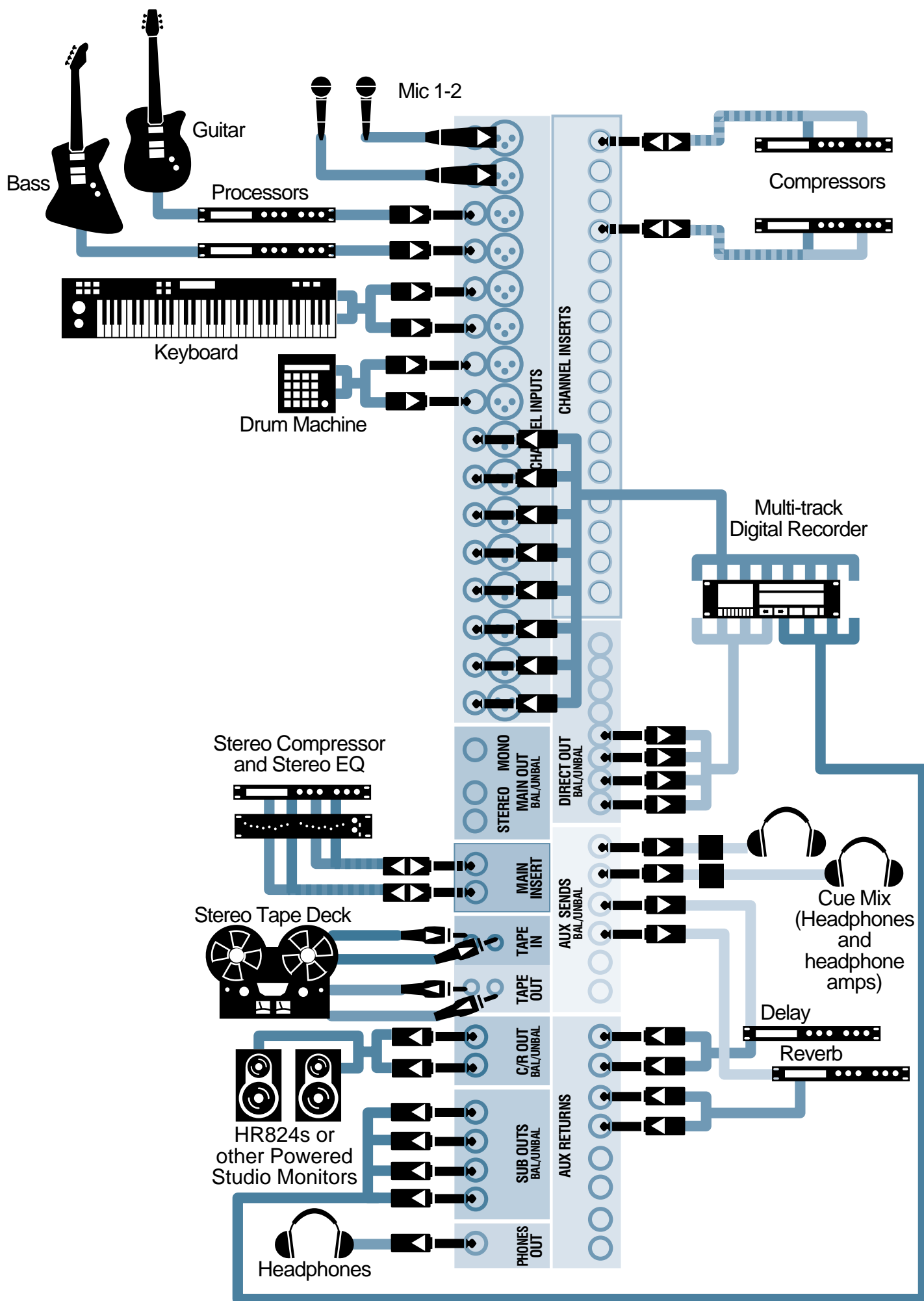
We want to send a post-fader signal to our effects, and AUX 3 and 4 are always post-fader. Chances are you'll be reaching for the Reverb send knob more frequently than the Delay knob, so we've made the bottom knob of the group the Reverb. That way it will be easier to find. It's attention to these little details that can help you to avoid careless errors after a looooooong mixing session.

We've chosen to bring the two effects back into the mix on RETURNS 1 and 2 so if we choose to create a separate headphone cue mix on AUX 1 or 2 (rather than use the PHONES jack), we can add effects to the headphone mix as well as the main (Control Room) mix.

Feeding the Recorder

In this setup, we're using two different ways to feed signals to the recorder. Tracks 1-4 are fed from the four subgroup outputs. By using the ASSIGN switches and PAN pots, those tracks can receive a signal individually or in odd-even pairs from any input channel. We've dedicated Tracks 5-8 to what's plugged into mixer Channels 5-8 (keyboard and drum machine) by connecting those recorder inputs to the DIRECT OUTs for Channels 5-8.

There's nothing sacred about this layout, but we had to pick something to talk about. Adapt it to your own choice of instruments and tracks.



Making Tracks

First we'll describe the most straightforward procedure using the system as shown in the diagram—eight sources, eight tracks. Then we'll do another take and show you some studio tricks. First, to avoid an inadvertent feedback path while you're getting things organized, pull the faders on Channels 1-8 down fully.

Ready

The keyboard and drum machine are already routed directly to tracks. To record the vocals, guitar and bass, you must first ASSIGN those channels to SUB OUTs 1-4 using the ASSIGN buttons and PAN controls.

Table 2:

Channel	ASSIGN	PAN
1	1-2	L
2	1-2	R
3	3-4	L
4	3-4	R

Here's the rest of your pre-flight checklist:

- Chan 9-16—ASSIGN L-R
- TRIMs—Unity Gain
- Chan 9-16 PANs—where you want them
- CHANNEL FADERS—Unity Gain
- EQs—Flat (all knobs in their centered position)
- MAIN MIX FADER—Unity Gain
- SUB FADERS 1-4—Unity Gain
- CTL ROOM SOURCE—MAIN MIX
- CTL ROOM/PHONES volume—up a bit

Set

Perform the Famous **Mackie Level-Setting Procedure** for all the instruments and mics. Don't be shy. When setting levels, sing and play as loud as you will during a real take. If you'll be recording parts one at a time, set levels one at a time. If the whole band will be playing together, set levels during a run-through.

Now the mixer will be happy, but what about the recorder?



WARNING: You're about to send the recorder outputs to the mixer, which is connected to the control room speakers, which can be picked up by the mics if they're in the same room as the

speakers. The mics feed the recorder, which feeds the speakers, which feeds the mics, which feeds . . . This is why the loud squeal is called *feedback*.



If you're recording with the mics in the Control Room with the monitor speakers, you **must** turn the speakers off—really off—switch off the power amplifier or powered speakers. Since, in this application, we're using the mixer's head-phone jack as a cue feed for the players, turning down the CTL ROOM/PHONES control, while it will prevent feedback in the monitor speakers, will also kill the cue mix to the musicians in the studio. If the live mic is in another room, or quite far away from the speakers, you're safe.

Set the recorder to Input Monitor mode (or Auto Input Monitor with all of its Record Ready buttons pressed) and you should see the recorder's meters move as you play. If the record levels on all channels look about right, you're good to go. But what if they aren't?

Used to be that all recorders had input level controls, but today many don't. If your recorder has input level controls, use them to adjust the record level according to your recorder's instructions. If not, you'll need to make adjustments from the mixer.

Chances are, if the recording level is way off it'll be much too high, because the recorder is expecting a semi-pro level and your Mackie is sending it a pro level. You can use the faders on channels 1-8 to adjust the level going to the recorder.

If you need to raise the record level, as you may to approach peak level on a digital recorder, listen carefully for distortion. When you push the faders up past unity gain, you'll be running closer to the clipping point of the mixer channel. Remember, particularly with digital recorders, it's better to set the record level conservatively than to risk distortion. See the metering discussion in the Tips section for some hints on setting levels and why you need not worry if you can't turn on all the meter lights all the time.

There's one more set of TRIMS to adjust, and those are on the Recorder Return channels 9-16. With the band playing and the recorder still set for Input monitoring, perform the **Level-Setting Procedure** on channels 9-16. You can cheat a little here—set

one TRIM using the SOLO button, then set all the rest of the Recorder Return TRIMs to the same position as the one you set using the official method. You can get away with this since if you set the record level properly, all the recorder returns will be coming in at roughly the same level. Once all your tracks are recorded, you'll want to re-check the TRIMs individually before doing the final mixdown.

This sounds complicated, but in reality, once you get the hang of the procedure it takes less time to do it than to read about it.

Go!

Now for the easy part. Start the recorder and play like you've never played before. Then rewind, press Play, and be amazed at your musical talent.

If you're a one-person band, you'll be recording tracks in multiple passes (which is probably why you wanted to get into multi-track recording in the first place). Keep your wits about you. Don't forget that once you've recorded a track, switch the recorder out of Record Ready so you don't erase the track on the next pass.

When you record the second pass, you'll hear your first pass playing back so you'll have something to play along to. Pretty soon you'll be overdubbing just like a pro. Use Channels 9-16 to adjust the mix that you're listening to while overdubbing.

Don't forget the warning about mics getting too close to speakers. You can record keyboards, drum machines, and electric instruments while listening to the control room monitor speakers. When it gets time to record the vocals or acoustic guitar tracks, however, kill the control room speakers and plug in the headphones.

Mixdown

A really cool thing about this setup is that at any time, you're ready to mix. No cables to patch, no buttons to press. In fact, you were probably doing some panning, level adjustments, and maybe some EQ on the recorder return channels as you were tracking. Those are all elements of mixing and the more you do it, the better you get. For now, the most important thing is to learn how to use the tools.

If it sounds great just the way it is, mixdown is as simple as checking the record level and

pressing the Record button on the 2-track recorder.

Do the vocals need some reverb? With AUX RETURN 2 set to its Unity Gain position, turn up AUX SEND 4 on the vocal tracks (in this example, Tracks 1 and 2 on the recorder, which are on channels 8 and 9 on the mixer). Need some delay on the guitar? The delay is waiting for you to turn up AUX 3. If you want to compress the bass track, unplug the compressor from INSERT 1 and plug it into INSERT 12.

Keep an eye on the LEFT/RIGHT meters as you're mixing. Levels tend to creep upward as you work. It's a fact of life. You may need to drop all the channels by a few dB in order to keep the mix level peaking around 0 VU. If you've added a lot of EQ boost on a channel, SOLO the channel and re-adjust its TRIM to bring its level back to normal.

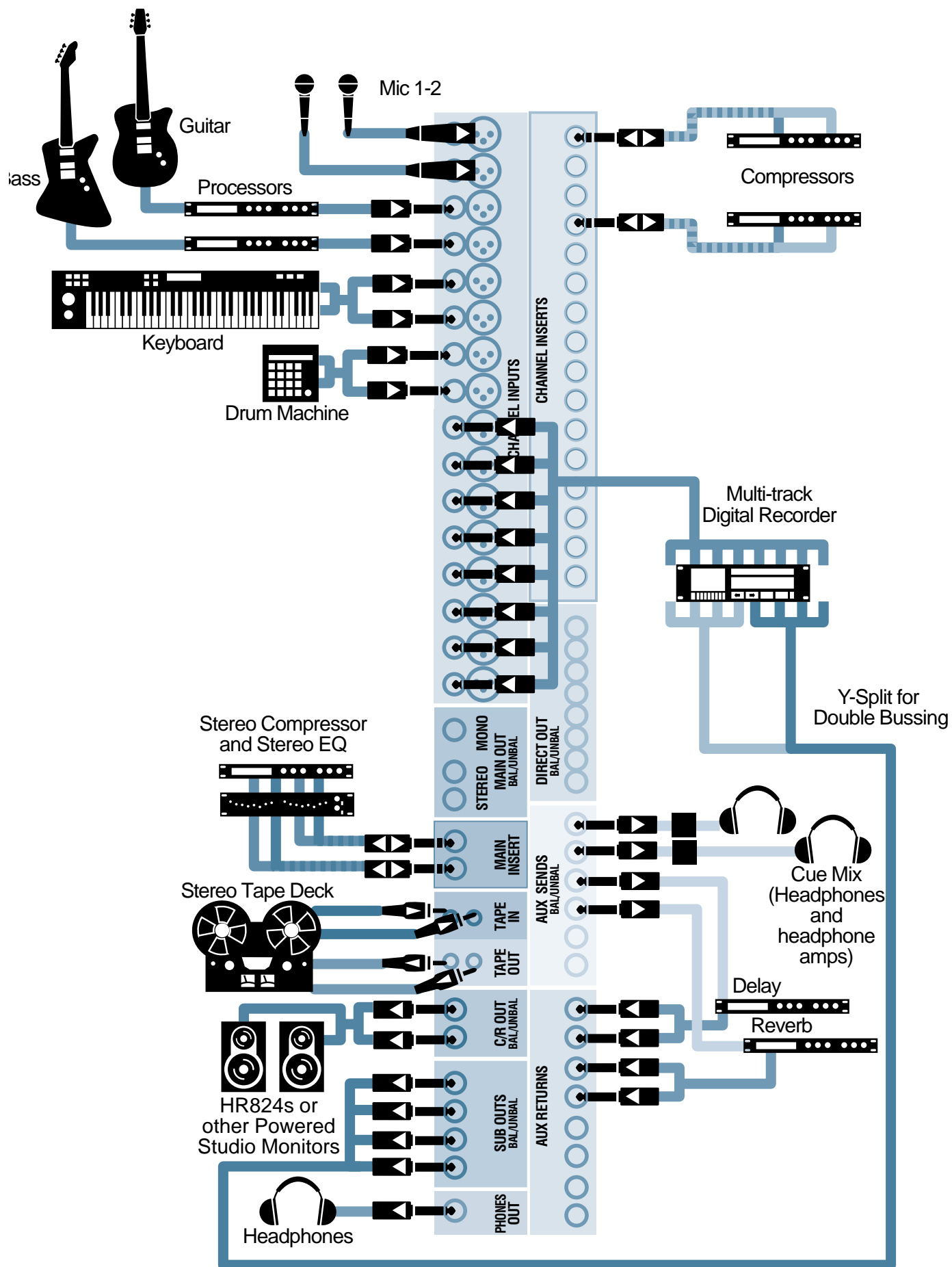
Our example shows a stereo equalizer and compressor in the MAIN inserts. Once you have a pretty good mix, that's the time to put those into service to see if you can make the mix any clearer or "hotter" (if that's your preference). There are some pretty remarkable digital "final processors" available today that can add a lot of punch and sparkle to a mix, but they're power tools, and a little goes a long way.

One your mix sounds good enough to show off, check the record level on the mixdown recorder and let 'er rip.

Other Cues

If you're recording bands in the studio, they may want to hear something other than a well balanced mix that you're creating in the control room. A headphone amplifier would be a good addition to your system. A headphone amplifier typically has several phone jacks, each with its own volume control so the musicians can adjust their own level without blasting each other or driving you out of the control room. See the next hookup diagram for a headphone amplifier suggestion

I604VLZ-PRO 8-Track Recording, Take 2



.Bouncing Tracks

If you like to build up a lot of layers in your recording, a quintuple-tracked guitar lead, for example, when using an 8-track recorder, you'll have to do a lot of track bouncing. Using this technique, you can make your eight tracks seem like a lot more by recording several related parts on individual tracks, then mixing those tracks while recording the mix on to one or two available tracks. Then, screw up your courage, erase the original tracks, and you have room to record more parts.

Double Bussing

One thing that can make bouncing easier is having the flexibility to send any channel or mix of channels to any recorder track. Using direct outputs requires patching or moving cables, but as long as you don't need to record on more than four tracks at a time, you can make the four SUB outputs do double duty by splitting each one out to two recorder tracks.

In the Double Bussing Hookup Diagram, each of the four SUB OUTs feed a pair of tracks. SUB 1 goes to Tracks 1 and 5, SUB 2 goes to Tracks 2 and 6, etc. The way to do this is to buy or build Y-Adapters or Y-Cables, which have one input and two outputs. (A Y-cable is NOT the same as an Insert cable) Once you've connected the recorder in this manner, you'll be able to route any input to any track by using the ASSIGN buttons and PAN knobs.



Some recorders and some multi-track recording software allows you to "double bus" right at the recorder's input. The Alesis ADAT™ is one example. On this recorder, you need only connect the SUB OUTs to recorder inputs 1-4 and a button on the recorder takes care of the other four inputs. The 1642VLZ-PRO has a double set of jacks on the SUB outputs, so no adapters are required.

Recording The One-Person Band

Here's an example of how you'd record using a double-bussed setup. Let's say you'll begin by playing rhythm guitar along with the drum machine, then add a bass track, a keyboard, sing the lead vocal, add lead gui-

tar and keyboard solos, then finally put on some background vocals. That's more than 8 tracks, isn't it?

What we're about to do is record the rhythm section, then bounce that to two tracks, and record lead parts over the original rhythm tracks. Let's see how this works.

Pass 1

We're still using Channels 9-16 for our recorder returns, so assign them to L-R. We'll be using the SUB outputs, so bring the SUB faders up to Unity gain.

Now, let's assign some inputs to tape tracks.

Table 3:

Chan	Source	Assign	Pan	Track
7	Drum L	1-2	L	1
8	Drum R	1-2	R	2
3	Gtr	3-4	L	3
4	Bass	3-4	R	4

Put tracks 1, 2 and 3 into Record, start the drum machine, and record the rhythm guitar along with the drums. Adjust the faders on Channels 9-10 (drums) and 11 (guitar) for a comfortable mix. It's OK to turn the guitar up too loud at this point if it helps you to play well.



When you're happy with the rhythm guitar recording, don't forget to take Tracks 1, 2, and 3 out of Record Ready or you'll erase 'em quicker than you can say "Oops!"

Pass 2

When the guitar part's done, rewind, put Track 4 into Record, and play the bass part while listening to the guitar and drums. The bass channel is already assigned to a recorder track, and it'll show up for monitoring on Channel 12. Adjust the 9-12 mix so that you can play comfortably and hear what you're doing.

Pass 3

This is a good time to add a keyboard part to the rhythm section before buttoning it

up. Let's make it a mono track since we don't want it to clutter up the mix.



Hey! We've just made a production decision and we've just started tracking! This is one of the things you'll need to do when bouncing. Might as well get used to it. There's no reason why we couldn't record the background keys in stereo, but it wouldn't illustrate the next point, so give us a break.



It's time for some housekeeping. We're done with the drums, bass, and guitar for a while, so un-assign them from 1-2 and 3-4. We need those busses for other things and we don't want stray electronic noise creeping in.

Now, let's send the keyboard to a recorder track:

Chan	Source	Assign	Pan	Track
5	Keys L	1-2	L	5
6	Keys R	1-2	L	5

Note that we've panned both keyboard outputs to the left, sending them to SUB OUT 1. Since that's wired to Track 5 as well as Track 1, the keyboard signal will get to the desired track.

Put Track 5 into Record, and tickle the ivories while listening to the guitar, bass, and drums. Starting to sound like a band? Good!

Bounce Time

Let's take inventory of our recording so far:

Track 1—Drums L

Track 2—Drums R

Track 3—Rhythm Guitar

Track 4—Bass

Track 5—Background Keyboard

Play through the rhythm tracks a few times, and work up a good mix using the levels, pans, EQ, and effects if they're called for. Give it plenty of thought since, after the next step, you'll have to live with this mix for the rest of the project. You'll probably want to pan the drums fully left and right, put the bass in the center, and pan the rhythm guitar and keyboard fairly far off center on opposite sides to leave room for the leads.

When you're happy with the rhythm section mix, it's time to record it. We'll put it on Tracks 7-8 since they're available. Un-assign all the input channels (1-8). Now, switch the assignment of the rhythm channels (9-13) from L-R to busses 3-4.

Put Tracks 7-8 into Record Ready and roll the tape. You should hear your rhythm mix coming in on Channels 15-16. Check the record level on Tracks 7-8 and use the SUB 3-4 faders to adjust if necessary.

When the level looks good and the mix sounds good, rewind, press the Record button, and see if you can dance to it.

Now, rewind and play back your rhythm mix a couple of times to be sure it's what you want. You'll be erasing the tracks that made up that mix, so there's no turning back.

Pass n

The good news is that you now have a rhythm section and six available tracks. What you do with them and the order in which you record further passes depends on your song and your creativity. A basic demo might only require a lead guitar or keyboard, a lead vocal, and maybe a background vocal. Plenty of room for that. Simply continue the procedure of assigning an input to a track, recording that track, and moving on until you've filled all the tracks and recorded all the parts you wanted.

You can continue the bounce process, though, if you need to record more parts than you have remaining tracks. Want a five-layer guitar solo? Record the parts on tracks 1, 2, 3, 4, and 5, then bounce them to Track 6. Want a four part backup chorus? Record on Tracks 2, 3, 4, and 5, then mix and bounce them to Track 1. It's up to you.

Studio Headphone Monitoring

The mixer's headphone output follows the control room source, which, most of the time, is the L-R MAIN MIX. When tracking, you may want to hear a something different in the headphones. The hookup diagram shows an external Headphone Amplifier connected to the AUX 2 OUTPUT.

To create an independent headphone mix, for example, one for the lead singer that's

heavy on vocals and bass but has just enough of the drums so he can keep rhythm, adjust the AUX 2 knobs on the recorder return channels to make the singer happy. You'll probably want that headphone mix to be pre-fader so the singer doesn't hear any tweaks that you make to your control room mix while he's singing, so press the PRE buttons on those channels.

Since we're using RETURN 2 for the reverb, adding reverb to the headphone mix is easy—just turn up the TO AUX SEND 2 knob.

Subgroups and Submixes

We've shown how subgroups can be used to mix several tape tracks down to one or two, but they can also be used to mix live inputs. In these examples we've used a stereo drum machine for our drum tracks, but if you have a drummer with a real drum kit, it's common to set up several mics on the drums and use subgroups to mix those mics to two channels which you'll send to the recorder.

Connect your drum mics to the lower numbered mixer channels, assign them to subgroups 1-2, and record as in the example. By monitoring the drum track recorder returns (channels 9-10 on the mixer) you can hear your drum mix and adjust the faders, pans, and EQ on the input channels for a good drum sound.

You may want to initially devote four tracks to drums, and if necessary, mix those down with the rest of the rhythm section on a bounce pass. Typically when working with fewer than 16 tracks, the kick and snare get their own tracks, with toms, cymbals, and overhead mics mixed to two other tracks.

This same technique works well when you're recording large groups—a string or horn section, or a vocal chorus—as part of a multitrack project. Assign the mics to a pair of subgroups and record the section as a stereo pair of tracks.

16 Tracks on the I604VLZ-PRO?

Sure, why not? After all, it's a sixteen channel mixer. Remembering that a channel INSERT is both an output and an input, you can put up to 16 recorder tracks "in

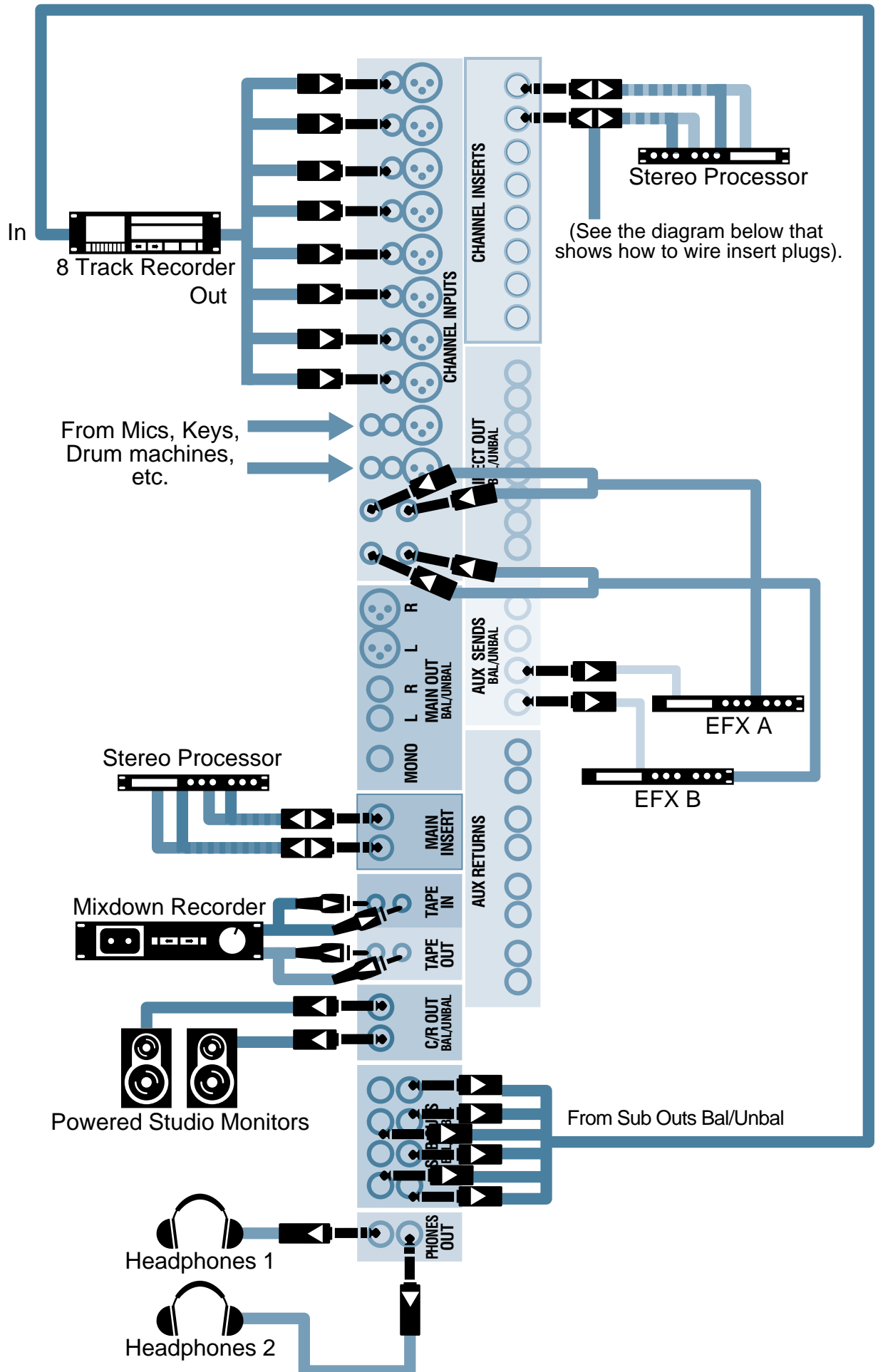
line" with the channels. The Insert Send gives you one mic preamp output to every track, and the Insert Return is a line level input for every channel.

If this isn't a big enough hint, see the 1202/1402 8-Track setup for a more detailed description of this setup.

Using the Insert Outputs connected to up to sixteen recorder tracks is an excellent way to make a live recording of your band. When you get home, just patch the recorder outputs to the mixer line inputs and it'll be just like you were there playing on stage, directly into the microphones.

I642VLZ-PRO

8-Track Recording



The 1642VLZ-PRO has a lot in common with the 1604. This hookup shows the 1642VLZ-PRO set up for 8-Track recording with split monitoring and using the built-in double-bussing feature on the SUB outputs. Refer to the 1604 operating procedures for a how-to-do-it description.

Inputs and Outputs

It's nice to have all the cables in place so you don't need to re-patch when going from tracking to mixdown, and that's a plus with this hookup.

Multitrack Recorder Connections

In this application, we're using Line Inputs 1-8 as the recorder returns. Those will normally be assigned to L-R so they will appear in the headphones, control room monitors, and at the inputs of the mixdown recorder. Since the 1642 has 8 mono and 4 stereo inputs, this layout is more appropriate for track bouncing and one-track-at-a-time recording than bringing the recorder returns in on the last 8 (stereo) channels.

Connect the sound sources to Mono/Stereo inputs 9-10 and 11-12. Use the PAN controls and ASSIGN buttons to route the inputs through the SUB Outputs to the recorder inputs. Like with the 1604 hookup, we're using subgroups as signal routers, even though they may not be summing more than one signal.

The 1642VLZ-PRO provides two jacks for each of the four SUB outputs. Connect these to the eight inputs to the 8-track recorder.

Effect Processor Connections

The two effect processors in this hookup are fed from AUX SENDs 3 and 4, and, in this example, are returned to the stereo mix through the two stereo line level input channels 13-14 and 15-16.

The effect processor outputs could also be connected to AUX RETURNS as shown in the 1604 recording hookup. This would free up the stereo line inputs for additional recording sources—a drum machine or another keyboard for instance. But variety is the spice of mixer hookups.

A Purist's Recording Path

For recording with the minimum electronics in the signal path, you'll want to patch channel direct outputs or insert outputs directly to the recorder inputs. Of course you can always connect a line level source directly to a recorder input for the purest signal path.

Since only Channels 1-8 have direct outputs and inserts, you'll need to free those up if you want the most direct path from the mic preamp to the recorder. During tracking, you can move the recorder returns to the eight stereo line level inputs 9-16 (you'll have to move or lose the effect returns).

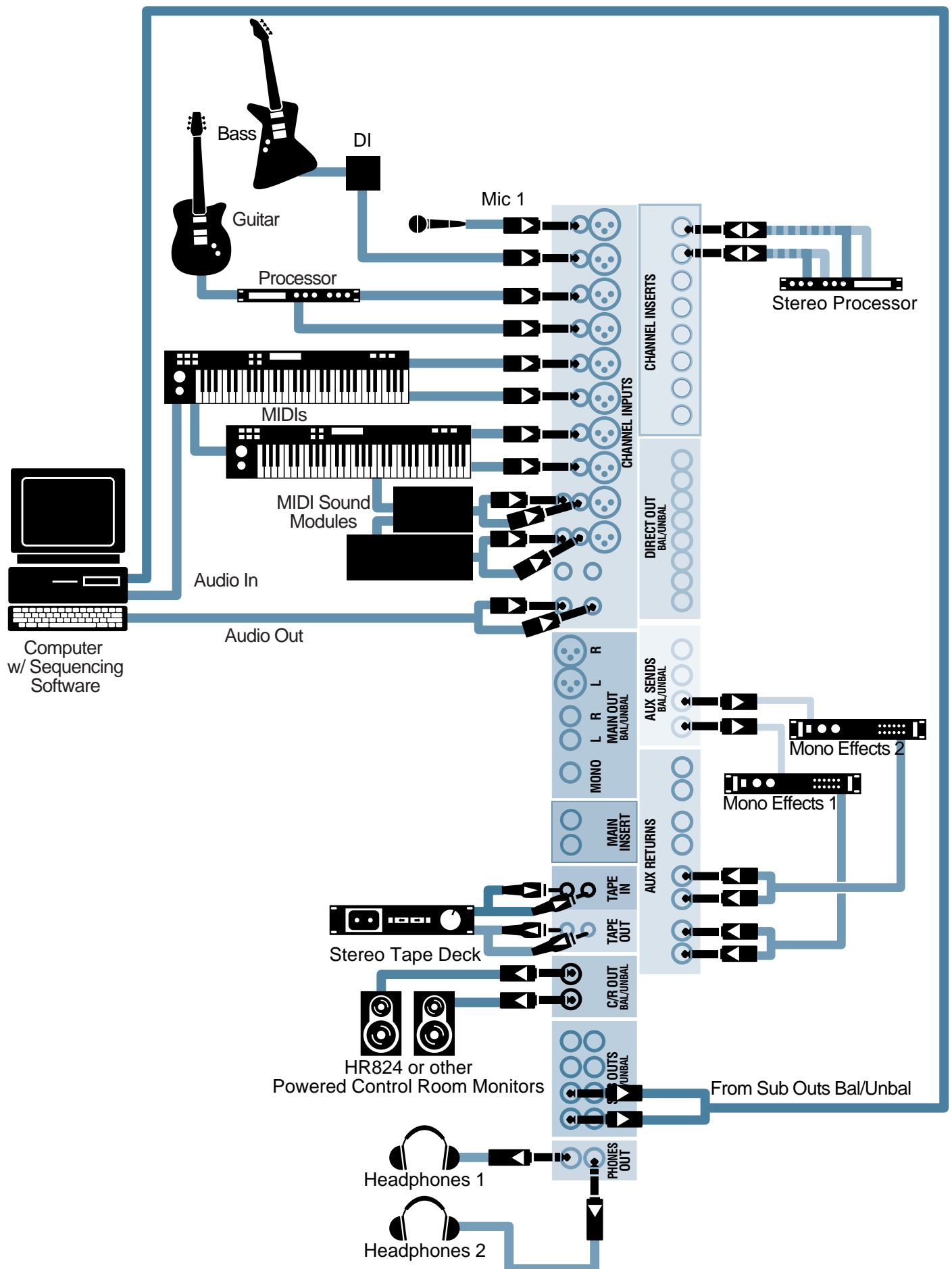
This won't give you all the panning flexibility that you have on the mono inputs, since when both inputs of a stereo channel are connected, each input is routed to either the Left or Right and the PAN only adjusts the balance between the two. It'll do for monitoring while tracking, however.

When you get ready for mixdown, you won't need the direct outputs so move the recorder outputs back over to Channels 1-8. After a few switcheroos like this and you'll be ready for a patchbay.

Recording, Overdubbing, and Mixing

Refer to the 1604VLZ-PRO section for a blow-by-blow description of a tracking session. The only variations are where we've chosen to connect things in this example.

Chapter 9: I642VLZ-PRO MIDI Studio



With its wealth of stereo line level inputs, the 1642VLZ is well suited for a MIDI-based studio. The computer serves as a MIDI sequencer and doubles as a stereo or even multitrack audio recorder for recording acoustic instruments and vocals.

Connections

Stereo inputs are used for the MIDI sound modules. Keyboards and the stereo outputs of the electric guitar processor are connected to mono line inputs. The bass is connected to a mic input through a direct box (DI), and the microphone is connected to a mic input.

The stereo outputs of the computer sound card are connected to the stereo LINE IN 15-16 jacks. Assign that stereo channel to L-R so you can hear audio playback from the sound card.



Inputs to the sound card come from SUB OUT 1 and 2. The reason we're feeding the sound card (which is the primary recorder in this setup) from the SUBs, and not from the MAIN or TAPE outputs is because it's awkward to turn off the "input monitor" function of many sound cards. If both the input and output of the sound card were connected to the MAIN L-R bus, feedback would result when you tried to listen to playback from the disk.

We're using those TAPE IN and TAPE OUT jacks for a cassette deck so that you can make a quick reference copy to pass around to friends. Of course you could substitute a CD recorder.



Be sure that you don't turn up the TAPE IN control when the deck is in Input Monitor or Record mode or you'll get feedback.

Operation

Sequencing, Rehearsing, or Jamming

In this mode, you won't be recording audio into the computer, so assign the instruments to L-R to send their outputs to the Control Room monitor speakers. At any time, you can record a scratch-pad mix to the cassette deck—it's at the ready, connected to the MAIN outputs.

Recording Audio to the Computer

Assign the instrument or mic you want to record to bus 1-2. Since you'll want to hear what you're playing and perhaps hear some MIDI tracks, too, don't unassign it from L-R. Mute the sound card playback channels (15-16) so you won't get feedback..

Latency



Traditional hardware recorders have a physical switch that connects their input and output jacks together when in the Input Monitor mode. Sound cards perform the same function with software, but the audio has to get converted from analog to digital and back to analog before getting to the output jacks. This takes some time, usually on the order of 1-3 milliseconds, and it's significant enough to have earned a name—"latency". Hearing the delayed and direct sound together can sometimes be annoying. With the hookup described here, you can hear the direct sound and mute the output of the sound card.

To listen to the playback of your audio recording, simply un-mute the sound card return channels 15-16.

Refer to the 1604VLZ-PRO 8-Track recording application section for an explanation of how to use the effects, and for making alternate headphone monitor mixes.

Overdubbing

You can overdub if you're using multitrack software that allows you to record several tracks, mix them in the computer, and send that mix out to the sound card's stereo outputs. Sound cards and recording programs work in many ways, so you'll have to pay close attention to their documentation to accomplish what's necessary.

For overdubbing, you need to hear what's being recorded, but you also need to hear what you're playing or singing. The recording software (or possibly the sound card's driver application) may contain a monitor mixer which allows you to send the track being recorded into the mix along with the previously recorded tracks. You can use this feature and just monitor the sound card output on Channels 15-16.

Sometimes this is clumsy, though, so it may be easier to mute the track being recorded in the sound card software—mute the out-

put of course, not the input, and assign the mixer channel you're using for the input to *both* the subgroup feeding the sound card (1-2) and the L-R busses of the mixer. In that way, you'll hear your input live, directly through the mixer, as well as the playback of your previously recorded tracks.



Remember our friend *latency* from a few paragraphs back? When sending a live input to the monitor mix, you *absolutely must* be sure that you're listening to only the live input, and not the live input plus that input delayed through the sound card and added to the monitor mix from the sound card output. Combining these two copies of the signal with one delayed by a couple of milliseconds will produce a strange effect called "comb filtering". Although your recording will be fine, what you hear in the monitors while you're recording it will sound funny.

Of course if you have an interface that has eight inputs as well as eight outputs, treat it like an 8-track recorder and refer to the 8-track hookups.

Alternate Hookups

Computer Audio Playback

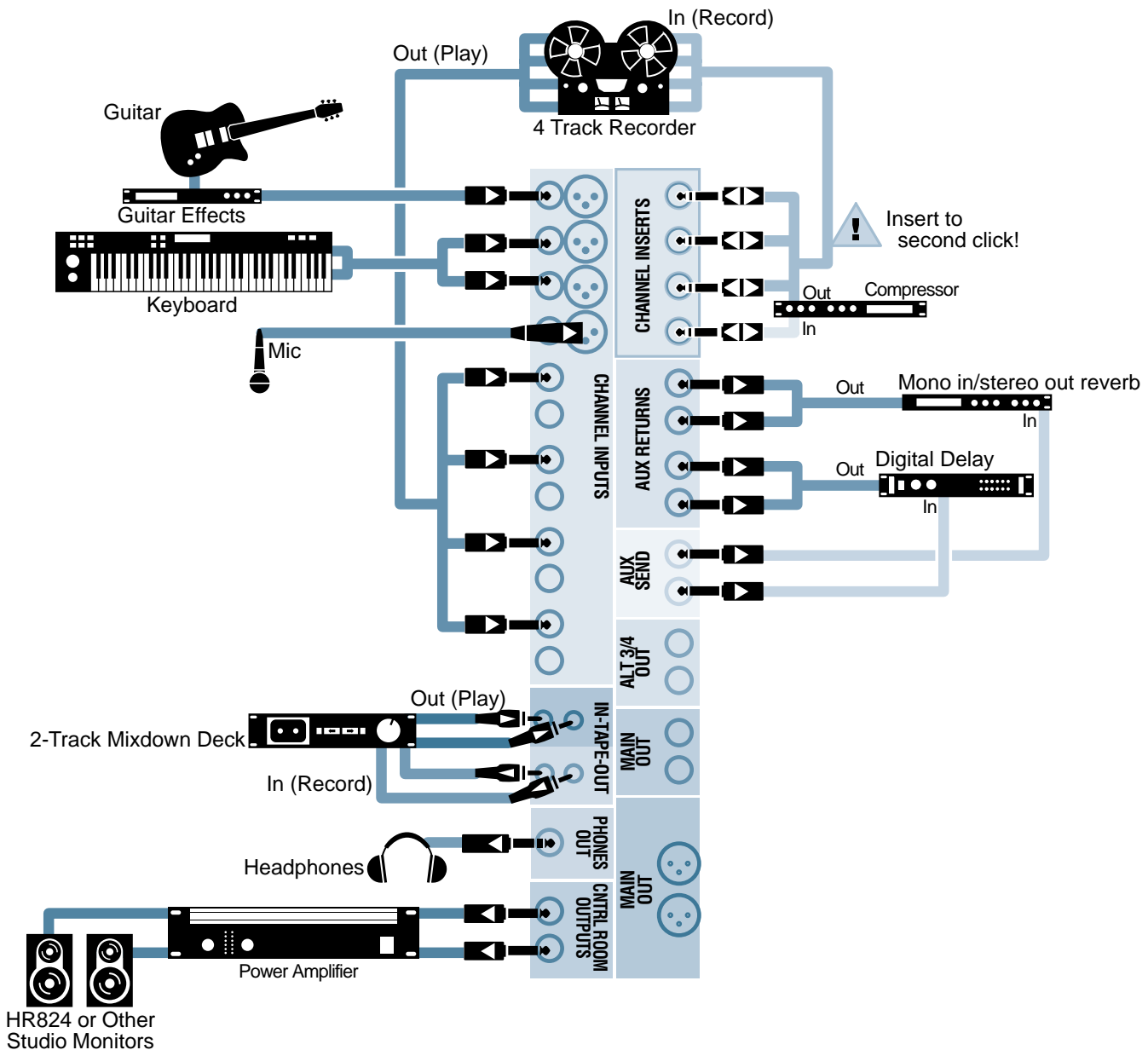
If you're cramped for input channels, the sound card outputs could be connected to the TAPE IN jacks instead of an input channel. In this case, you'd turn up the TAPE IN knob when you wanted to hear playback from the disk. This is less convenient than using a channel with a MUTE switch.

Multi-Channel Audio Interfaces

A standard stereo sound card has two inputs and two outputs. If you build up multiple tracks on the computer, you'll be mixing them using the computer's tools, not your Mackie mixer.

Many people prefer hands-on mixing, however. There are several audio interfaces available that offer stereo inputs (on the premise that you'll only be recording one or two instrument or vocal tracks at a time) but rather than two, they provide four or eight outputs. With this type of interface, in the recording software you can assign tracks to separate outputs. This is sort of like half of an 8-track recorder—eight tracks for playback but only two tracks for recording. If you have an interface like this, you can free up some line level inputs on the mixer and connect the eight audio outputs from the computer to eight mixer inputs for more mixing flexibility.

Chapter 10: I202VLZ-PRO and I402VLZ-PRO 4- and 8-Track Recording Applications



The 1202 and 1402VLZ PRO mixers are very similar in architecture, differing primarily in the number of inputs. We'll describe applications for these two mixer together in this section.

Connections

We're using the mixer in a split monitor configuration, connected to a four-track recorder. Each recorder output is connected to the mono (odd-numbered) LINE INPUT of a stereo channel, allowing the PAN knobs to act as true pan pots rather than stereo balance controls. The recorder return inputs are mixed to the MAIN L-R outputs.

This hookup is suitable for basic 4-track recording. The 4-track recorder is fed from the channel inputs, using the INSERT jacks as direct outputs. If you expect to be bouncing tracks, you'll probably want to use an alternate hookup that reduces the amount of re-patching you'll need to do.

The LINE INPUTS on the 1402VLZ-PRO have switches to set their input sensitivity to match -10 dBV or $+4$ dBu outputs. Choose the level that's appropriate for your recorder. The input sensitivity of the 1202VLZ-PRO's LINE INPUTs is fixed, but there is sufficient range on the channel GAIN control to accommodate either operating level.

When recording with a split monitoring setup, you listen to a mix of the recorder outputs rather than the channel inputs, so we want to keep INPUTs 1-4 (which feed the recorder directly) out of the MAIN L-R MIX. Inserting unbalanced plugs fully into the INSERT jacks (to the second click) accomplishes this by opening the normalizing contacts in the jacks.



Pressing the MUTE button on those channels will also remove the channel from the MAIN MIX without interrupting the signal to the INSERT OUT. Why didn't we do it that way? We just wanted to show you another way to do it which may come in handy for something else some time. It's a good trick to know. Besides, we'll show you another application which makes use of the MUTE/ALT 3-4 switch. Hang in there.



Note that the mic on Channel 4 has a compressor inserted in line between the channel INSERT output and the recorder input. This illustrates how you can use an in-line signal processor for recording even though you're using the INSERT jack as a direct output.

A stereo mixdown recorder is connected to the TAPE OUTPUTS and INPUTS. The MAIN MIX always appears at the inputs of the 2-track recorder. Selecting TAPE as the CR/SOURCE allows you to hear your mix without any re-patching.

We have two effect processors in this setup. They're fed from AUX 1 and AUX 2 outputs, and the processor outputs return to the mix through stereo AUX RETURNS 1 and 2. AUX 2 (EFX) is permanently post-fader, which is appropriate for effect sends. AUX 1 (MON/EFX) is normally pre-fader for a monitor send, but can be switched post-fader by pressing the AUX 1 SELECT (PRE/POST) switch.

Recording



If your mics are in the same room as the control room speakers, avoid feedback by keeping the recorder return channels for the mics muted. When recording from the mics, turn the control room speakers off by turning off their amplifier or turning its input level control all the way down.

Now that it's safe, set the recorder to Input Monitor or Auto Input Monitor and start tracking.

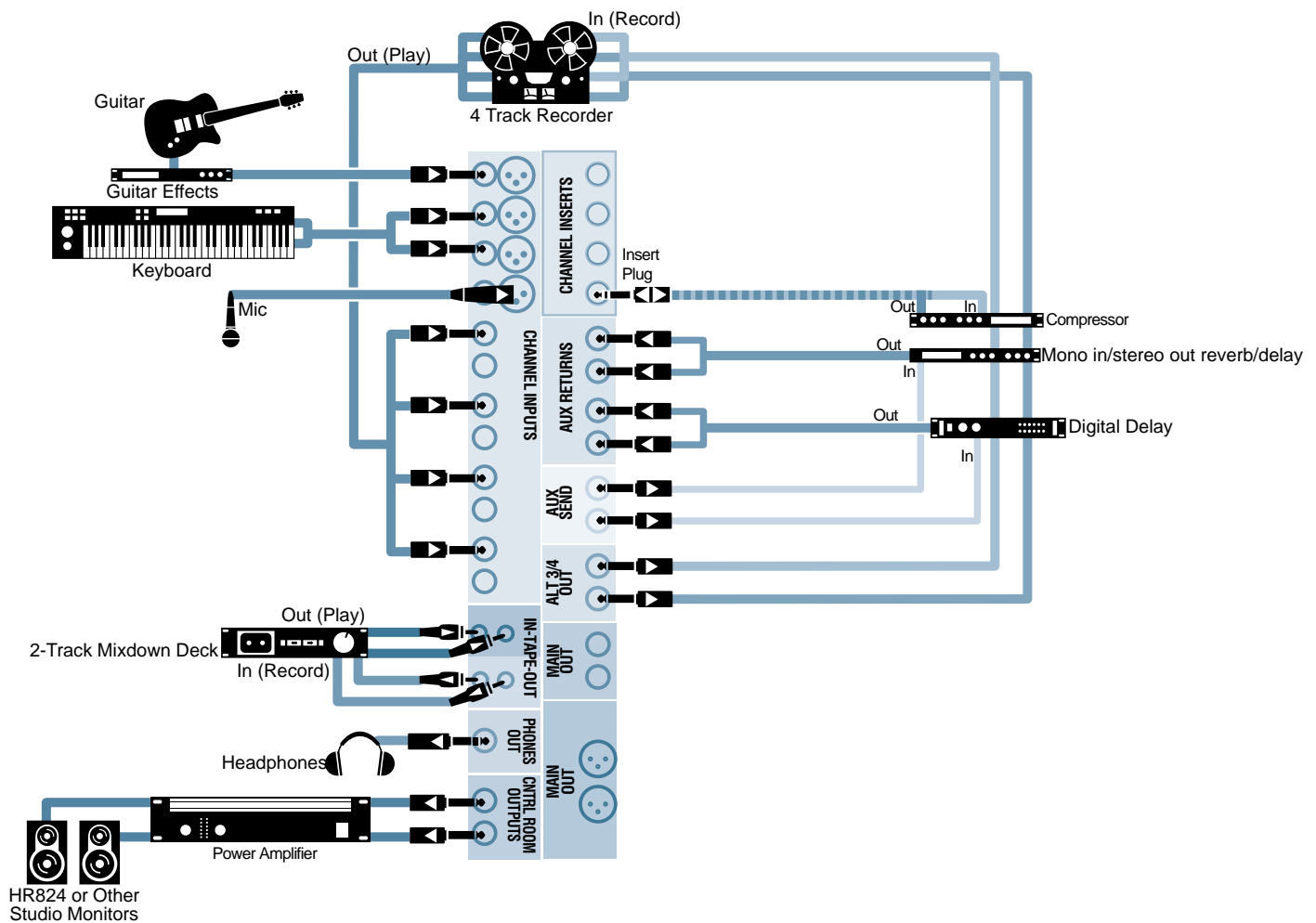
Perform the **Level-Setting Procedure** on the input channels. Refer to the 1604 setups for step-by-step procedures.

Mixing

Everything's already set up for mixing. Adjust the faders, pans and EQ on the recorder return channels to taste. Effects can be added to the mix by turning up the AUX1 and AUX 2 knobs on the recorder channels.

If you need to use an in-line processor such as a compressor on a recorded track, disconnect that track's output from the mixer input and insert the processor in line. The hookup is the same as for the processor that's shown patched in line with one of the recorder inputs.

Chapter II: I202/I402VLZ-PRO 4-Track Recording—An ALT Approach



Using the ALT Outputs

This variation on the previous setup shows how the ALT3/4 OUTPUTs can be used to feed the recorder. This will give you more flexibility when bouncing tracks.

As before, the recorder outputs are connected to mixer line inputs which are routed to the MAIN L-R bus so we can always monitor the recording.



Remember that the recorder is part of the monitoring path, so you *must* enable Record on a track in order to hear that channel in the control room monitors and phones.

Double Bussing

We've shown the ALT OUTPUTS split with ALT L/3 feeding the odd numbered tracks and ALT R/4 feeding the even numbered ones. See the discussion on "double bussing" for more info.

For a cleaner signal path, you can patch an INSERT output to a recorder track, but you'll need the ALTs in order to bounce tracks without re-patching.

INSERTS

In the previous setup, the INSERTs served as direct outputs to feed the recorder, so we had to patch our compressor in line with the channel which we wanted to compress. Using the ALT outputs to feed the recorder frees up the INSERT jacks, so in this setup, you can use an Insert Cable to patch a compressor or other signal processor into a channel path.

Bus Assignments

When something's connected to the ALT outputs, MUTE/ALT 3-4 serves as a bus assignment switch. In the normal (*up*) position, inputs are routed to the MAIN L-R outputs. In the MUTE/ALT (*down*) position, the channel signal is removed from

the MAIN outputs routed to the ALT outputs.

The PAN control functions on the ALT busses just as it does on the MAIN L-R busses. Panning fully in one direction (“hard panning”) assigns the channel to only one bus. Panning between extremes routes the signal to both outputs in proportion to the PAN setting.

Making Tracks

To record the guitar on Track 1, press the ALT switch on Channel 1 and turn the PAN knob fully left. Be sure that recorder Track 1 is in Record, and that Track 3 (which will be getting the same signal) is in Safe or Play mode.

More Guitars

Let’s say you want to record three guitar parts. Since you’ve routed the guitar channel to the odd numbered tracks, simply take Track 1 out of Record and put Track 3 into Record. Now, you’ll hear the playback of your first guitar track and you can play along, recording the second part.

To add a third guitar track, swing the PAN knob fully clockwise and you’ll send the guitar signal to the even numbered tracks. Take Track 3 out of Record, put Track 2 into Record, and add the third part.



We’ve been working with a direct guitar output here, but when working with the mic, if it’s in the same room as the speakers, be sure to kill the speaker output or you’ll get a blast of feedback!

Bounce Time

We’ve filled up three of our four tracks and all we have is the guitar parts. What we’re about to do is assign the recorded tracks to the ALT bus so we can mix them and record the mix to the remaining track.

First, though, since we want to record *only* the guitar mix and not extraneous signals and noises from our input sources, switch the input channels from ALT back to MAIN by releasing the MUTE/ALT buttons. So stray sounds don’t detract us, turn the GAIN controls on the input channels all the way down.

Now, assign the Tracks 1-3 return channels (5/6, 7/8, and 9/10) to the ALT bus by pressing the MUTE/ALT switches on those channels. Since you’ll be recording on Track 4, set the PAN knobs on those channels fully clockwise.

Leave Track 4’s channel (11/12) assigned to the MAIN bus so you can hear the mix of your guitar tracks. Since you’re mixing to mono, pan this channel to the center for now. In the final mix, it can go wherever you want, but it’s usually easier to hear your bounce mix if it’s in the center.

Be sure Tracks 1, 2, and 3 are in Repro (play) mode. Put Track 4 into Record Ready.

Now rewind, roll the tape (or spin the disk) and you’ll hear all three of your guitar parts. Adjust the EQ and GAIN on the guitar channels until you have the right blend. Keep an eye on the recorder’s meter while you’re mixing. Since there’s no Master level control for the ALT 3-4 bus, the record level will be determined by the levels going into the ALT 3-4 mix. When you have both a good mix and the proper record level, rewind, press the Record button, and relax while the recorder is making your composite guitar track.

Wow! Three More Tracks!

Now you have three tracks you can erase and record additional parts. A six track recorder for the price of four! Or you can continue the bouncing process until you run out of ideas or start building up too much noise, which will eventually happen, even with a digital recorder.

More Stereo

With only four tracks, you can have only one or two stereo tracks going into your final mix, but you can still get pretty creative.

If you have a good quality stereo mixdown recorder (and you should!), you can do what’s called an off-line bounce. Instead of bouncing three tracks to the fourth track of the 4-track recorder, first mix them to stereo and record that mix on your mixdown recorder. Then record that stereo mix back to two tracks on the 4-track. You’ll need to move some cables to do this, but it’s not complicated. Unplug the recorder input cables from the ALT OUTPUT jacks and plug them into the 2-track recorder outputs.

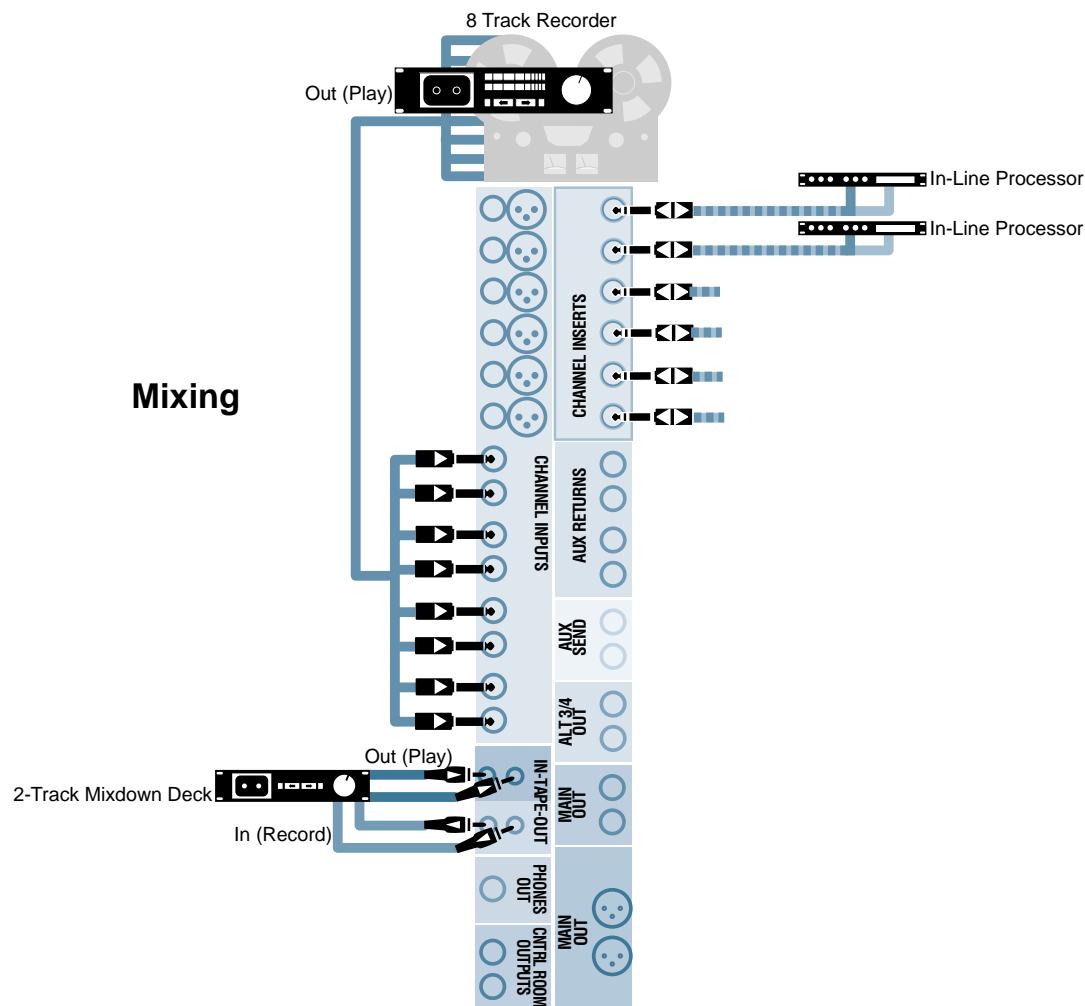
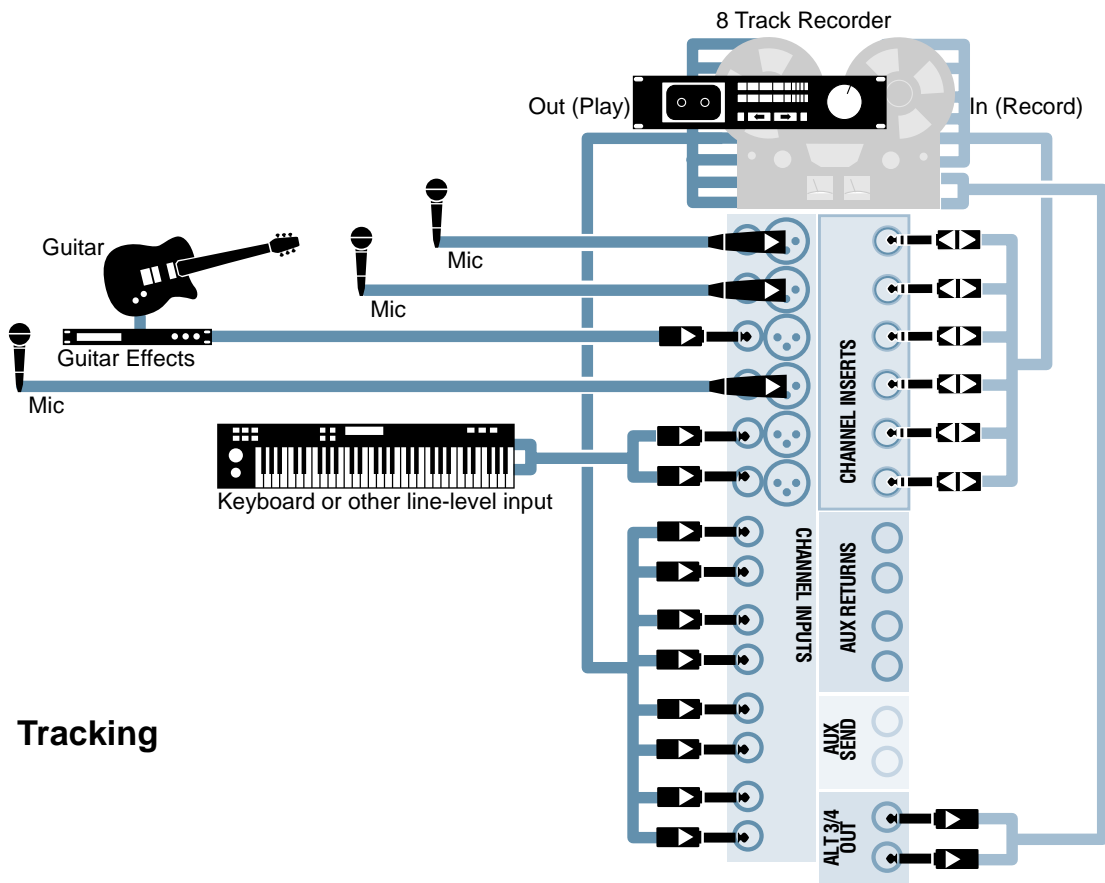
After this bounce, you can add two more tracks, mix those together with the first bounce mix, and then you'll have six parts in stereo on two tracks.

You can get extremely creative if your stereo mixdown recorder is a computer with a sound card. You can record tracks, build loops, and then use the multitrack to build songs around them. The sky's the limit.



Since there's no means (at least none we've described here) to synchronize the stereo and multitrack recorders, this technique only works when you're bouncing *all* of the tracks in one pass. Other than with good coordination and some luck, there's no way to, for example, leave one guitar track intact on the 4-track, mix the other three to the outboard recorder, and then put that mix back in sync with the remaining guitar part on the 4-track.

Chapter 12: I202/I402VLZ-PRO 8-Track Recording



So you bought a brand new 8-track recorder. Much as you love peanut butter sandwiches for three meals a day, maybe it'll still be a few weeks before you can swing the purchase of that new larger mixer. Don't despair. You can do 8-track recording with your 1202 or 1402VLZ-PRO!

Connections for Tracking

Versatile as these small mixers are, pressing them into 8-Track service without moving some cables between tracking and mixing is a bit of a stretch. The same concepts used in the 4-Track setup are used here—split monitoring and use of the ALT outputs.

Monitoring Connections

To reduce the amount of cable swapping while tracking, we've connected the recorder outputs to the four pairs of stereo line inputs and assigned them to the MAIN L-R bus (MUTE/ALT switches *up*). This will allow you to hear all 8 tracks while tracking, but with a bit of a compromise.

Since the stereo inputs are hard panned left and right, anything recorded on odd numbered tracks will appear in the left speaker (only); even numbered tracks will appear in the right speaker only. We'll fix that when mixing, but for tracking, it's the best we can do if we want to keep the maximum number of sources connected to channel inputs.



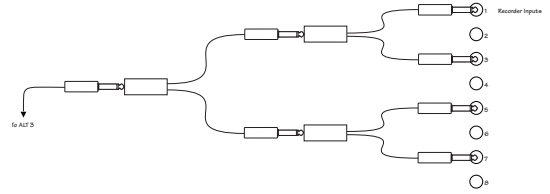
You'll need to use *some* of the stereo inputs for monitoring, but if you have only one or two sources for recording, or you're recording only a single part at a pass and you don't mind sharing an input or two among mics, guitars, keyboards, etc., you can move some of the recorder returns to line inputs of the mono channels, giving you full panning capability on some of the tracks while overdubbing.

Recorder Inputs

This diagram shows two tracks fed from the ALT OUTPUTs and the remainder of the tracks fed from INSERT outputs. You can tailor this to suit your own setup. See the 4-Track Hookup for some of the details and tradeoffs.

Having the ALT outputs connected to the recorder inputs will prove handy when bouncing tracks or if you've set up six mics on the drum kit and want to record two

tracks of drums. Using Y-Cables to split the ALT outputs to all odd and even numbered tracks (you'll need to Y the Y's) allows you to quickly route any input to any track.



Connecting an INSERT OUT directly to the recorder input gives the cleanest signal path from the mic to the recorder. Feel free to unplug and move cables—that's why they put connectors on them. If you're recording several vocal tracks using the mic connected to Channel 1, move the plug from track to track at the recorder end. Better than that, put a patchbay between the mixer and recorder and just move a cable on the patchbay.



Line level sources such as keyboards can be connected directly to the recorder inputs for tracking as long as the source can provide the proper record level. Certain sources may require additional gain from the mixer. Don't forget the **Level-Setting Procedure** when using the mixer as a preamp.

Connections Not Shown

Effects such as reverb can be connected to AUX SEND and RETURN jacks as shown in other applications.

Processors such as compressors or equalizers can be connected in line with the recorder inputs as shown in the 1202VLZ-PRO 4-Track Recording setup, or connected to the INSERT jacks with an "Insert" cable if the INSERT outputs aren't being used for direct outputs.

Control room speakers and headphones are connected to their respective outputs as shown in other setups.

Making Tracks

Refer to the 4-Track Recording setup using the ALT OUTPUTs for details. Monitor the stereo channels while tracking by assigning those channels to the MAIN L-R bus (MUTE/ALT switch *up*). With the recorder set to the Auto Input Monitor mode, you'll hear the input to any recorder track that's

record-enabled, and the playback from any recorded track.



Remember that the recorder is part of the monitoring path. In order to hear the input signal in the phones or control room monitors that you're sending to a track, you *must* enable Record on that track.

Input Assignments

For tracking, inputs are assigned to the ALT OUTPUTs (MUTE/ALT switch *down*) regardless of whether you're recording from the ALT or INSERT outputs. This is so what you hear in the monitors is only what's coming back from the recorder, not what's coming into the mixer from the recording source.



Feedback Alert! Don't forget to switch off the control room speakers and monitor on headphones when recording with live mics in the same room.

Bouncing Tracks

With eight tracks available, you'll have less need to bounce tracks than with a 4-Track recorder, but you'll need more tracks sooner than you think. See the 4-Track setup for a description of the procedure.

Panning

There's one big difference between the 4- and 8-Track setups. With only four recorder tracks to return to the mixer, we were able to employ the stereo mixer channels in their mono mode and could freely pan and assign our recorded tracks when making a bounce. Since all the recorder outputs in this 8-Track setup are returning to stereo channels as stereo pairs, you can't mix odd and even numbered tracks together and send them to one of the

ALT outputs. Nor can you pan six tracks throughout the left-right field and record that as a bounced stereo pair. The PAN pots on stereo channels don't allow this. The 1402 has a few more inputs than the 1202 so it'll take you longer to find yourself backed into this corner, but it's bound to happen some time.

The solution is to temporarily unplug some of the sources connected to the mono channels and connect the recorder returns in

their place. Tracks recorded as pairs and which will remain panned fully left and right in the mix can remain connected to the stereo inputs, but those which you want to pan somewhere in the middle must be re-patched to mono channel inputs.

Remember that a stereo channel with a plug in the LEFT input only becomes a mono channel with full Left-Right panning. If you've filled Tracks 1-7 and intend to bounce those to Track 8, you need only pull the Right inputs to the first three stereo channels and plug them into mono channels. You'll also need to pull the Left input to the 4th stereo channel (that's the one coming from Track 7) and move that over to a mono mixer channel. Here's why:

You need to be able to hear the mix of tracks you're bouncing, so Track 8 must be assigned to the MAIN L-R bus. But since you want Track 7 in the bounce mix, you can't assign it to ALT 4 bus if it's "input mate" is assigned to L-R.



While you're in the neighborhood, move the Track 8 output cable over to the now vacant Left input of its stereo channel so you can pan it to the center.

Once you've re-patched, assign the four mono channels and the first three stereo channels to ALT-3/4 by pressing the MUTE/ALT buttons. Turn the PAN controls fully right. The 4th stereo channel (which now is receiving only the Track 8 signal) remains assigned to MAIN L-R.

Housekeeping

After the bounce pass is completed, to return to tracking, put cables back as they were. Remember to return the stereo channels to their MAIN L-R bus assignment.

Since you'll be moving cables a few times during this procedure, it really helps to label them. Neatly printed labels secured to the 1/4" phone plugs with clear tape will save you from an unplanned tour of the spaghetti factory.

Bounce Variations

In the next pass, we'll bounce Tracks 1-6 (and maybe even Track 8) to Track 7, so we'll do some similar cable juggling only in this case leaving the Track 7 cable plugged into the stereo channel input alone. Assigning tape return channels to ALT-3/4 and

panning full left will send the bounce mix where you want it.

If you have tracks which you want to bounce to a stereo pair, assign them to the ALT busses, monitor the pair of recorder tracks to which you're bouncing (assigning them to the MAIN busses), and adjust the PAN controls of the tracks being bounced so that you hear the stereo positioning that you want to appear in the final mix. A typical application would be that you've recorded your drums on several individual tracks and now want to use those tracks for other purposes, but want to keep the drum recording in stereo.

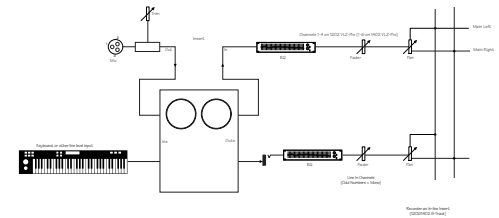
Live Multitrack Tracking Variation

Say your band's got its act together and you're not interested in a lot of fancy overdubbing and track bouncing, you just want to capture eight sources direct to eight tracks and mix afterward. Here's a really neat trick.

Remember that a channel INSERT is both an output and an input. By using the channel inserts of the first four (or six on the 1402) mixer channels, you have clean mic preamp outputs with which can feed recorder tracks directly. You *also* have four (or six) mono inputs for the recorder returns for monitoring and mixdown. With the recorder connected as if it were an in-line effect processor, the TRIM control serves as a record level control, and the rest of the channel controls operate on the recorder return signal.

When tracking, with the recorder in Auto Input Monitor mode, you'll hear the mic input when recording (or in Record Ready). When playing back, the track will go through the channel EQ and fader, as well as be available to the effect sends.

A typical setup might be with four mics going to tracks 1-4 straight from the Insert Sends, and four line level sources (keyboards, guitar processors, or outboard mic preamps) connected directly to the recorder inputs for tracks 5-8. With recorder outputs 1-4 connected to mixer channels 1-4 through the insert returns and tracks 5-8 connected to odd-numbered stereo line inputs, you'll have full pan, EQ, and level control of all eight tracks for both monitoring while recording and for mixdown. Here's a simplified diagram of this setup



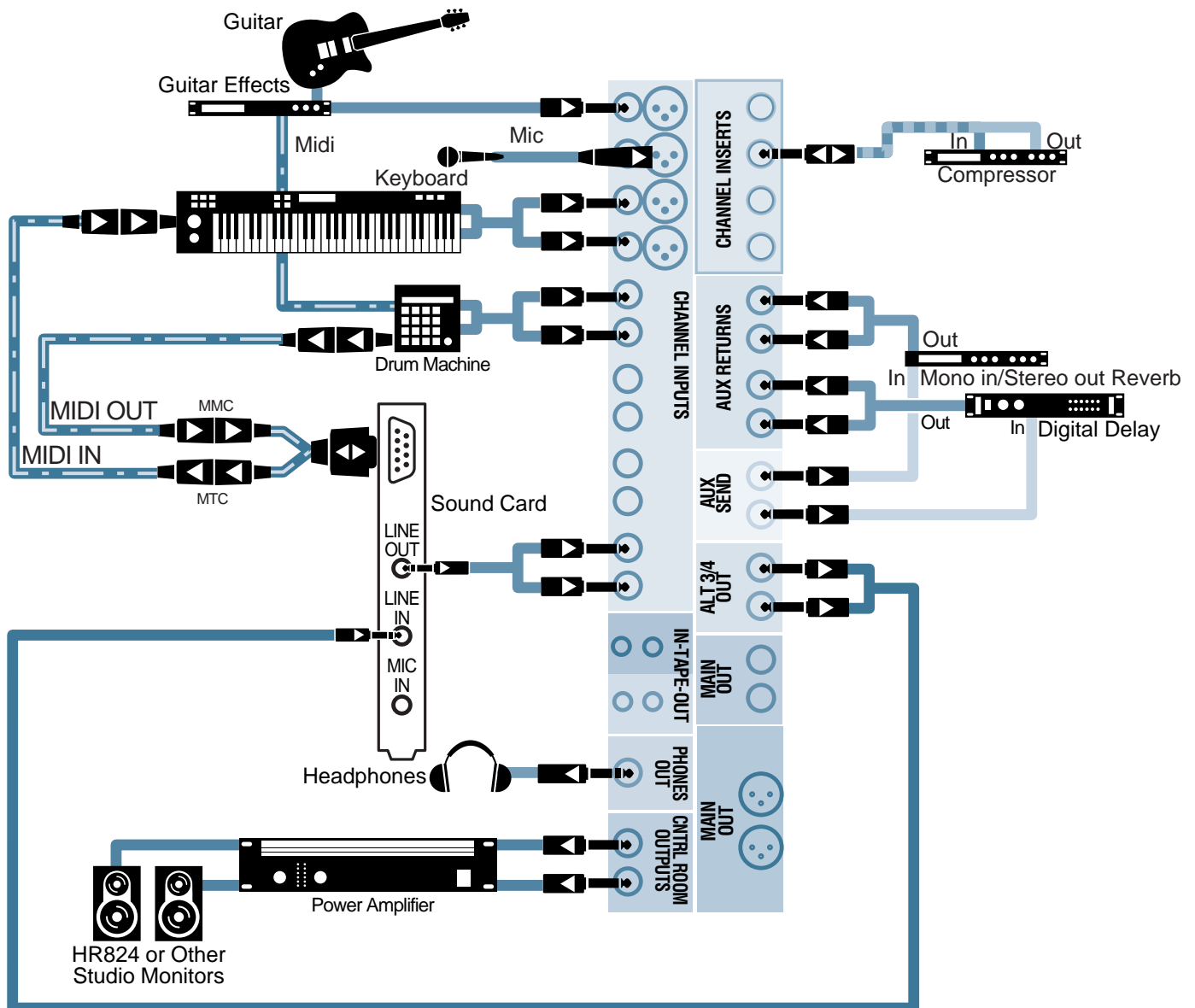
Since the insert sends and returns are unbalanced, if your recorder uses balanced inputs and outputs you may need some adapters on the recorder end. You'll have to do some patching if you want to bounce tracks. This same arrangement also works just fine for larger mixers, such as the 1604VLZ-PRO.

Mixdown

Once you've recorded all the parts, re-patch once more so that the recorder returns are connected to mono inputs, either on the mono channels or the Left input of stereo channels. Assign all the recorder returns to the MAIN L-R bus, and mix to taste.

The 2-track mixdown recorder is shown connected to the TAPE OUTPUTs where it will receive the stereo mix. It could also be connected to the MAIN OUTPUTs if they better match the operating level and/or connector type. Listen to your mix by selecting TAPE as the C/R SOURCE.

Chapter 13: I202VLZ-PRO and I402VLZ-PRO Computer Sound Card/MIDI Setup



The 1202 and 1402VLZ-PRO mixers make managing a computer-based audio and MIDI system easy. In this setup, we've shown a computer with a sound card and MIDI interface combined with both MIDI and audio sound sources. The computer serves as the main audio recorder as well as the MIDI sequencer. This setup has a lot in common with the 1642VLZ-PRO MIDI Studio, so refer to that section for details.

Connections

Stereo and mono sound sources are connected to mixer channel inputs. The ALT 3-4 outputs are used to feed the computer sound card. Sound card playback is through

a stereo pair of LINE INPUTs. An optional mixdown recorder such as a DAT or CD-Recorder can be connected to the TAPE our MAIN OUTPUTs.



The sound card outputs can also be monitored through the TAPE INPUT rather than through a pair of channels assigned to the MAIN mix if you find it difficult or impossible to control the "input monitor" function of your sound card. Connect the sound card outputs to the TAPE inputs and press the TAPE button in the CR/SOURCE group to hear playback from the computer. You'll still be able to monitor the sound card output along with sequenced MIDI synths

which are not recorded, but played as *virtual tracks*.

In this example setup, the keyboard is used as the MIDI master controller for sequencing, and the computer's MIDI OUT feeds the drum machine, the keyboard, and the guitar effects processor. The MIDI connection to the guitar processor allows the sequencer to switching to the appropriate patch on cue, freeing up your hands or feet. It's optional, of course.

Real and Virtual Tracks



You can record the audio outputs of your MIDI synths on to the computer disk or leave them as *virtual tracks*—your choice. A virtual track is one that's controlled by the sequencer. You can hear the synth tracks play, but they aren't actually recorded as audio until the final mixdown. Virtual tracks were pretty important for achieving a clean recording when working with analog tape (particularly cassettes), but digital recording has eliminated the primary reason for leaving recording of the synth sounds to as late in the project as possible. It's your choice, and it will be largely dictated by your software, how well it integrates audio and MIDI recording, how many audio tracks it can handle, and how easy it is for you to operate.



When MIDI instrument outputs are assigned to the MAIN bus, as long as they're receiving MIDI data, you'll hear them playing even if the audio has not yet been recorded on disk. Once you've recorded audio from the MIDI synths to disk, be sure to MUTE the synth channels on the mixer when listening to playback from the computer. Since the computer will be spitting out MIDI data when it's playing, there's a possibility of hearing the same MIDI sound twice—from the computer audio playback and again from the synth directly. Since response times of the audio and MIDI paths are different, you'll hear an "echo" or perhaps "phasing"—rarely desirable.

Recording

To assign any input source to the computer sound card, send it to the ALT 3-4 BUS. (MUTE/ALT button *down*)

To hear playback from the computer, bring up the faders on the stereo channel which is connected to the sound card outputs. (11-12 in this example) With the proper sound card monitor settings, you can do most of your monitoring while tracking by listening to these inputs without the need to switch monitoring sources.

Overdubbing and Mixing

Since there are only two tracks coming out of the sound card, you'll be doing your mixing in the computer, and that mix is what you'll monitor while overdubbing. Consult your software manual for specific instructions.

If you're using MIDI controlled synths as virtual tracks, you can control the balance between those synths and the recorded audio from the computer by using the mixer faders. In this case, you're mixing to the MAIN outputs, where the mixdown recorder is connected.

Input Monitoring While Tracking

We've been steering you toward your sound card software to send the input signal to the output mix so you can hear yourself while tracking, but there's another way you can monitor which may be easier to work with in your software.

All multitrack audio recording software has the ability to turn off a track in the output mix, even if it's only to turn that track's volume down all the way. Usually there's a Mute button or check box. Find it.

So we have several tracks recorded, you have them mixed, at least roughly at this point, and the mix is coming out the sound card's stereo output and into the mixer—either to a pair of line inputs which you've assigned to the L/R MAIN MIX. With the MAIN MIX button in the CR/SOURCE matrix pressed, you'll hear that sound card mix playback in the headphones and control room monitor speakers. Just like normal.

Now the cool thing about these CR/SOURCE buttons is that you can press more than one at a time, combining sources into the monitor section of the mixer. Remember the source we're recording (the mic, the guitar, whatever) has been assigned to the ALT 3-4 bus? Well, there just happens to be a button for monitoring the

ALT 3-4 bus. Press that button together with the MAIN MIX button and now the sound card tracks (with the input source muted) will be mixed in the headphones with the input source coming from the ALT 3-4 bus.

If you want to tweak the balance in the headphones between the previously recorded tracks and the new track you're recording, it's pretty simple. Just remember that the record level going into the sound card is set by the fader or gain knob on the input channel, so once you've established that setting, leave it alone and use the faders on the sound card output channels to adjust the balance between the new and original tracks.

If you've opted for connecting the sound card outputs to the TAPE inputs, you can use this same technique (selecting TAPE rather than MAIN MIX as one of the CR/PHONES sources) only you won't have control over the balance between the new and original tracks since the Tape input is at a fixed (unity) gain.

How Latency Affects Monitoring



Aside from not having to fool with your software quite so much, there's another good reason for using this method for monitoring your inputs when recording with a sound card. When you're singing and listening to yourself on headphones, there are actually two sound paths between your vocal cords and your eardrum. The first is through the headphones, the other is through the holes in your head—everyone has some. It takes about the same amount of time for sound to travel through these two paths, so unless your mic or headphones are out of phase, they combine in phase at your eardrum.

But along comes Mr. Latency, delaying the headphone sound by a millisecond or two. Add a sound to itself with a small delay and you'll notch out a band of frequencies—the result is that your own voice sounds funny when you're recording but OK when you listen to the playback. This is not only confusing and confounding, but annoying when you're trying to pay attention to what you're singing. You may have noticed that effect. Now you know why it occurs. By listening to the direct sound from the mixer rather than the delayed sound from the sound card, we've solved that little problem.

Effects

We've included a couple of effect processors in this example. Their inputs are connected to AUX SENDs, and outputs connected to AUX RETURNs in the conventional manner. They're right where you expect them, convenient for use in mix-down.

When recording in this manner, since tracks are being mixed in the computer, you can't get to the effect processors to add an effect to a track when mixing, so sometimes you'll want to record a track "wet"—with an effect. If you move the processor outputs from the AUX RETURNs to a pair of Line inputs, and then assign those channels to ALT 3-4, you'll be able to add the effect in when you're recording. With practice, you'll be able to get the right amount of the effect mixed in. You can also patch the effect in line with the source, either directly (if it's a line level source) or by using the channel's Insert jack.

Recording an Effect Track

One neat trick that can give you some leeway in mixing is to record an effect track to go along with a dry track. If, when mixing, you decide that a dry track needs some reverb, you can send that track back to the mixer, turn up the AUX send that's feeding the reverb unit, and record only the output of the reverb on a new pair of tracks.

How? Patch the reverb's output into a pair of Line Inputs, and assign them to ALT 3-4. Mute all the tracks in your recording program but the one to which you want to add the reverb. Now, on the sound card return channels, turn up the AUX SEND that's feeding the reverb. You don't have to worry about how much level to send—give it plenty so you're recording a nice hot reverb-only track. Now, while playing the track that needs the reverb, record the reverb on a new track.

Now, when you mix the tracks in the computer, you can adjust the level of that reverb track just like you'd adjust the send level to a reverb processor in mixdown. Clever, eh? You can of course do this with a chorus or any other effect, not just reverb.

Today, many audio recording programs have effect plug-ins so you don't have to do this, but it's nice to know how.

A Last Word about Sound Cards

The idea of using a computer as a recorder, particularly a multitrack recorder, is very seductive. It's an inexpensive approach that's capable of producing very high quality audio. But working with a computer, sound card, and mixer in a multitrack recording and overdubbing environment can be very tedious and, at times frustrating. Doubly so if you're making a move up from a nice, secure and reliable 4-track cassette recorder/mixer. There will be times when you'll want to throw the darn computer out the window, and your Mackie mixer along with it.

What's the problem? Well, aside from the fact that you have to get the recording software working reliably, you may find it awkward or next to impossible to follow our instructions for monitoring. If you've read any of the discussions here about multitrack recording, you'll know that we've stressed the importance of monitoring recorder returns, and the use of a recorder's automatic switching from input monitoring to playback that facilitates that.

Well, switching the sound card's output from input to playback is the responsibility (and privilege) of the card's driver software, not the computer, the recording program, or even the Mackie mixer. Until the ASIO driver specification was developed, there was no way for the driver to know whether the program was recording or playing, so it didn't know when you wanted the input signal or the playback to appear at the sound outputs.

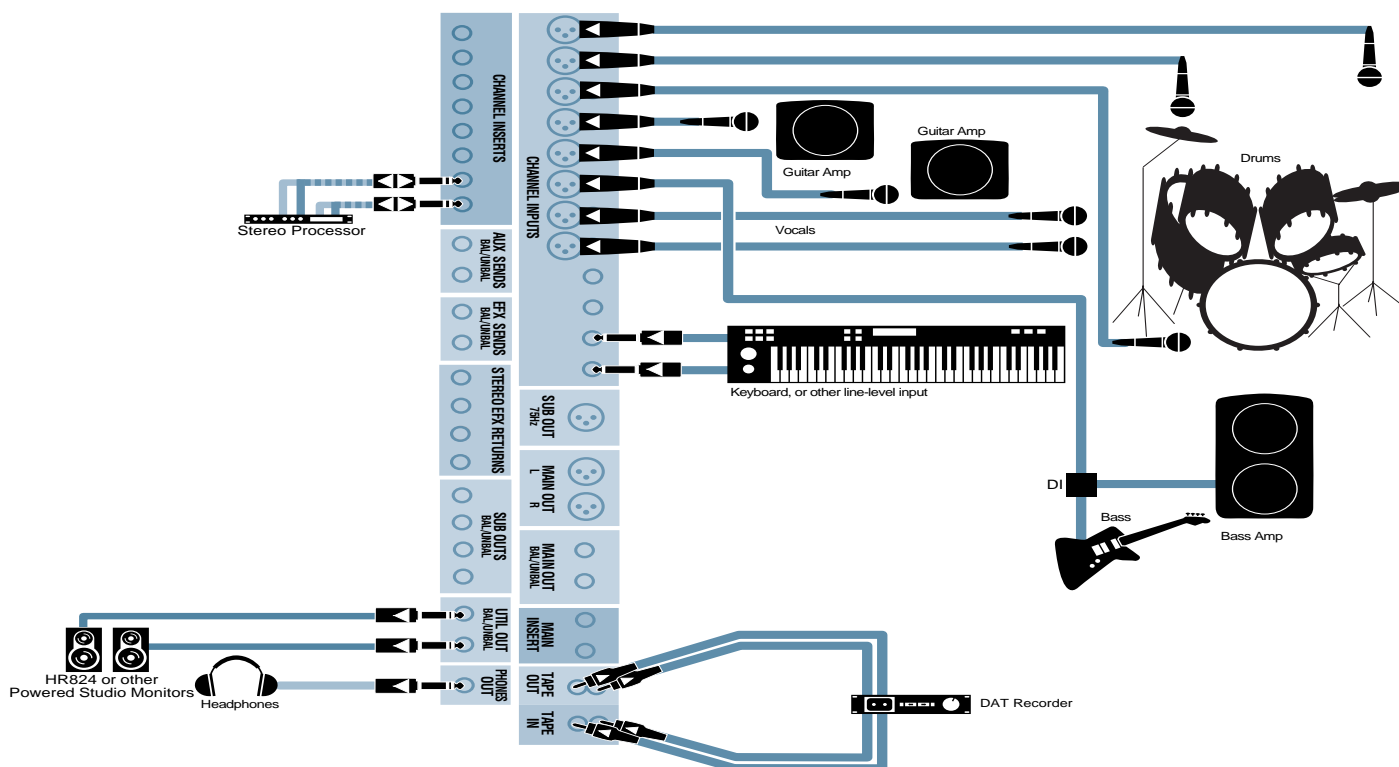
All sound card drivers provide an on-screen control panel to switch the output to input monitoring, but you have to find it, learn how to use it, and when overdubbing, you'll have one more mouse click to deal with. You'll have to fill in that step yourself. There's nothing that Mackie can do to make this any easier. We've provided the best general instructions we can, but your mileage can, and will vary.

As more modern audio software is written and computer audio hardware evolves, this function will become as transparent on a computer workstation as it is with a dedicated hardware recorder.

The good news is that when using the computer as a two-track mixdown recorder and

editor, life is beautiful. Take the path of multitrack recording with a stereo sound card in small steps and make sure you understand the operation of both your recording program and the sound card's mixer/control application. It will help you to understand how to relate our operating guidelines to your particular hardware and software combination.

Chapter 14: CFX Live Stereo Recording



So your band really sounds great playing live, and you want to put together a powerful demo from recordings of your rehearsals. You already own a CFX mixer that you're using for PA at club gigs, so why not use it for recording, too? Set up the band as you normally do for a gig, plug in your mics and a recorder, and go. Here's a setup for live, direct-to-stereo recording. Play well and you'll be surprised at how good the recording sounds.

Inputs

We've shown mics on the guitar amplifiers and drums. Go ahead and use your spiffy condenser mics overhead for the drum kit. You've got the power—phantom power, that is.

Going Direct

We're using a Direct box (DI) for the bass. A DI has two outputs—a mic-level output which is connected to the mixer, plus an instrument-level output for the amplifier. Direct recorded bass is usually very clean and solid, but experiment with the sound you get with a mic on the amplifier. You might prefer that, or like the sound of the combination of a mic and DI.

Some bands use a keyboard amplifier, others plug the keyboard directly into the mixer and send it straight to the PA. We've taken the direct approach here, but there are some considerations which we'll get into in a couple of paragraphs. Stay tuned.

Assignments

When recording a small group such as this, there's probably no need to separate the band using subgroups. You'll set up a good mix during a run-through and, unless you have the luxury of having an engineer to mix while you're playing, settings will remain fixed. On the CFX, however, in order to get a signal from the input to the MAIN OUTPUTs, it must be routed through a subgroup.

Assign all the input channels to 1-2, then in the Master section, assign SUB 1 to LEFT and SUB 2 to RIGHT. Set the SUB 1-2 faders to their Unity Gain position.

Recorder Connections

We've shown the recorder connected to the TAPE outputs. This is preferable to using the MAIN outputs, which go through the Graphic Equalizer. That equalizer is useful to fine-tune the PA to the room acoustics, but there's no need for it when recording. Alternately, you can connect the recorder

to the balanced 1/4" TRS SUB OUTPUTs 1 and 2.

The recorder outputs are connected to the TAPE IN jacks so that you can listen to the playback by pressing the BREAK switch.

Monitoring

The UTILITY OUTPUTs are convenient for feeding the control room monitor speakers for setup and playback. Since these outputs have their own OUTPUT LEVEL control, you can turn off the speakers when you're recording in the same room.

Hearing Yourself

In a live gig, there are almost always monitor speakers so that the singers and unamplified instruments such as directly-connected keyboards can be heard over the guitar amplifiers and drums. In the studio, monitor speakers are usually replaced by headphones so that the sound from the monitors isn't picked up by the mics, compromising the recording.

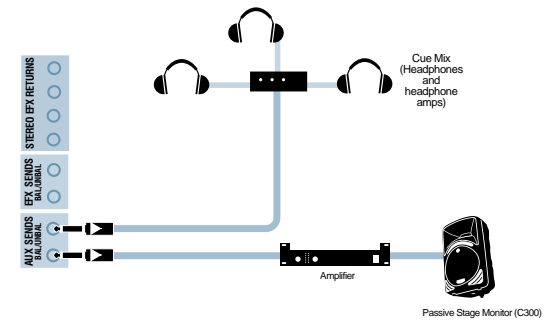
It's possible that the keyboardist will be able to hear as much as he needs to hear simply by plugging a set of phones into the headphone output of his keyboard, hearing the rest of the band from what leaks into the headphones acoustically (or keeping one earphone off). The singers will most likely find that they can't sing well without monitors of some sort.

The AUX SEND 1 and 2 outputs are available for monitor outputs. One option is to use a stage monitor just as you would at a gig. Keep the volume down and with careful placement of the monitor relative to the vocal mics, sound leakage from the monitor speaker into the mics can be kept to a workable level.

A more studio-like option is to connect a headphone amplifier to one (or both, if you want two different headphone mixes or one stereo mix) of the AUX SEND outputs and give headphones to those who need them.

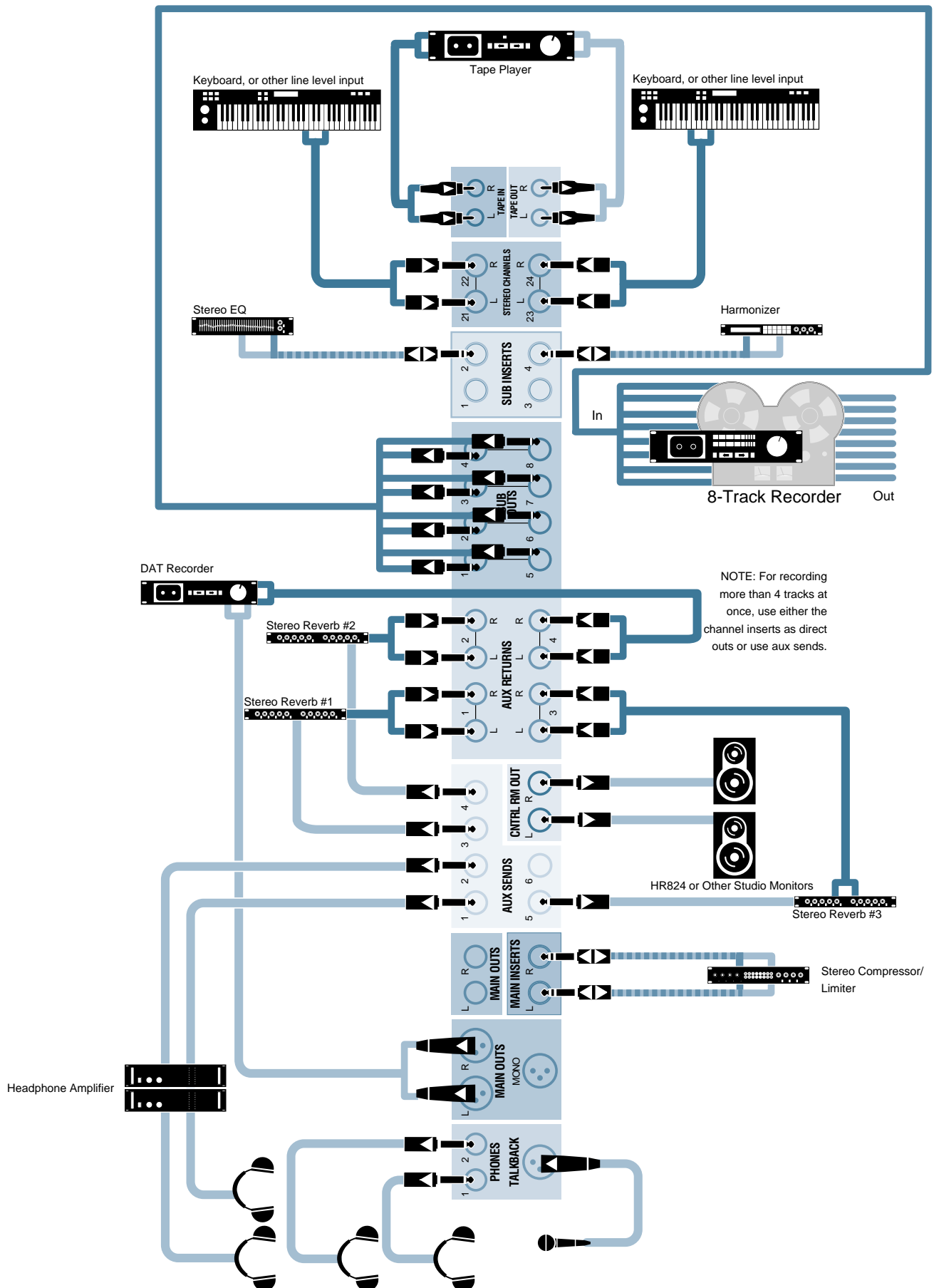
Adjust the speaker or headphone monitor mix by using the AUX 1 and 2 controls.

Effects



In this example, we haven't connected any outboard effects, but are using the built-in EMAC effect processor. If you desire to use a second effect processor, for instance for a different reverb sound on drums than on vocals, it can be fed from EFX SEND 1 and returned to the mix through the EFX RETURN 1 jacks. This will retain the use of the EMAC processor on EFX 2.

Chapter 15: SR Series 8-Track Recording



You probably purchased your SR-24-4 or SR-32-4VLZ-PRO to use as a PA console for your band, but it makes a fine recording console as well. In this setup, we've shown it used together with an 8-track recorder, with a computer equipped with a sound card serving as a mixdown recorder and editor. A secondary recorder (DAT, cassette, or CD-R) is also included for the purpose of making copies of your edited master recording.

The console is set up for split monitoring with the multitrack tape returns connected to the last eight mono channels (13-20 on the 24 input model, 21-28 on the 32).

Refer to the 1604VLZ-PRO 8-track recording application for a blow-by-blow description of the tracking and mixing process. Other than the names and location of the controls, it's all applicable here.

Inputs

Plug in your mics or line level sources as required for your recording. For live tracking you have up to 20 mic or line level inputs available on the SR-24-4, and up to 28 on the SR-32-4, plus the two stereo line level inputs. In this setup, we've shown two stereo keyboards connected to those stereo line inputs, leaving the mono mic/line inputs free for mics, instrument processors, or direct boxes.

In this example, we've used stereo AUX RETURNS 4 as inputs from the auxiliary 2-Track tape deck.

Outputs

The SR series provides two jacks for each SUB output. (see "Double Bussing" for details) We're using the SUB outputs to feed the tape deck, allowing you to record up to four tracks in a pass, with each track being either an individual source or a mix of input channels.



To record a mix of several inputs (drum mics for example) to a pair of tracks, assign those drum mic input channels to one pair of SUBs, adjusting the level and panning on the input channels for the desired balance and position.

To record more than four tracks in one pass, re-patch the 8-track recorder inputs to channel INSERT outputs using an unbalanced

(TS) plug inserted fully into the INSERT jack. This will mute the input in the mix but send the channel signal to the recorder.

We've shown a computer with a sound card connected to the TAPE OUT and TAPE IN jacks. This serves as the master stereo recorder and is used for mixing or, if you choose, recording live, direct to stereo. The auxiliary 2-track recorder is connected to the MAIN outputs. Since the TAPE and MAIN outputs carry the same signal you can swap the inputs of the main and auxiliary stereo recorders to match operating levels and connector styles.

AUX SENDS 1 and 2 feed outboard headphone amplifiers for headphone cue mixes for recording.

Assigns

Use the ASSIGN buttons and PAN knobs to route any input channel to any of the four SUB outputs for recording. See the Bus Assign section of the Grand Tour for the fifty cent explanation.

The recorder return channels should be assigned to L-R. Other than for setup and rehearsal, all other input channels should be assigned to a subgroup or un-assigned so that you hear only the recorder returns in the control room monitor.



When recording only a single mic to a track, for the shortest and cleanest signal path, connect the recorder input to the channel INSERT output.

Effects

Three reverbs are connected to AUX SENDs 3, 4, and 5 with AUX 3-4 set to PRE-FADER. Reverb outputs are connected to AUX RETURNS 1, 2, and 3. In this way, they're convenient for mixdown, and Reverbs 1 and 2 can be sent to the headphone monitor mixes using the EFX TO MONITOR 1 and 2 controls.

Mixdown

To mix your tracks, un-assign all the mic and line inputs other than the recorder returns (which should still be assigned to L-R). If you have virtual MIDI tracks, assign the MIDI sequenced synth input channels to L-R.

A computer workstation is an excellent mix-down tool for recording garage band style, where you record several takes of the same song, mix them all, or at least mix the best parts of each take, then use the computer to edit the pieces together into a complete version.

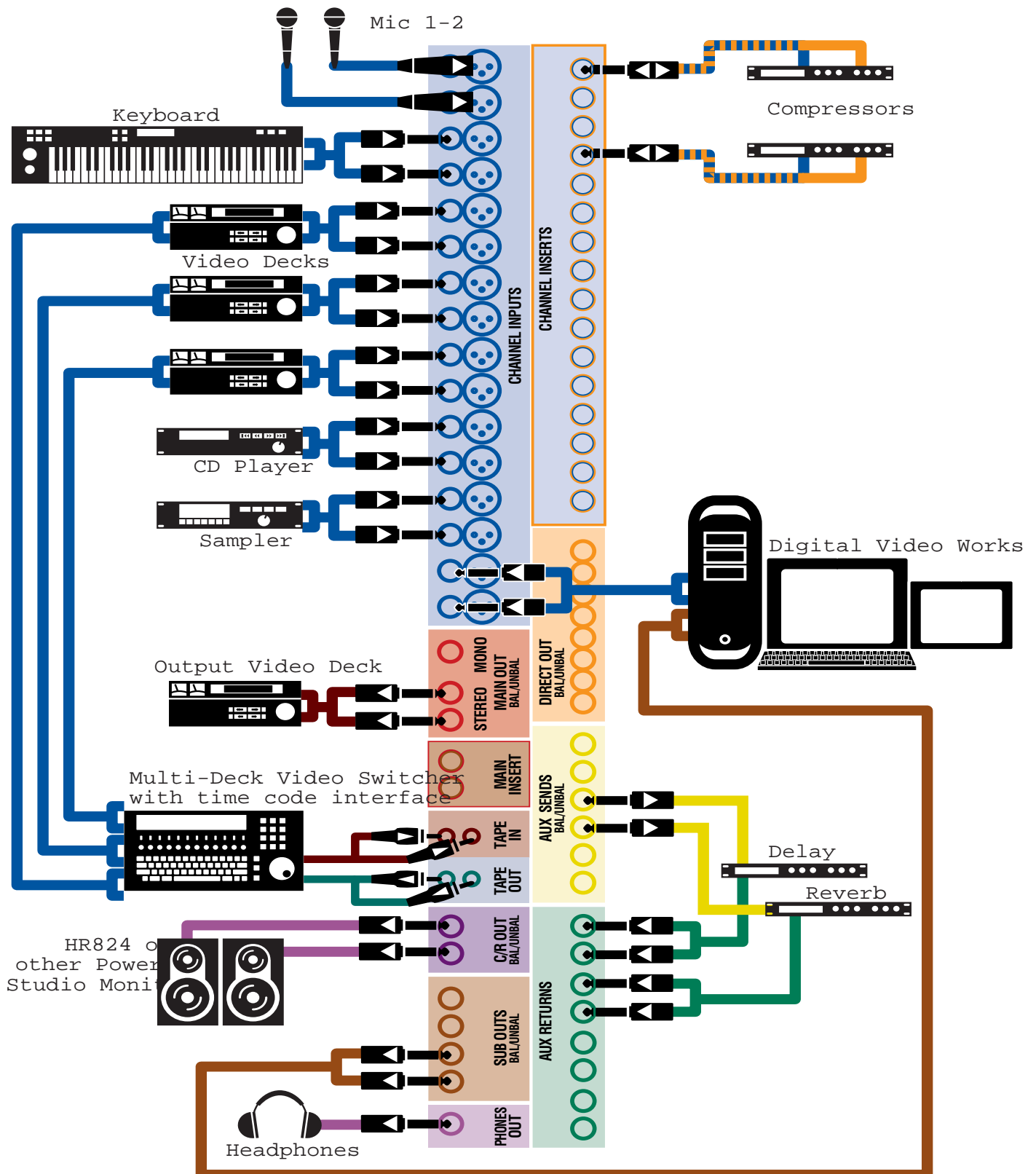
Listen to the computer sound card playback by pressing the TAPE RETURN TO PHONES/CR button. The TAPE RETURN knob will adjust the playback volume.

When you're satisfied with your editing, you can use the computer to arrange the songs in the desired order and, if your computer is equipped with a CD-R drive, make a CD directly from the edited tracks. You can also make copies to the auxiliary recorder. Route the sound card output to the main L-R bus by pressing the TAPE RETURN TO MAIN MIX button.



Be sure to disable the INPUT MONITOR mode on your sound card before assigning it to the MAIN mix. Otherwise, feedback will result. If you don't know how to disable input monitoring on the sound card (it's usually in a software control panel), simply disconnect the sound card inputs from the TAPE OUT or MAIN OUT jacks. You won't need them for playback.

Chapter 16: Video Production Setup—I604VLZ-PRO (I642, I202, I402VLZ-PRO, and 8-Bus Too!)



This hookup shows the 1604VLZ-PRO in an audio-for-video production suite. In this type of studio, most of the sources are line-level, with the occasional mic for narration. The principles illustrated here can be applied to any of the other Compact mixers.

Connections and Routing

In general, recording will be done on the Master Video recorder connected to the MAIN Left and Right outputs. Any input source intended for that destination gets assigned to L-R. The three source video decks can also record the MAIN L-R mix through the switcher connected to the TAPE OUT jacks.

The computer, equipped with a sound card, serves as an alternate recorder and provides editing capability. Its inputs are connected to SUB OUTPUTS 1-2, and its outputs come into the mix on channels 15 and 16.

Using the Computer

Typically, you'll record narration into the computer by assigning the mic input channel to Subgroups 1-2. Most sound cards are equipped with "input monitoring", but the switch is in software and you'll need to consult your sound card manual to find it. When input monitoring is engaged, you'll be able to hear what you're sending to the computer by bringing its outputs (Channels 15-16) up in the MAIN mix.



To avoid feedback when mics which find their way back to the main mix are live in the control room, kill the control room monitor speakers. Turn off the power to the amplifier, or turn the gain down. Simply turning down the CR Monitor won't do since this will also kill the headphones.

That will do for a simple setup and occasional narration recording, but here's a better way. You can avoid latency issues (see the 1202 Sound Card Recording section for a detailed discussion) when recording a narrator by using an onboard headphone amplifier fed from an AUX SEND. You'll still need to kill the control room speakers if they're in the same room as the mic, but since you're now monitoring a different output, you can simply turn down the CR/Phones volume control on the mixer.

If you're narrating to picture, by adding the Source VCR's audio to the AUX headphone mix, you can hear the audio playback from the video deck in the phones, but record only the new narration on the computer so that it can be cleaned up and edited later.

SMPTE Time Code

Time code is the glue that keeps audio and video playback synchronized. A thorough discussion of time code is beyond the scope of this book, but we'll mention a couple of things.

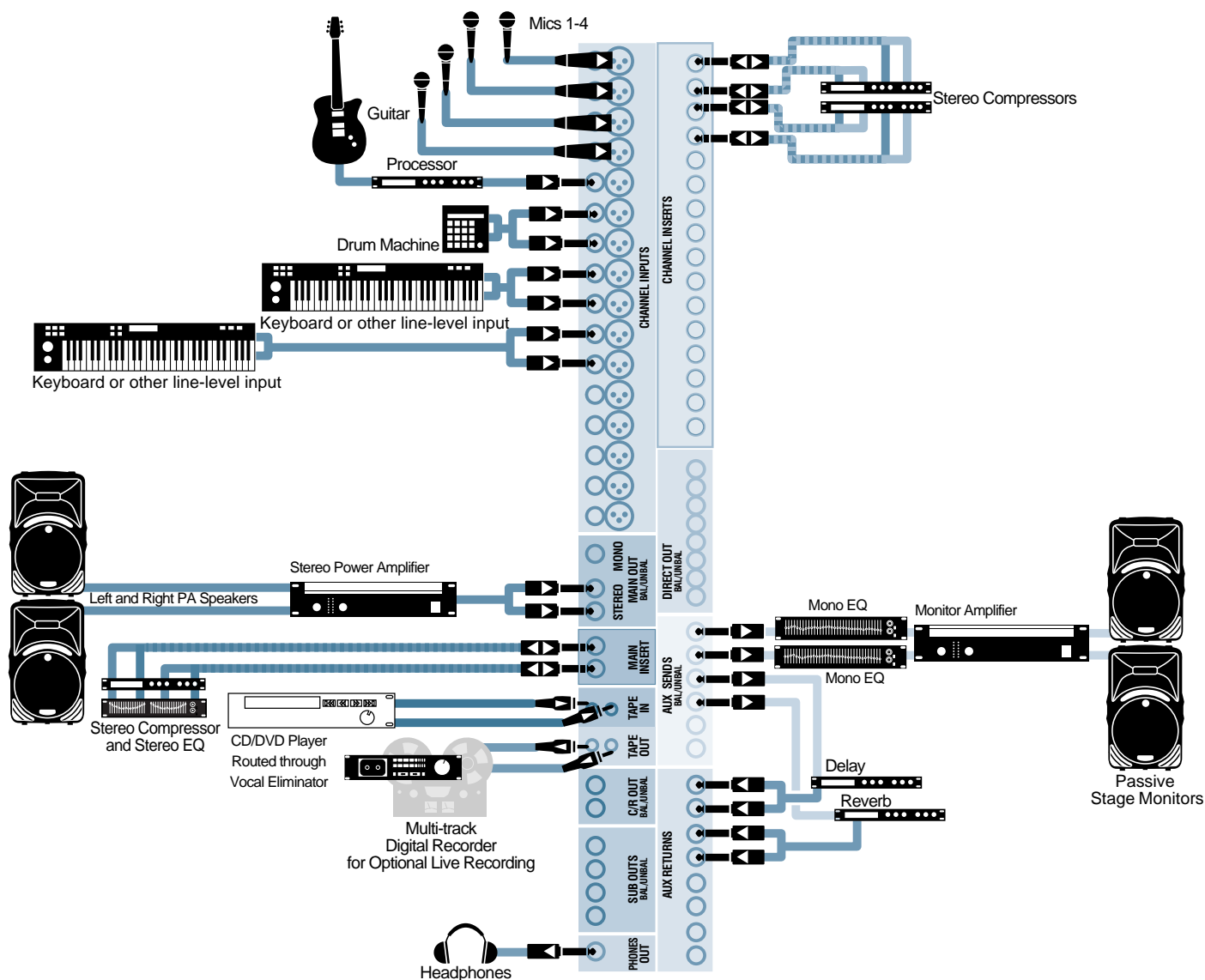
MIDI sequencing is a powerful tool for composing music for video. In this example, a multi-timbral keyboard synthesizer is connected to the computer's MIDI interface so that it can be used as an input device during composition, and a playback device when it's time to record music onto the videotape.

Most MIDI sequencers can use either SMPTE or MIDI TIME CODE (MTC) to start the sequence playing at the correct spot in relation to the video. When the time code offsets have been determined and properly set, the sequencer will wait patiently for the proper time in the video to come along, and then start playing the keyboard.

In order to keep things in sync, you must know the video frame rate of the time code source and set everything to be synchronized to this to the same rate. Typically in the US, video producers use 29.97 frames per second (that's the standard for color TV), most often with drop-frame numbering. This leaves out a few frame numbers several times an hour so the time code time will agree with clock time at the end of the hour. (it's sort of like leap year)

Music producers who don't work with video usually use 30 frames per second in the non-drop-frame format because the numbers are easier to deal with. If you're composing music on a sequencer which you intend to sync up with video, be sure to use the same time code format as is on the video tape you've probably been handed. When in doubt, ask.

Chapter 17: PA Setups: I604VLZ-PRO Stereo PA



The Setup

Sound Reinforcement (PA) systems cover a lot of ground. This setup is typical for a band that plays small clubs or lounges. We've shown compressors patched into the INSERTS for the mic channels (the vocals). They're optional of course, but if you need 'em, that's where to put 'em. There's also an optional compressor and equalizer on the MAIN outputs.

Where's The Bass Player?

More than likely he has an amplifier so big that he needs to travel in his own van and it's loud enough so that it doesn't need to go through the PA system. Or maybe the keyboard player is playing the bass part.

The guitarist is traveling light, running his guitar straight into the mixer through an effect processor that replaces his amp, so be sure to give him plenty of volume in the monitors.

Assigns, Gozintas, and Gozoutas

Main House Mix

The simplest way to run this system is to assign all the sources directly to L-R. Set the PAN controls to your liking (most likely in the center since most small PA systems are run in mono), set the TRIM controls using the **Level-Setting Procedure** and go. The MAIN MIX fader is your master volume

control. We'll talk about using the SUBs later.

Monitor Mixes

This setup shows two stage monitor mixes fed from AUX SEND 1 and 2. Each output goes through a graphic equalizer to a power amplifier, then to the stage monitor speakers. Typically, a stereo power amplifier is used for the monitors, running one mix through each channel. The graphic equalizers are for fine tuning the frequency response of the monitors and to attenuate frequencies that have a tendency to feed back when the monitor volume gets too loud (and it always does).

Press the PRE buttons on the channel strips to make AUX SENDs 1 and 2 pre-fader. This way you can have different mixes in the house and on stage. The master volume controls for the monitor mixes are the AUX SEND 1 and AUX SEND 2 knobs. To check the monitor mix in the headphones, press the AUX SEND SOLO buttons.

Effects

We've included a reverb and a delay in addition to whatever the guitarist has in his trick bag. These are fed post-fader, the reverb from AUX SEND 3 and the delay from AUX SEND 4.

The reverb signal comes back into the mix through AUX RETURN 1 and the delay through AUX RETURN 2. Start mixing with those RETURN levels set to their Unity Gain position and adjust the amount of reverb and delay in the mix by using the AUX SEND 3 and 4 controls on the channels.

To add reverb to the SEND 1 monitor mix, turn up the TO AUX SEND 1 knob in the Aux Master section of the mixer. If you want to send the same reverb to both monitor mixes, instead of using the AUX RETURN, connect the outputs of the reverb to a pair of channels and turn up the AUX 1 and AUX 2 Sends on those channels. If you're squeezed for channels, unless stereo reverb is an important effect in your house mix (remember, most small PA systems are run in mono), you can use just a single output from your reverb and connect it to one channel rather than a pair.

Singers usually like to hear some reverb in their monitors, but delay in the monitors is often confusing. None the less, if you want

to add delay to the SEND 2 monitor mix, the TO AUX SEND 2 knob will do it.

Matching the Mixer to the Power Amplifiers

The volume in the house and stage speakers is a function of the gain of their power amplifiers. In a properly matched system, when the mixer's meters are hitting 0 VU, the volume should be loud enough so that you want to turn it down. This will give you about 20 dB of headroom for transients before the power amplifier clips.

There's a detailed discussion of setting the power amplifier gain in the Gain Structure section of this book, but since you're here now, here's a brief "how to".

If you've connected the optional compressor and graphic equalizer into the MAIN INSERTS, set the compressor's threshold to maximum so that it won't try to undo your power amplifier gain adjustments. If your power amplifier doesn't have a gain control, you may be able to use the compressor's output level control to match levels.

It's OK to run the MAIN MIX fader (or AUX masters for the monitors) below its Unity Gain position if you can't turn the power amplifier down to a tolerable roar when the mixer's meters are hitting 0 VU. You want to avoid the situation, however, where you have to crank the mixer's output so that its meters consistently read *above* zero in order to get adequate volume. This will lead to the mixer clipping before the power amplifier does. Get a bigger amplifier.

Once the power amplifier gain is set for reasonable house level with plenty of headroom, you can set the compressor threshold so that it starts working a little before things get too loud. It will protect your speakers and your audience.

Intermission

When it's time to take a break, MUTE all the channels to keep stray noises out of the PA while you're away from the mixer. Since the MUTE comes after the pre-fader AUX SENDs, it's a good idea to either turn the monitor power amplifier off or turn AUX 1 and 2 SENDs all the way down.

To play a CD during the break, press the TAPE TO MAIN MIX button and adjust the playback volume using the TAPE IN control.



We shouldn't have to remind you but if we don't, the band will. Don't forget to un-mute the channels and turn the monitor sends back up when they return to the stage for the next set.

Mono PA

In small clubs, the main speakers are often fed in mono rather than stereo. The reason for this is that it's likely that a significant number of audience members are much closer to one speaker than the other, and you wouldn't want them to miss out on what's being fed to the other channel.

The 1604VLZ-PRO has a MONO output, which is an equal mix of the MAIN Left and Right channels. The MAIN MONO output has its own level control right next to its jack, but it also follows the MAIN MIX fader. Once you get the level set properly in the house, you can make volume adjustments with the MAIN MIX fader.

Connect the MONO output to one channel of a stereo power amplifier to feed the main speakers in mono and, if you need only a single monitor mix, you can use the second channel of the amplifier to power the monitors. Your roadie will appreciate not having to carry an additional amplifier.



Just be sure that your power amplifier is rated to take the load of two speakers in parallel. If you're using 8 ohm speakers, the amplifier must be rated for 4 ohm operation. Most are, but read the label.

Since the MONO output is a sum of the Left and Right MAIN OUTPUT, the position of the PAN controls is irrelevant to the MONO mix, thanks to Constant Power panning. The PAN position still functions on the stereo outputs, though, so if you're recording the PA mix from the MAIN or TAPE outputs, you'll get things placed in the stereo field of your recording in relation to their pan pot settings.

Using Subgroups

We've bypassed the subgroups in this basic application by simply assigning all the

inputs to the L-R MAIN bus. There are some instances where subgroups can make mixing easier.

Stereo SUBs

Let's say the lead singer is on Channel 1, with background vocals on Channels 2, 3, and 4. To make a background vocal subgroup pair (we're in stereo, remember?), unassign Channels 2, 3, and 4 from L-R and assign them to 1-2. Then, over in the Master section, assign SUB 1 to LEFT and SUB 2 to RIGHT. Now, SUBs 1-2 as a pair will control the level of the background vocals in the mix.

While moving two faders rather than three might not seem worth while, by using the vocal subgroups, you can adjust the volume of the background singers while still maintaining the balance you've established with the channel faders. If the channel faders were in three different positions to get the right vocal blend, adjusting the background vocal would require you to move all three together, maintaining their relative positions. With the subs engaged, you only have to move two, and they'll both be in the same relative position. Some people tie the two knobs together using a broken pencil and a rubber band. (You can order a broken pencil from the Mackie Parts Department)

The value of subgrouping becomes much more evident when working with a live drummer and you're using seven or eight mics on the drums, or when the keyboard player has a stack of keyboards that he plays at different times in the set. It's faster to have one fader to dive for when the keyboard is too loud or soft, rather than having to figure out which of the eight he's playing on at the moment.

Mono Subs

If you're running the PA in mono, you can use mono subgroups. Then you'll have four individual subgroup controls rather than two pairs.

In this example, assign all four keyboard channels to SUB 1 by pressing ASSIGN 1-2 and setting PAN fully left. Then, in the Master section, assign SUB 1 to *both* LEFT and RIGHT busses. Now you can control the level of all the keyboards using just the SUB 1 fader. Other subs could be used for drums and the background vocals.

Mono in Stereo?

An effective use of stereo PA in a small club is to run the main mix in mono, but pan the effects in stereo. This will give a sense of spaciousness while still putting the primary mix everywhere in the room. Since AUX RETURNs are stereo and go directly to the MAIN L-R bus, when using mono subgroups for keys, background vocals, and drums, you can do most of your balance adjustments right from the Master section.

Recording The Band

Live recordings are good. They'll let you hear your strengths and weaknesses, and a good live recording makes a great demo for getting gigs. Who knows? It might even become your next CD.

Recording Direct to Stereo

In this example, we've connected a recorder to the TAPE OUT jacks to record the MAIN stereo mix. This would work out pretty well in this example since every voice and instrument is going through the PA system, so they'll all end up in the MAIN mix, and the PA speakers will serve as your control room monitors.

But there may be a problem. Remember the invisible bass player we spoke of? He's got his Apocalypse Mach XXX amp cranking loud enough to give the doorman a headache, and the only way that bass will get into the recording is from leakage into the vocal mics. You have no control over the level of the bass going to tape, and unless you're very lucky, it'll be wrong.

One approach is to turn things upside down. Give up one of the effects and use one of the AUX outputs to feed a mono post-fader mix to the main speakers. That digital delay was only used on the lead vocal anyway, wasn't it? It could just as well be patched in line through the channel INSERT.

Now, you can put a mic in front of the bass cabinet (or connect the bass through a DI), run it into a spare channel, and assign that channel to the MAIN mix. Plug your headphones into the recorder's phones jack and adjust the level of bass in the recording using the fader on the bass mixer channel. Unless you turn up the AUX feeding the house speakers on the bass channel, you

won't add any bass to the PA mix. Or you can sneak in a little if it helps.

Multitrack Recording

While in principle it's pretty simple to send a stereo mix to a two-track recorder, in practice, it's difficult to get good results. Since you're mixing for the house, that mix will take into account the sound on stage—monitors, amplifiers, and loud instruments such as drums and brass, that may not need much support in the PA in order to be heard clearly by the audience.

Even if you set up the mixer for independent recording and PA mixes, it's difficult to mix on headphones. It's doubly difficult to hear the recording mix with the live sound leaking into the phones, and it's googly-difficult to mix the recording if you're playing at the same time. Capturing the elements of the mix on a multitrack recorder and mixing afterward is a good approach.

If you have as many recorder tracks as you have active channels, the simplest and most straightforward approach is to record one track per channel, connecting the recorder inputs to the channel DIRECT OUTs, or for a cleaner, pre-EQ signal path, to the INSERTs. Plug and play, just watch the levels.

Multitracking and SUBs

If you don't have a recorder track available for every channel, you can get good results by using the SUBs to make mono or stereo submixes, and record the submixes along with a few direct (one-to-one) tracks. For this example (including the loud bass player), you might arrange the tracks as follows:

Track 1—Drums L

Track 2—Drums R

Track 3—Bass

Track 4—Guitar

Track 5—Keys L

Track 6—Keys R

Track 7—Background Vocals

Track 8—Lead Vocal

Connect the recorder inputs to the DIRECT outputs of the drum machine, bass, guitar, and lead vocal channels. Assign

all of those channels to L-R so they'll go to the main PA mix.

Assign the four keyboard channels to 1-2, PAN them as you'd like them to appear in the recording, Assign SUB 1 to the Main Mix LEFT and SUB 2 to the RIGHT by pressing the LEFT and RIGHT buttons above the SUB 1 and 2 faders. Connect SUB OUT 1 to recorder Track 5 input, and SUB OUT 2 to Track 6.

Assign the three background vocal channels to 3-4, PAN them fully left (this will be a mono subgroup), and assign SUB 3 to both LEFT and RIGHT with the buttons above the SUB 3 fader. Connect SUB OUT 3 to recorder Track 7 input.

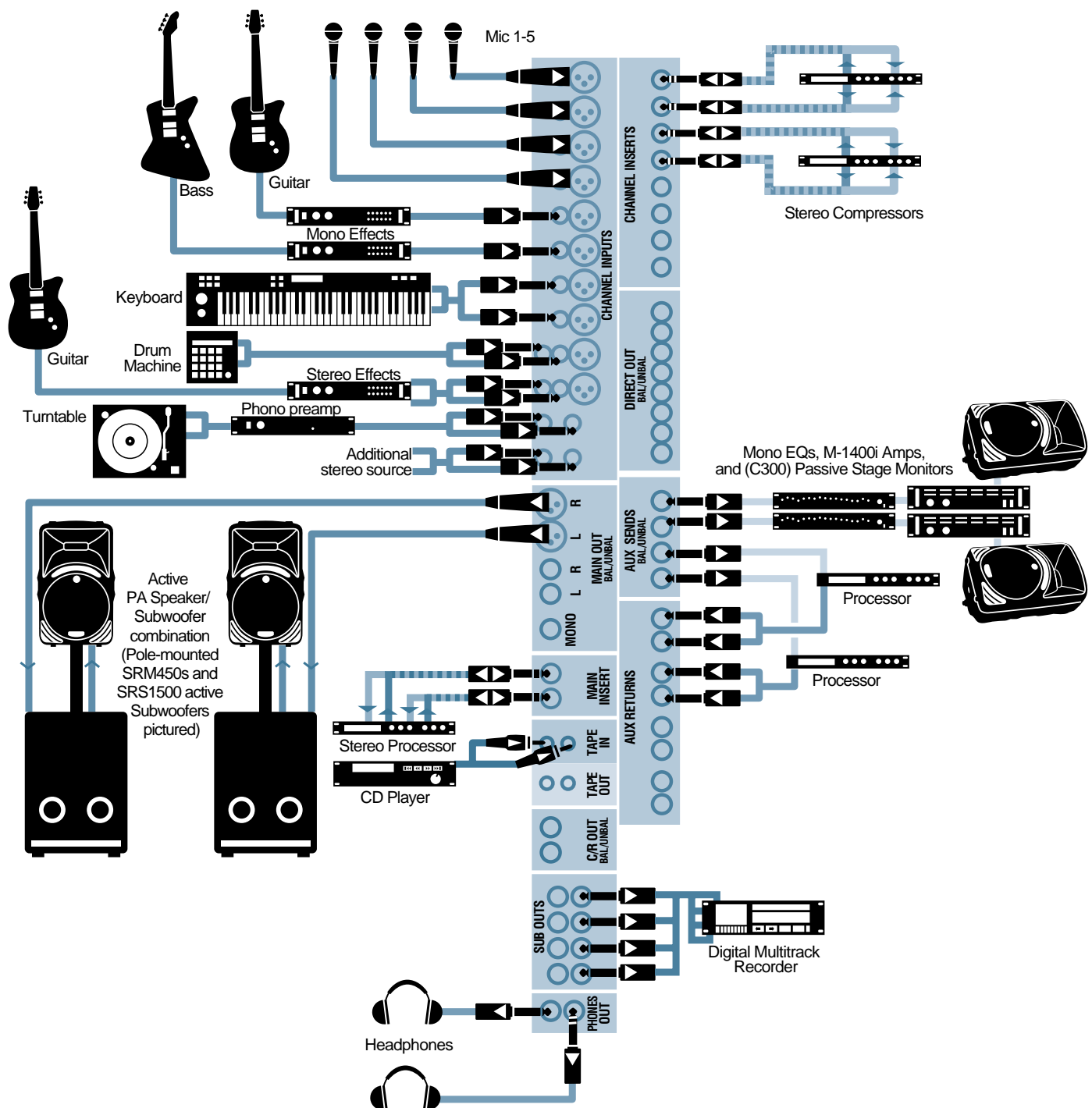
If you decided that recording stereo background vocals more important than stereo keyboards, you could assign the background vocals to a subgroup pair and assign the keyboards to a single subgroup. Or if you wanted stereo keys and vocals, perhaps you could settle for mono drums. Decisions, decisions!

Making Your Live Recording Live

Another possibility is to combine the keyboards and guitar into a subgroup or pair, juggle some other things around, and free up a pair of tracks that you can use to record the audience and the sound in the house. This is what *really* makes a live recording come alive. A promoter listening to a demo tape will appreciate hearing an enthusiastic audience response, and blending a touch of the house sound into your recording might be all the reverb you'll need.

Connect a pair of mics to an open pair of mixer channels, don't assign them anywhere, but connect the direct outputs of those channels to a pair of recorder tracks. You'll be glad you did.

Chapter 18: PA Setups: I642VLZ-PRO Stereo PA



This hookup is a variation on the 1604VLZ-PRO stereo PA application. Refer to the 1604 hookup for details.

This example takes advantage of the 1642's multiple stereo line inputs. Main active speakers with built in power amplifiers and active subwoofers are shown here, but of course you can use separate speakers and power amplifiers.

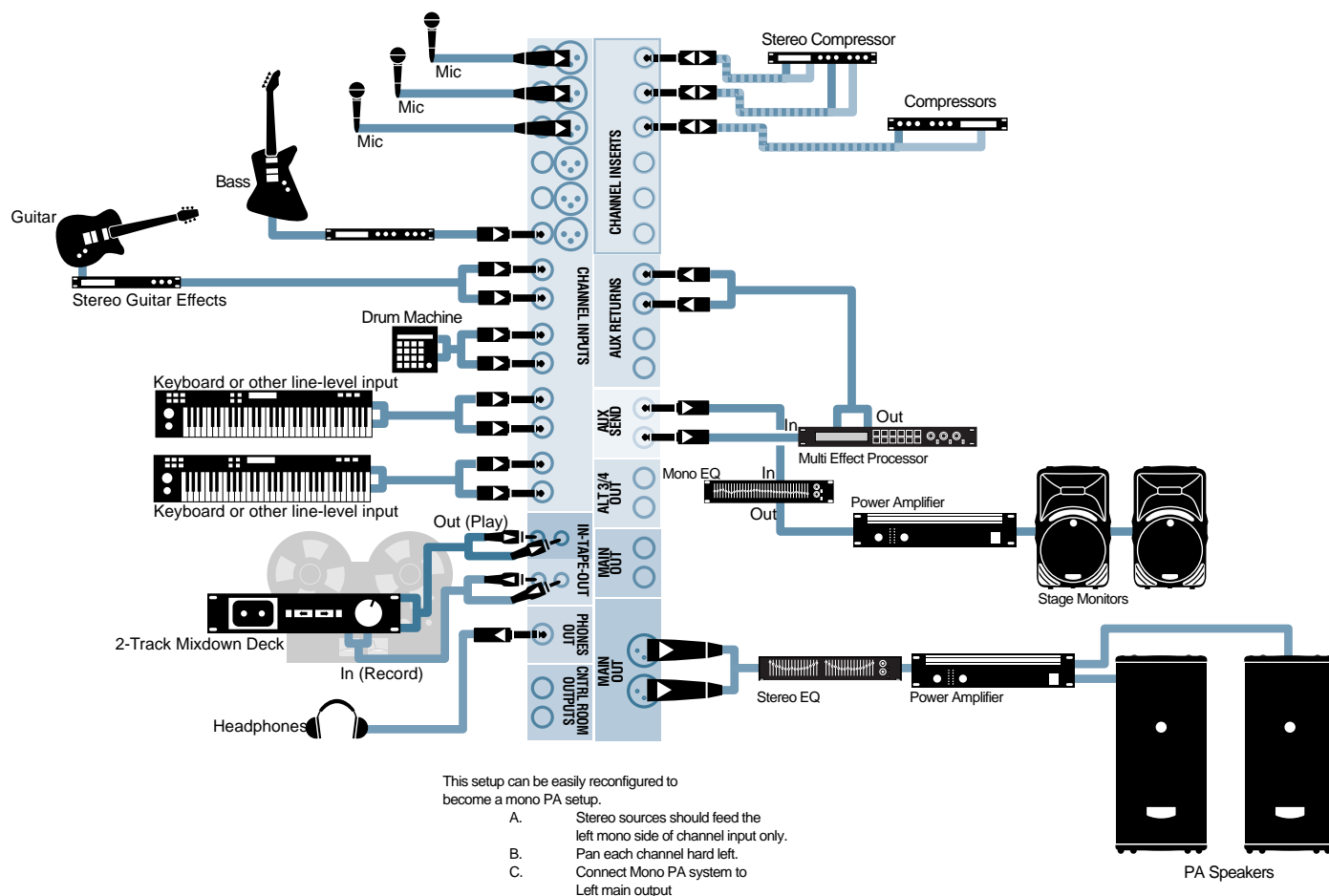
AUX 1 and AUX 2 are used for two monitor mixes. Two stereo effect processors are fed from AUX 3 and AUX 4, returning to the mix on stereo RETURNS 1 and 2. The

EFFECTS TO MONITOR control TO AUX SEND 1 sends the output of the processor connected to RETURN1 to the monitors fed from AUX SEND 1. TO AUX SEND 2 sends the output of the other effect processor to the monitors fed from AUX SEND 2.

A recorder is shown connected to the SUB outputs. If you have an 8-track (or more) recorder, you'll probably want to use both the SUB OUTPUTS and DIRECT OUTPUTS. Divide your sources up so that those connected to the stereo inputs (which don't

have INSERTs) are routed to the subgroups. Assign the subgroups to the LEFT and RIGHT busses to put them into the main mix.

Chapter 19: PA Setups: I202/I402VLZ-PRO PA



This application is very similar to the 1604VLZ-PRO PA setup. The PA is set up in stereo, but here are some tips for scaling your system down in size if that's appropriate for your venue.

Scaling Down—Mono PA

When working with a small mixer in small clubs, you may find it advantageous to operate the PA in mono rather than stereo.

A common setup is to use one channel of a stereo power amplifier for the main speakers and the other channel for the monitors. Since there's only one pre-fader AUX send on these smaller models, you're likely to be using only a single monitor mix anyway.

Mono MAIN Output

The simplest way to get a mono output to the main speakers is to pan all the input channels to one side and simply use that one MAIN output to feed the speakers. You

could also pan all inputs to the center and use either output.

You can also use a Y-Cable to combine the mixer's LEFT and RIGHT MAIN OUTPUTs (either the XLR or 1/4" jacks) without harming the mixer.

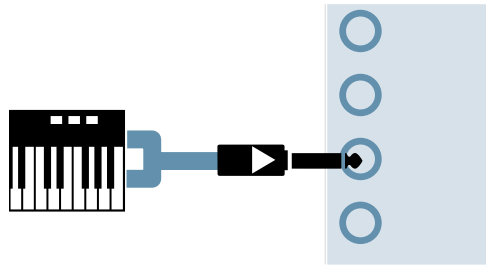
Stereo Sources in Mono

A closer look True stereo sources can be a bit of a problem. Turning the PAN knobs fully to one side will send only one channel of the stereo source to the panned output. That could mean that you won't hear one end of a keyboard or won't hear the normal balance of drums from a drum machine. Panning to the center and using only one output will provide a mono mix of stereo inputs.

Fortunately, most stereo instruments provide a mono output by plugging into only one of the outputs. Check the manual or the label on the jacks. If you can get a mono output from your stereo keyboards or drum machine, simply plug it into the LEFT input of a stereo channel. Then you

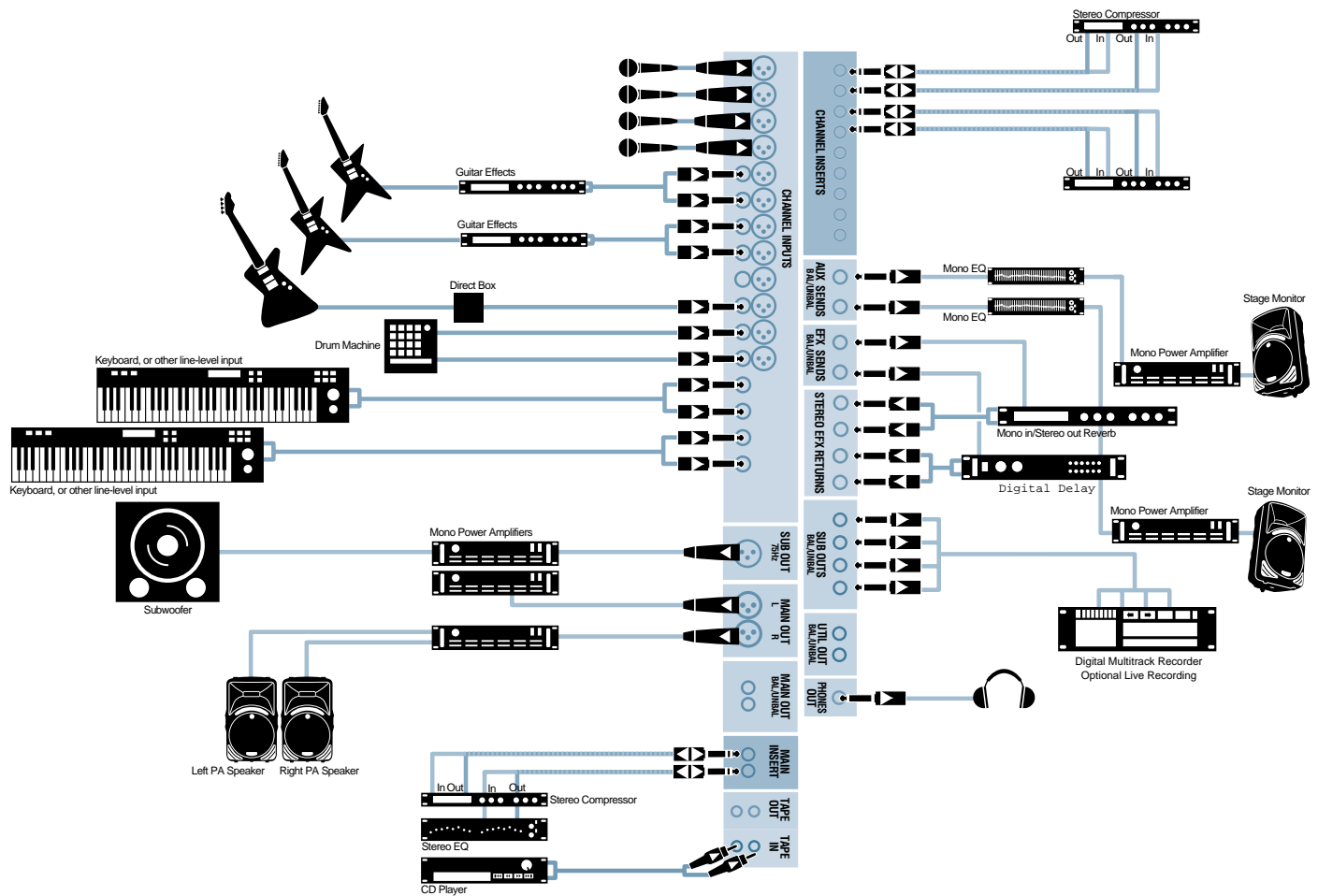
can pan the mono source wherever necessary.

If the instrument doesn't provide a mono output from one jack, you can use a Y-cable to parallel the two outputs.



Listen closely to your synth patches if you're running the PA in mono. Some synths have stereo effects or patches that rely on out-of-phase material in the two channels. They sound gloriously spacious in stereo, but the effect, or even the entire sound, can disappear when the two channels are summed to mono.

Chapter 20: PA Setups: CFX Large Club or Auditorium PA



This PA hookup is similar to the 1604VLZ-PRO example. Here are a couple of variations:

Subwoofer

A subwoofer power amplifier and speaker are shown connected to the 75 Hz SUBWOOFER OUTPUT. This output follows the level of the MAIN outputs, so use the gain control on the subwoofer power amplifier to balance the level of the sub with the mains.

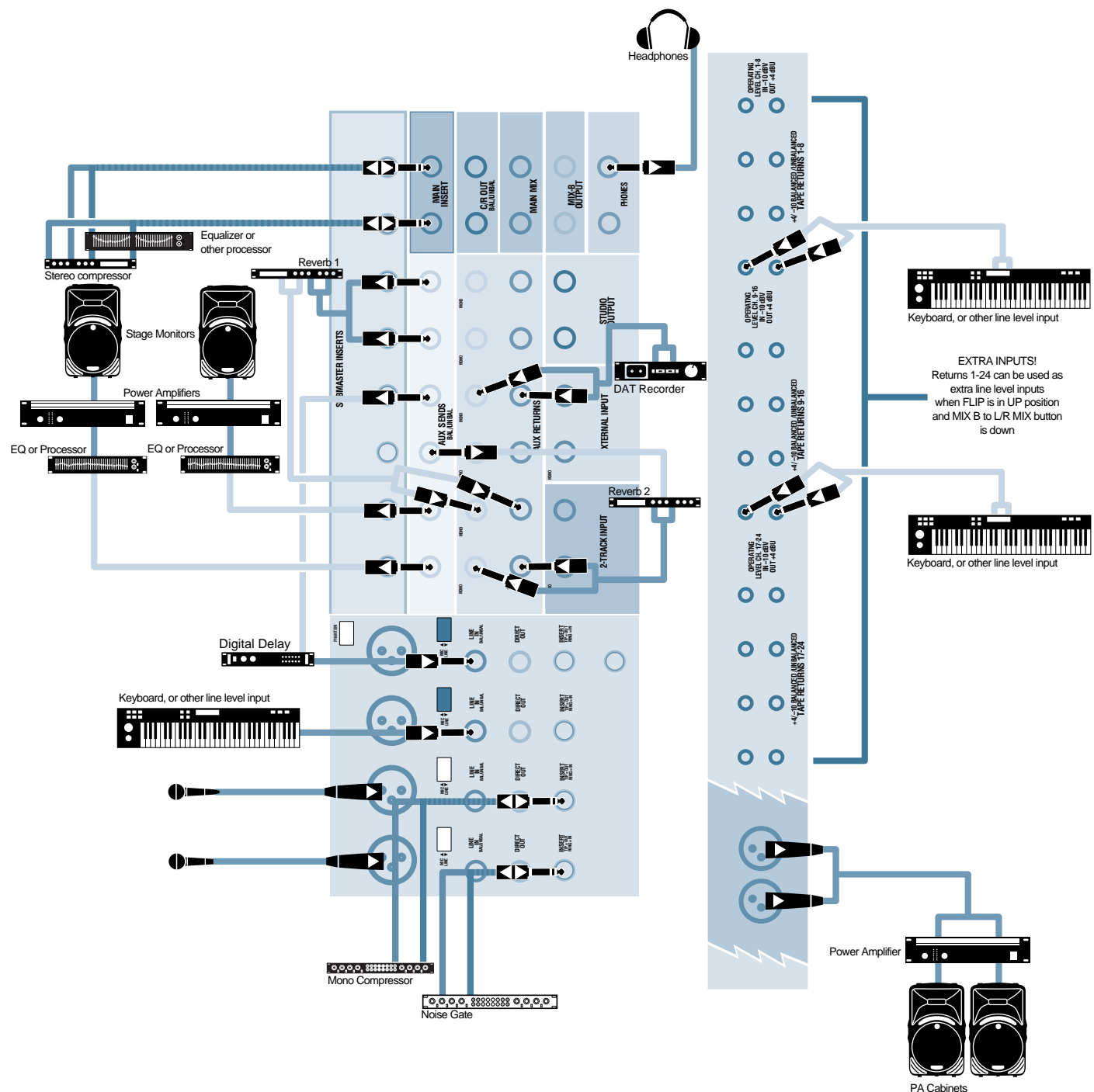
Effects

An external reverb unit is connected in the EFX SEND/RETURN 1 loop. The send level to the reverb is controlled by the EFX 1 knob in the MASTER SEND section. The amount of reverb sent to the mix is controlled by the EFX 1 RETURN knob.

Note that the Delay isn't quite plugged in. The graphic artist who drew this illustration obviously didn't care for the built-in EMAC effect processor so he bypassed it and connected an external digital delay in its place. The sound engineer who wrote this manual likes the EMAC just fine so he unplugged the outboard delay. We left it there so you could see how to hook up a second outboard effect processor if you agree with the graphic artist.

Seriously, with nothing plugged into the EFX SEND 2 and EFX RETURNS 2 jacks, the internal EMAC effects processor is ready to go to work for you. With the delay plugged in as (almost) shown, the internal effect processor will be bypassed and the external processor can be used. You might want to keep this in mind if you need an effect for one song that isn't included in the EMAC processor. Just patch it in for that one song and unplug it when the song's over to go back to the internal effect processor.

Chapter 21: PA Setups: 8-Bus PA Applications



The Mackie 8-Bus series was designed from the ground up to function both as a multi-track recording console and for sound reinforcement applications. We'll discuss recording applications at the end of this section. In SR service, it can handle front-of-house, monitors, both simultaneously, as well as setting up an independent stereo recording mix along with the standard SR chores.

In general, the MAIN L/R bus will be used for the house mix and AUX Sends will be

feed monitors and effect processors as needed.



If you've had experience with live sound mixing, you understand the importance of the pre-fader solo function—to hear what's coming into a channel before you put it into the mix for the whole audience to hear. The solos on the 8-Bus are post-fader as it comes from the factory. If you'll be using yours primarily for live sound applications, you might consider making the PFL Solo modification

described in the Modifications section of this book.

Setup

We strongly suggest that you go over the 8-Bus controls in the Grand Tour section of this book. There are some unique terms and functions that we'll be discussing here. You should be familiar with them.

Inputs

Normally inputs will be coming from mics or direct boxes, with an occasional line level source such as a keyboard or a guitar processor. To use the Mic and Line inputs, be sure the FLIP button at the top of the channel strip is in the MIC/LINE position (*up*). And while you're there, don't forget to adjust the TRIM, following the now famous **Level-Setting Procedure**.

Main Outputs

Refer to the hookup diagram on the previous page. The MAIN outputs are connected to the front-of-house speakers through the main power amplifiers. You might choose to put a graphic equalizer and/or compressor/limiter in line between the mixer and the amplifier for more control over the house mix.

Monitors

Any of the AUX SENDs are suitable for feeding the stage monitors. Juggle then between your monitor mix and effects requirements. Since the AUX 1 and AUX 2 outputs are balanced, they're better suited for the typically long cable run between the house mixing position and the stage (where the power amplifiers are usually located) than the other AUX outputs, which are unbalanced. For monitor mixing, switch the AUX 1-2 sends to pre-fader by pressing their associated PRE button.

Effects

We've shown two reverbs and a digital delay, and just to prove that there's no fixed relationship between sends and returns, we've fed Reverb 1 from Send 5 and returned its output to Aux Return 2. Reverb 2 gets its feed from Send 3 and it returns through Aux Return 1.

There's a good reason for this, of course. Sends 3 and 5 share the same knob, with the SHIFT button selecting whether the channel signal is sent to AUX 3 or AUX 5. Therefore, the third knob down always controls a reverb, and the SHIFT button selects which of the two reverbs will be used for that channel. Set one reverb unit up with a short, bright program and the other for a longer, darker program. Use the SHIFT button to select the reverb program appropriate a particular channel's source and the third SEND knob will control that channel's reverb level. No need to remember to turn one knob for the bright reverb and another for the dark reverb.

There's no particular reason why Reverb 1's outputs are connected to Return 2 and vice versa other than the artist thought it was easier to draw the picture that way. Since it's usually easier to run cables than to draw them neatly on paper, feel free to connect them "straight".

The delay is fed from AUX Send 4 and its output is returned to a channel Line Input. This allows EQ to be applied to the delayed signal, and also allows more flexibility in assignment of the delay output.

For example, you could send vocals to the delay, and assign the delay return channel to the same subgroup as the vocals. This way, the vocal delay effect will follow the subgroup fader.



Another reason to return the delay to a channel would be so that its output could be sent to one of the reverbs by turning up the AUX 3/5 Send on the delay channel. Send the vocal to the delay (turn up its AUX 4 knob), then add reverb to the delay channel to get a pre-delay on the vocal reverb. Send a keyboard channel directly to the same reverb for a non-delayed reverb effect.

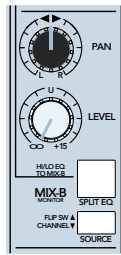


You can create feedback around the delay by turning up its own send (AUX 4) on its return channel. This is sometimes a desirable effect (for repeated decaying delays), but when the gain around the feedback path becomes greater than unity, it'll take off and *HOWL*! Be very careful when feeding a processor signal back on itself.

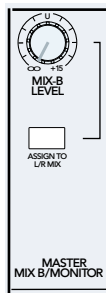
Mo' Inputs

You thought that 32 inputs would be plenty when you bought the console, but then they keyboard player won the lottery and showed up at the gig with three new samplers with eight separate outputs each, which of course you can't trust him to mix. We slipped in some extra inputs that you can use to cover yourself nicely.

Mix-B



Connect those sampler outputs to the TAPE RETURN inputs. Now you have 32 (or as many as your mixer has channels) more inputs to the stereo mix. You can adjust them using the MIX-B PAN and LEVEL controls.



Assign MIX-B to the L/R MIX by pressing the button in the MIX-B MONITOR section just above the meters. The MIX B LEVEL knob above the MIX B ASSIGN button is the Master for this extra mix.

You can even apply EQ to these MIX-B inputs by splitting the channel EQ. See the section on the 8-Bus channel strip for details.

AUX Returns

Unused AUX RETURNS can be used for additional line inputs. All AUX RETURNS can be routed to the MAIN L/R mix, and in addition, AUX 1 and 2 can be assigned to subgroups.

Submaster Inserts

In a pinch, you can use the SUB INSERT RETURNS as extra line inputs. Insert a plug to the first click only, or use a home-made cable with a TRS plug wired with the signal going to the ring only.

Buy Something Mackie

Expand your Mixer with the Mackie 24-channel Expander Console. It adds 24 more full input channels. Better yet, buy several.

Making a Simultaneous Recording

You're all set up, ready to go, when the band's manager comes up to you with a DAT machine and goes "The producer wants us to send him a recording of tonight's gig," and you say, "OK, I'll patch it in to the house mix," and he says, "No, we need a separate recording mix so we can get all the stuff you'll have turned down in the house because this band plays so LOUD." You roll your eyes and go, "You should really get a remote truck," he rolls his eyes and goes, "Here's the tape. Make it sound good and we might hire you next time."

Mix-B To The Rescue

Piece of cake. Use MIX-B! Just plug the stereo recorder inputs into the MIX B OUTPUT jacks. Using the MIX-B SOURCE button, select CHANNEL (*down*) as the MIX-B input.

L/R Mix is still your house mix, the AUX buses are for effects and stage monitor mixes, and MIX-B provides an independent mix for the stereo recording.

Mixing the Recording

Use the MIX B LEVEL and PAN controls for your recording mix. You'll probably want to leave the SPLIT EQ buttons in the normal (*up*) position. This applies the channel EQ to both the main L/R signal and to MIX-B.

If you want to have independent control of the recording and house equalization, the EQ SPLIT button divides the channel EQ into two sections, with the Hi and Low shelving sections going to MIX-B and the middle parametric sections remaining in the main signal path.

Effects for the Recording

If you want to use effects in your recording mix, you'll need to return them through channel Line Inputs rather than AUX RETURNS. The MIX B LEVEL on those channels becomes the effect return for the recording mix. Since MIX B SOURCE is pre-fader (unless you've made the MIX-B Source modification), you'll have independent control of the effect return level in the house, monitor, and recording mixes.

Monitoring the Recording Mix

Select MIX-B as the source for the Phones output that you're using. With the recorder output connected to the EXTERNAL INPUT jacks, press the EXTERNAL Phones select button to check the playback.

Meters for Recording

It's true, there are no meters on the MIX-B buses. The ultimate reference is the meters on the recorder. If you can't see them from where you're sitting, there are a couple of patches to get metering of the record level:

- Since you're not using the CONTROL RM or STUDIO outputs, you can use the monitor section for metering. Simply engage the Mix-B Switch (only) in the MONITOR SOURCE section to send MIX B to the MAIN meters.
- If you have two free submaster buses, patch the MIX-B outputs into the submaster insert return. Push the 1/4" plug only halfway in to the Insert Jack, to the first click. Set the submaster faders to -6 dB to compensate for the insert point gain difference. Now you will have MIX-B levels showing on those two submaster meters, and you can use the SUBMASTER OUTPUTS to feed the recorder.
- If you have the optional meter bridge fitted and you have two extra inputs, patch the output of MIX-B into the open channels (either Line In or Tape In, depending on how you have set the global source switch on the meter bridge), set the levels at unity and watch those two channel meters for recording levels. You can now feed the recorder from the two channels' DIRECT OUTPUTS. Make sure all the channel bus and AUX assign switches and controls are off, so you don't accidentally assign your recording mix back into the house or monitor mix. In addition, with this connection, you could add a little EQ or compression via the channel inserts.
- In either of the above cases, you may want to keep the extra circuitry out of your recording signal path. If so, just mult the MIX-B outputs to both your recorder and to the patch point for metering. Use a pair of Y-cables.

Making a Simultaneous Multitrack Recording

You can make this simple or you can make it too complicated. The simple way is to patch each channel INSERT Output to its own track. That way, mixing the multitrack recording will be just like mixing the show. Insert a TS plug halfway into the INSERT jack, to the first click to get a clean direct output from each channel you're sending to the recorder..

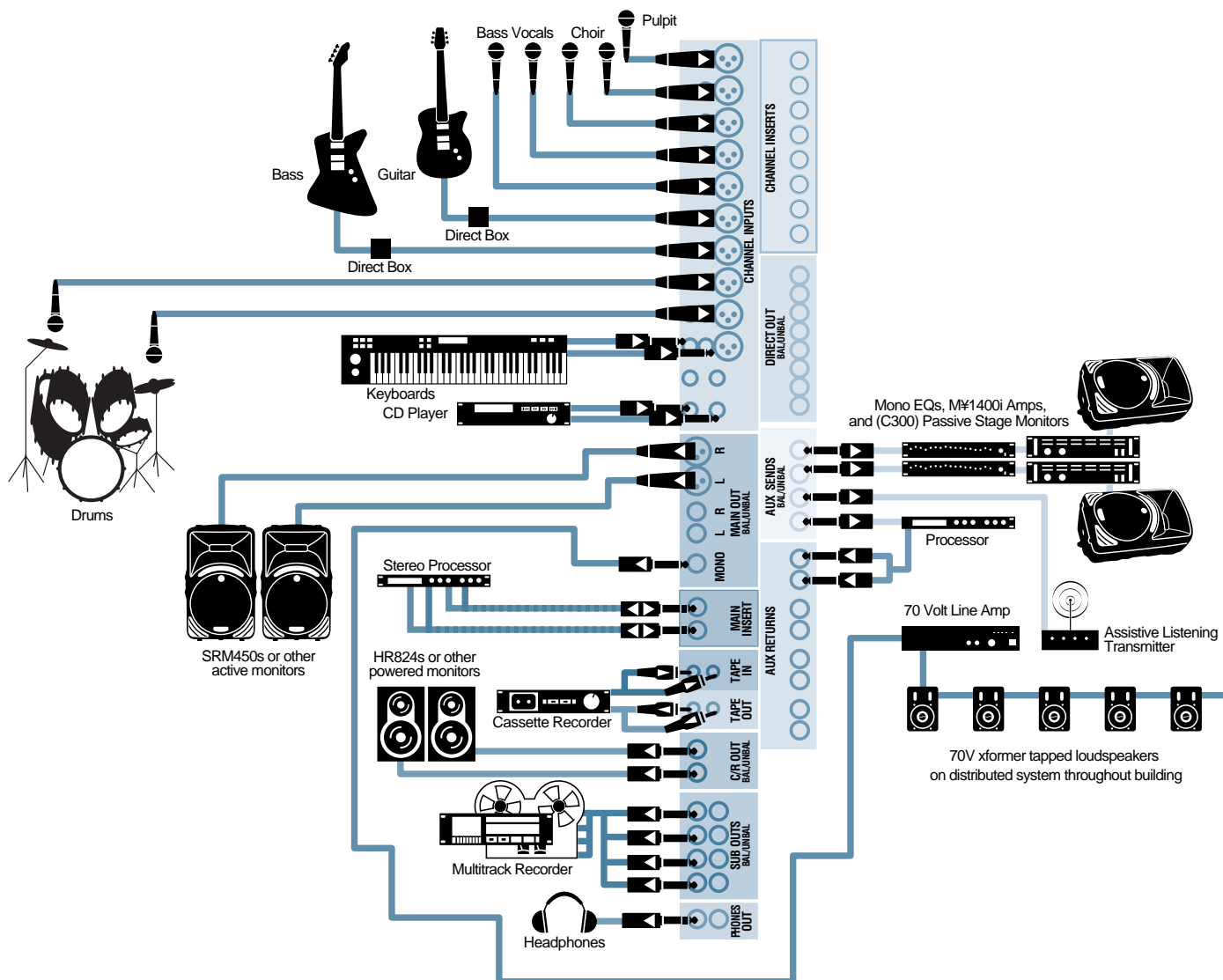


Since the DIRECT outputs come after the EQ and fader, it's really preferable to take the recording signal from further upstream. Chances are when you mix in the control room, you'll want different EQ settings than what you were using for the house mix, and it's difficult to un-do equalization once it's applied.



If you're already using the INSERTs for signal processors, you can get creative using Y-cables to split the signal at the INSERT Output. You could in theory pass the signal through the recorder, returning it to the channel from the recorder's output, but this is risky. If for some reason the recorder goes bonkers mid-show, you could lose that channel in the house mix until you discover the problem and re-patch. In the meantime, the audience, who doesn't care a hoot about the recording, will be staring at you, and that's not a good feeling.

I642VLZ-PRO House of Worship Installation



This setup illustrates a typical installation for a contemporary house of worship. It includes a setup for a folk/pop-style band, a pair of mics for the choir, a pulpit mic, and a CD player for playback of incidental music. In addition to the standard main and monitor speakers, we've incorporated a couple of outputs that are typically found in institutional installations, an *assisted listening* transmitter and speakers placed throughout the building fed from a 70 V distribution system.

Inputs

All ten mic inputs are used in this setup. MIC INPUTs 1, 2, and 3, the pulpit and choir mics, will probably be part of a permanent installation, mounted and hung in appropriate spots with their cables remain

ing in place. Since the band configuration may change from week to week, those inputs will be set up as required. We've shown a typical setup for a church youth band, including direct boxes (DI's) for the acoustic guitar and bass, a couple of mics for the band's singers, and a pair of mics on the drum kit.



Note that mono/stereo channel 9-10 is used for one of the drum mics, so it becomes a mono channel.

The keyboard is connected to the stereo line inputs of channel 11-12. We've left stereo channel line inputs 13-14 open to your imagination. A wireless mic with line level output might fit in here. Finally, stereo line inputs 15-16 receive the CD player.

Mains and Monitors

We've shown powered main speakers connected to the MAIN L/R OUTPUTs. Of course passive speakers and power amplifiers could be used in their place.

Monitors, through graphic equalizers and power amplifiers, are fed from AUX 1 and AUX 2 OUTPUTs. See the 1604VLZ-PRO Stereo PA setup for further details on the controls for mixing mains and monitors.

70 V Distribution System

When it's necessary to distribute speaker-level audio to multiple destinations located at fairly substantial distances from the power amplifier, we often use what's known as a constant voltage distribution system.



A conventional power amplifier is designed to work most efficiently into a low impedance (2 to 8 ohm) loudspeaker. At that impedance, a substantial amount of current flows in the speaker wires, something on the order of 10 to 15 amperes—about what an ac circuit in your house carries. That's why we recommend using at least 14 gage wire for connecting speakers.

Wiring a building with hundreds of feet of 12-14 gage wire for speakers isn't very practical, so we use a distribution system that's similar in concept to what the power companies use to get large amounts of power from the substation to your house—they carry that power at several thousand volts, but at a lower current, which reduces the required wire size for the same amount of power. A transformer on the utility pole outside your house drops the high voltage down to standard line voltage.

In a constant voltage speaker system, we run the amplifier so that it's putting out 70 volts at its rated power, and tap off it wherever we need a speaker—in the lobby, in the social hall, in the offices. A transformer installed at each speaker delivers the required amount of power to that speaker. You might need 25 watts in the lobby, 50 watts in the social hall, but only 10 watts in the offices. With a constant voltage distribution system, adding another speaker to the string creates negligible voltage drop in the wiring, so you only need to be concerned that the total speaker load (in watts)

not exceed the power rating of the amplifier.

Since distributed speakers are almost always mono, we've chosen the MONO output of the mixer to drive the 70 V amplifier. It has its own volume control which will aid in matching the mixer to the amplifier.

Remember that all speakers are fed from the same amplifier, so they all follow this one volume control. When they tell you that it's fine in the lobby but too loud in the office, change to a lower wattage transformer in the office. Typically the transformers on the speaker end of a constant voltage distribution system have taps for several different power levels, so select the tap that best matches the desired volume for that location.

Assisted Listening System



Institutions often provide a wireless transmission system that makes it easier for those with hearing disabilities to hear the program. There are two common types. One is a radio system that's like a limited range broadcast transmitter—those needing assistance are given radio receivers with earphones so that they can adjust the program to a comfortable listening level in their phones. Another type is what's known as an "induction loop". A power amplifier feeds a one turn loop of wire which encircles the entire room and is usually installed around the baseboard. This serves as one winding of a transformer, the other winding being a pickup coil that's standard part of a hearing aid designed to work with such a system.



We're using AUX 3 OUTPUT to feed the assisted listening transmitter. Since there's no master level control for AUX 3, it's necessary to adjust the level at the assisted listening transmitter. Like any other sound system component, it can be overdriven to clipping distortion (which is *really* painful when injected directly into an ear!).

Since those listening through this system will want to hear the full mix, a good way to establish the proper level into the transmitter is to set all the AUX 3 controls to their Unity Gain position and just nail them in place. Then, with a full mix cranking, set the input gain of the assisted listening transmitter for a loud but undistorted volume at a receiver. Since AUX 3 is a post-fader send,

with all the send controls set at the unity gain position, you'll have the same mix in the Assisted Listening system as you have in the house. If the drummer is a serious pounder or the bass player has a giant amplifier, you might want to save what hearing those listeners have left by tuning the AUX 3 controls on those channels down a notch.

Recording

Worship houses often like to record their services, so we've provided a cassette recorder for that purpose. Connected to the TAPE OUT jacks, it will record whatever is happening. The recorder's outputs are connected to the TAPE IN jacks, so you can check the recording in the phones. Select TAPE as the CR SOURCE, and bring up the TAPE IN level control.



Feedback Alert! Make sure you don't accidentally assign the TAPE to MAIN MIX while you're recording.

If you want to listen to the tape playback through the PA speakers, press the TAPE TO MAIN MIX button. You probably *don't* want to do that while the service is in progress, however.

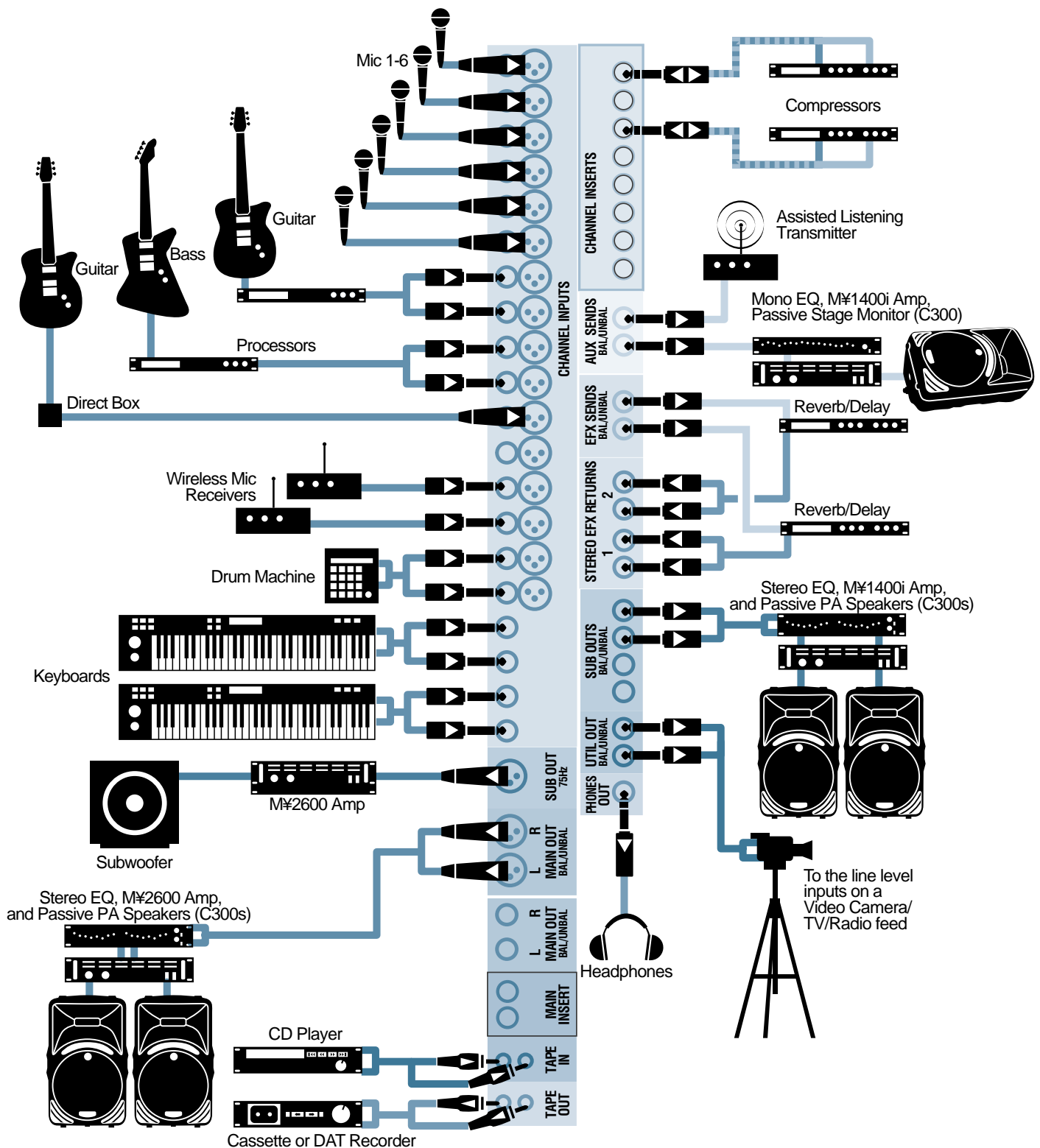
Adjusting the Record Level

We know that a lot of church sound folks don't run the system at rock'n'roll levels. The PA volume is fine, but the mixer output level is low, and they can't turn the gain of the recorder up high enough to get a good record level. The solution is to turn the gain of the PA amplifiers down so that the mixer can be run at a higher level. The PA volume will remain the same, but the recorder will now be seeing a higher level out of the mixer. See Gain Structure in the Tips section for a further discussion.

Multitrack Recording

We've shown a multitrack recorder connected to the SUB OUTPUTs for the purpose of recording the band or choir. Refer to the multitrack recording setups for suggestions on routing and patching to best fill the number of tracks you have available.

Chapter 22: PA Setups: CFX House of Worship Installation



In this setup, we've taken a little different approach to feeding the multiple outputs typically required for a house of worship installation.

Inputs

These are conventional, whatever's required for the setup. The TAPE IN jacks are used to bring a CD player into the mixer. For filler music, the BREAK switch sends the CD player through the TAPE LEVEL con-

trol to the MAIN and UTIL outputs and mutes all the inputs. If you want to send the CD output to all of the remote speakers, connect it to a channel line input or AUX RETURN rather than the TAPE input.

Outputs

The main PA power amplifier and speakers are fed from the MAIN outputs. A sub-woofer is fed from the 75 Hz SUB OUT. The built in Graphic Equalizer operates on the MAIN outputs, so this can be used to tailor the frequency response of the system to the room.

The UTIL OUTPUTs are used to feed the chapel speakers. This output duplicates the MAIN mix but has it's own level control. The UTILITY OUT LEVEL control is fed from the MAIN MIX fader and graphic equalizer, so the level in the chapel will follow changes to the MAIN mix.

SUBGROUP OUTPUTs 1 and 2 are used to feed the nursery speakers, allowing the volume there to be controlled by the SUB 1-2 faders. Those are ahead of the MAIN fader so the level in the nursery will remain constant when changes are made to the MAIN mix.

ASSIGN Tips

The CFX doesn't have direct assignments to the MAIN L-R bus. Everything must go through the SUBs in order to get to the MAIN outputs. Assign all the channels to both 1-2 and 3-4. Assign SUB FADER 3 to LEFT and SUB FADER 4 to RIGHT and set those faders to the Unity gain position. That will route all channels to the MAIN outputs in stereo.

Although SUB Outputs 1 and 2 are unassigned in the Master section, they'll still receive the mix from all the channel faders, but they won't contribute to the MAIN mix. They can therefore be used as an independent level control for the nursery. You also have the option of *not* assigning certain channels to the nursery feed. By assigning the pulpit and wireless mics to 3-4 only, the nursery speakers will receive the music but not the talk.

AUX SENDS

AUX SEND 1 is used to feed the stage monitor. If a second monitor mix is required, it could be connected to AUX SEND 2. Set AUX 1-2 to PRE FADER for a monitor mix that's independent of the main mix.

EFX SEND 1 is used to feed the Assisted Listening system. This is a post-fader send which follows the mix set on the channel faders.

Effects

We like the built-in EMAC processor so we're relying on it for effect processing in this application. An outboard effect could be used to replace the EMAC by connecting its input to EFX SEND 2 and its outputs to EFX RETURNS 2.

Since the EFX SEND 2 jack is normalled through like a channel INSERT jack, it's possible to send the same EFX 2 mix to both the EMAC processor and an outboard processor by inserting the plug for the cable feeding the external processor halfway into the jack (to the first click). This will tap the EFX 2 signal off to the processor without interrupting the signal to the EMAC. With the outboard processor outputs connected to RETURNS 1, either the EMAC or the outboard effect could be used in the mix. Turn up the EFX 1 RETURN knob when you want to use the outboard processor effect. Turn up the TO MAIN MIX knob in the EMAC section to bring the EMAC effect into the mix.

If an outboard effect is required on one source only, it can be inserted in-line (serial) using the channel INSERT jacks, or by sending a direct output from the channel INSERT jack to the effect input by inserting the plug to the first click, and then returning the effect output to the mix either through RETURN 1 or the line input of an open channel.

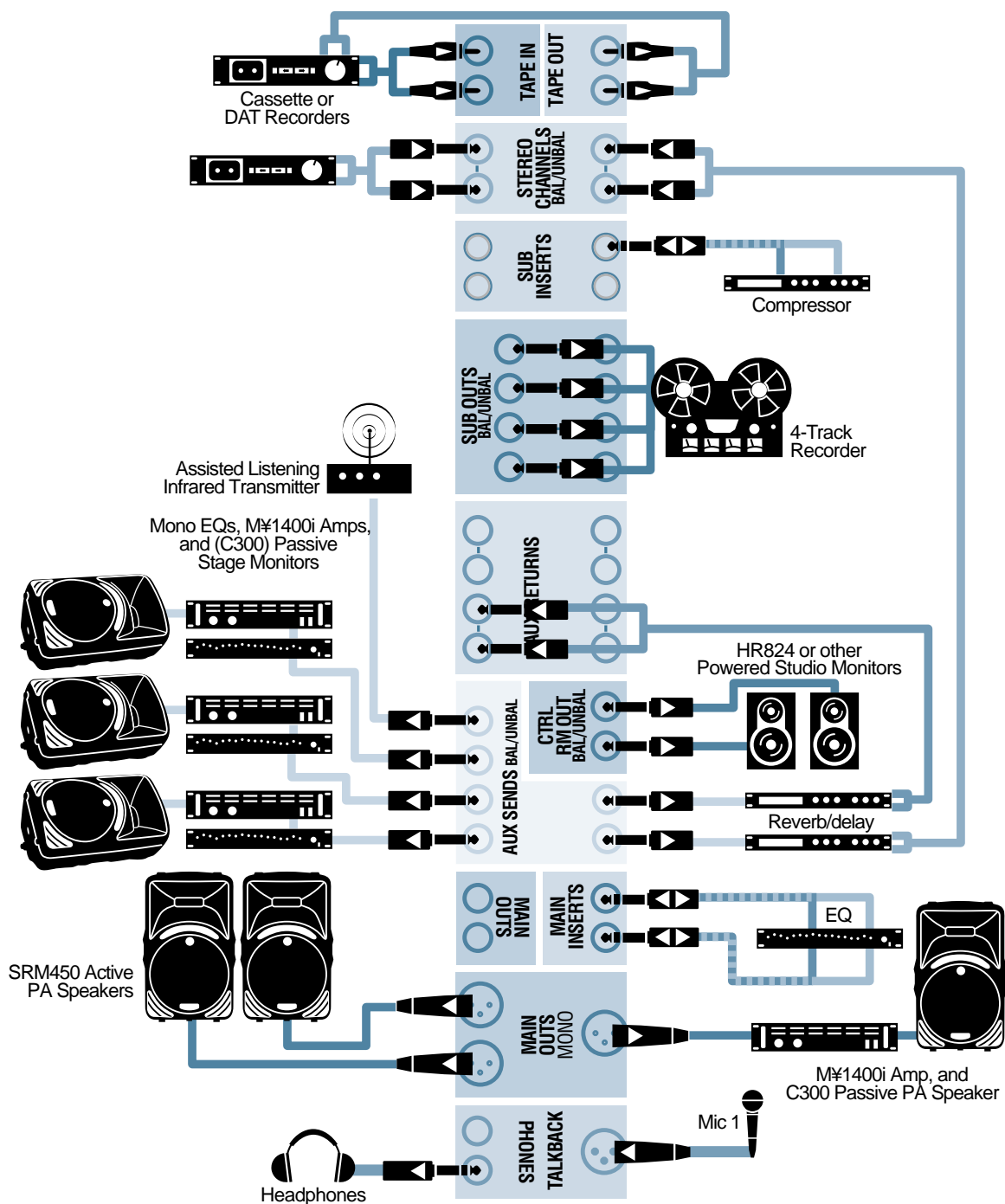
Recording

We've shown a cassette or DAT recorder connected to the TAPE OUT jacks for the purpose of recording the main mix.

If the PA level is fine, but a satisfactory record level can't be obtained, this probably means that the gain of the power amplifiers is set too high, requiring the mixer gain

to be run too low. Reduce the power amplifier gain and raise the mixer level. When the mixer meters are reaching 0 VU on peaks, you should have ample level on a recorder that's designed to work with the -10 dBV (consumer) operating level. A recorder designed for the professional +4 dBu operating level can be connected to the 1/4" MAIN OUT jacks. Note, though, that those jacks follow the Graphic Equalizer, so any equalization applied to the main front-of-house mix will also be applied to the recording.

SR House of Worship Installation



Here's one more suggestion for a house of worship setup built around the SR series mixer. Check the 1642 and CFX church setups for operating tips.

Inputs

Microphone inputs are as required. A playback tape deck or CD player is connected to stereo channel 21-22.

Outputs

As in the other institutional setups, the MAIN L-R outputs are used to feed the main speakers. In this setup, all of the remote speakers are fed a mono mix—the chapel from the MONO output and the 70 V distribution system AUX SEND 3. An assisted listening transmitter is fed from AUX SEND 4.

The four SUB outputs are used to feed a 4-track recorder, while TAPE outputs feed a stereo recorder. If you can't get a satisfactory record level from those outputs, check

the gain setting of amplifiers in the powered speakers, and reduce the gain if necessary in order to allow the mixer to run at a higher level.

Effects

Two effect processors are shown, with their inputs fed from AUX SENDS 5 and 6. The outputs of one processor are connected to stereo channel 23-24, while the other is returned to the mix through AUX RETURN 1.

Assigns

Assign all of the active input channels to L-R to send them to the main mix. Assign channels to SUBs 1-2 or 3-4 to send channel inputs to the 4-track recorder. If an 8-track (or more) recorder is used, INSERT outputs (insert a plug to the first click only) can be used to feed a track directly from a channel.

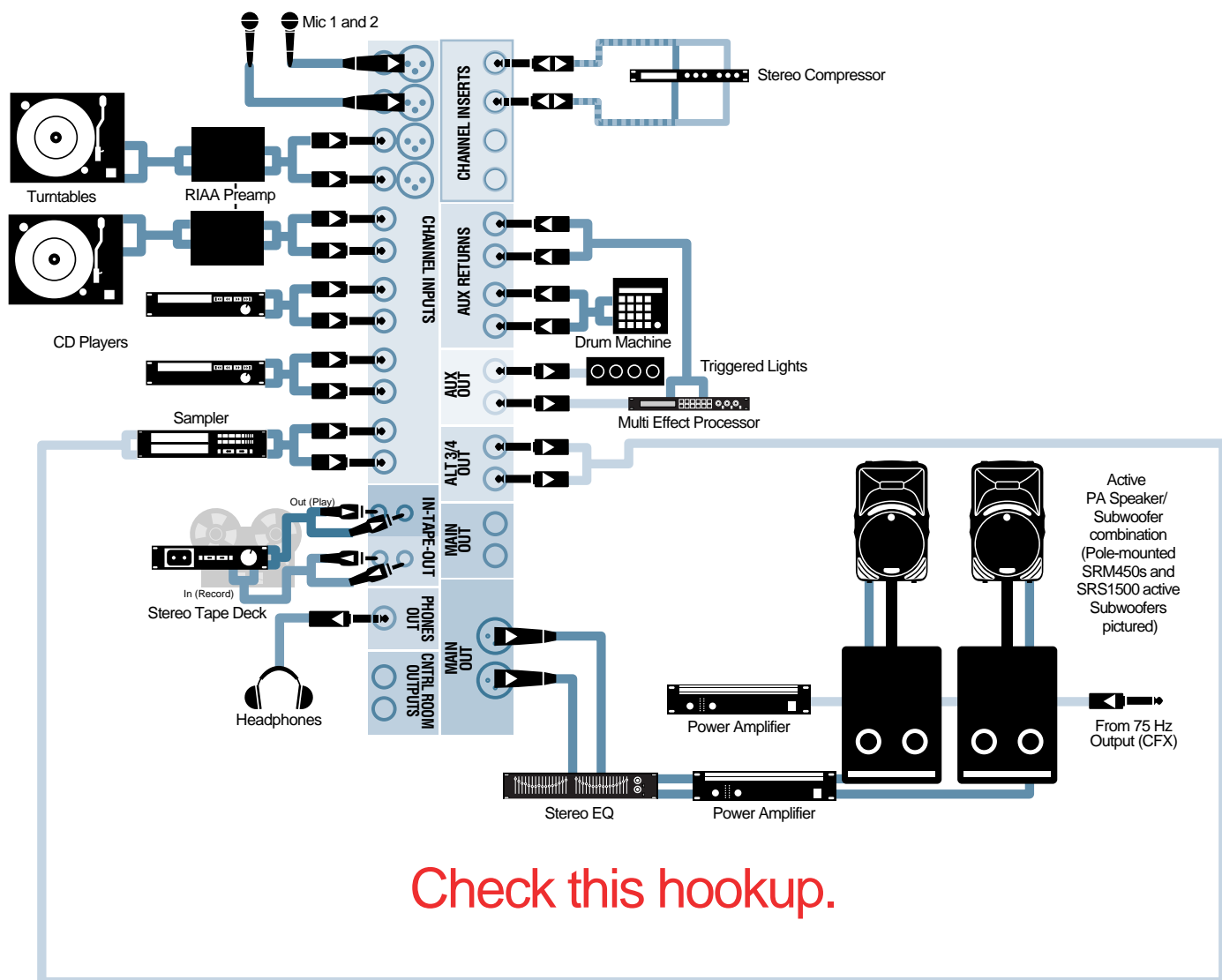


Channels which feed recorder tracks directly should *not* also be assigned to subgroups.

Mixdown

The same setup can be used to mix the 4-track recording. Connect the recorder outputs to mono line level inputs and mix while monitoring on the Control Room speakers.

Chapter 23: PA Setups: DJ Setup



A DJ system is a special PA application, so you'll find this to be similar to other PA hookups. We've shown a 1402VLZ-PRO in this setup, but it's applicable to the 1202, 1604, 1642 and CFX series also

Connections

The basic goal of a DJ setup is to create a stereo mix of several inputs and send it to a set of PA speakers—BIG and LOUD speakers. The usual sources are turntables, but CD players, drum machines, and samplers are also common in DJ systems. The music sources are connected to mono or stereo inputs which are assigned to the MAIN L-R outputs.

Turntables

An RIAA-equalized preamp is necessary to interface a turntable to the mixer's line level

input. Many commercial DJ turntables have suitable preamps built in, but audiophile turntables don't. Those without built-in preamps expect to be connected to the Phono input of a receiver which is equipped with the proper equalizer.



The pickup cartridge in a turntable puts out a signal that's at about the same level as a microphone, however you can't successfully connect a phono cartridge directly to a mic input for two reasons. The low impedance of a mic input is a poor match for the fairly high output impedance of a phono cartridge. Also, RIAA equalization is required to compensate for high frequency boost applied during the record cutting process. An unequalized phono cartridge output sounds awful. Believe it!

Sampler

Phrase samplers are becoming fairly common DJ tools these days. The sampler outputs are connected to a pair of stereo line level inputs as you'd expect, but a sampler is a recording device, too. The recording inputs to the sampler are shown connected to the ALT-3/4 OUTPUTs on the 1402. On a true 4-Bus mixer, connect them to a pair of SUB outputs (SUB 1-2) instead. This will allow you to sample a phrase that you're playing live from a turntable or from the microphone, then manipulate it and play it back. You can sample from the MAIN outputs, too.



By connecting the sampler inputs to a separate bus, you can sample "off line". By pressing the MUTE/ALT button on the 1202/1402 mixer or changing a CD or turntable's output ASSIGN from L-R to 1-2, you can play that source into the sampler without it being heard by the audience. Pressing the ALT 3-4 button in the CR/SOURCE group will allow you to listen to the source you're sending to the sampler—for instance to cue or scratch a record.

AUX Returns (Drum Machine)

This is a pretty full-up system and we ran out of stereo line level inputs, so in this illustration, we're using stereo AUX RETURN 2 inputs for the drum machine. Those inputs go straight into the MAIN mix through the AUX 2 RETURN level control, so they can be used for additional line level inputs. The limitation is that the AUX inputs on the smaller mixers cannot be re-assigned to the subgroup busses, so with this hookup, you can't sample the drum machine.

AUX Sends

We've shown one conventional and one unconventional application. The conventional one is an effect processor that receives its input from SEND 2 and its output is connected to RETURN 1.

The unconventional application is the use of the SEND 1 output to trigger a light sequencer. This allow you to choose (with the AUX SEND 1 controls) which channel's signal, rather than the whole mix, triggers the light sequence. You probably don't want the lights to trigger from your voice, so keep those AUX 1 knobs down. If you

want the light show in sync with one CD player but not the other, turn one CD channel's AUX 1 knob up and the other down.

On the 1202 and 1402, RETURN 2 can be sent to AUX SEND 1 by engaging the EFX TO MONITOR switch. That's the button that will make the lights flash in step with the drum machine that's connected to RETURN 2.

If you have an effect processor such as a triggered filter or pitch shifter, where you want to hear *only* the effect, not the "dry" source, here are two suggestions:

One approach is to connect the processor's input to AUX SEND 1 and bring its output back into the mixer through either an AUX RETURN or channel LINE INPUT. By switching AUX 1 to PRE, with the source channel's fader all the way down, turning up AUX 1 will send the channel's signal to the processor without sending it to the main outputs. With the processor set for fully wet output, you'll hear only the processed sound in the mix.

Another approach is to use the Channel Insert. With the processor set for full wet (processed) output, you'll hear only the processed version of the channel.

Channel Inserts

These are available for outboard signal processors such as compressors, using "Insert" cables. As a reminder, you can also put individual effects in line (serial) with a channel using the INSERT jacks. You may wish to dedicate a multi-effect box to your voice in this way rather than use the processor in the SEND/RETURN (parallel) loop.

Recorder

Recording isn't normally a part of a DJ operation, but we've shown one connected to the TAPE OUT and IN jacks. Whatever goes to the MAIN speakers goes to the recorder. When it's karaoke night, this will come in handy for those who want to take home a souvenir of their art.

Subwoofer (CFX Series Only)

The CFX series provides a low-pass filtered output which is a mono mix of everything that's below 75 Hz in the MAIN outputs. Connect this to a power amplifier and subwoofer for an exciting low end boost.

Operation

Normally all the sources will be assigned to the L-R MAIN outputs, so turning up the channel faders with the MASTER set to its Unity gain position will get your sources going to the speakers. No secrets here.

Level Setting



Adjust the TRIM controls on those channels that have them by using the **Level-Setting Procedure**. Since the level on vinyl records, particularly older ones, varies widely and you don't want to have to keep fooling with this control all night, keep the TRIM setting conservative. Instead of setting it so that the meters peak at 0 VU, back it off a bit so that they peak at -4. That will keep louder records from clipping.



Successful DJ-ing is all about getting things to sound a little different than the next guy. While throughout this book we keep reminding you to set the TRIM properly to avoid clipping and achieve the best signal-to-noise ratio, you might just find that sometimes your voice really sounds cool when the mic preamp is clipping. This is OK as an effect, but don't make it a habit, and don't fry your tweeters by sending them a loud, clipped signal.

Cueing Up

To cue a record by listening to it in the headphones without sending it to the MAIN mix, MUTE the channel or pull its fader all the way down, and press the channel's SOLO button. On the 1604 and 1642 mixers, be sure you have the SOLO MODE PFL/AFL switch in the PFL position.

Chapter 24: PA Setups: PPM Series PA Applications

These powered mixers make PA setup simplicity in itself. There's little more to do than plug in mics, speakers, ac power, and go. Lest you ignore some general precautions which apply to all PA setups, we'll review them here. Most of this is common sense, but it's worth mentioning since this might be your first experience with professional sound reinforcement equipment.

Thermal Considerations

The PPM Series Powered Mixers contain two powerful built-in amplifiers. All amplifiers produce heat, and the more power the amplifier produces, the more heat is generated. That heat has to go somewhere, and it's important to get it away from an amplifier as quickly as possible. Providing adequate ventilation around the amplifier will result in increased reliability and longevity for your PPM.

Heat from the output transistors is coupled to the massive die-cast aluminum heat sink on the rear panel. This draws the heat away from the transistors and out to the radiating fins. Air flows across the fins by a process called convection, drawing the heat away from the fins—just like the radiator in your house.



In order for convection cooling to work effectively, it's important to provide adequate air space behind the mixer. When you set up the mixer, we recommend leaving at least six inches between the back of the mixer and a wall or curtain to allow adequate air circulation.

Thermal Shutdown

In the unlikely event that the amplifier overheats, a thermal switch will activate, shutting down the mixer until it cools off. If your mixer shuts down, double-check the speaker connections to be sure you haven't loaded either of the amplifiers with a lower impedance than 2 ohms. Overloading the power amplifiers can cause overheating, even at less than maximum volume.

If the load impedance is okay and the amplifier has plenty of breathing space, overheating may be caused by excessive ambient temperature (Is it sitting in the sun?). Try aiming a fan at the heat sink to move air across the fins faster.

AC Power Considerations



Be sure your PPM Series Powered Mixer is plugged into an outlet that is able to supply the voltage specified for your model. While the mixer will operate from 75% below to 110% above its rated line voltage, if the line voltage should drop below 97% of the specified voltage, the amplifiers will no longer be able to supply rated power, resulting in a loss of headroom.



When setting up for a gig, don't plug the mixer into the same outlet or the same electrical circuit as the beer cooler, air conditioner, or other plant equipment that draws heavy current intermittently. It's not unusual for a refrigeration compressor, particularly in a building with old wiring, to drag the line voltage down by as much as 20% for several seconds whenever it starts.

Under typical conditions of reproducing music with the musical peaks just below the amplifier clipping level, the powered mixers draw the following average currents:

Average Current (120 Vac Line)		
Speaker Load (per amplifier)	406M, 408M, 408S	808M, 808S
8 Ohms	2.0 A	2.2 A
4 Ohms	3.1 A	3.3 A
2 Ohms	4.8 A	5.0 A

NEVER bypass the ac plug's ground pin. This is dangerous! Find a properly grounded ac outlet.



It is recommended that the PPM be connected to a stiff (robust) supply of ac power because the amplifiers place momentary high current demands on the ac line. It's a good idea to carry a heavy duty extension cord (12-3 wire) along with your mixer. That will allow you to find a "clean" outlet rather than just the closest one, and the heavy extension cord will have minimal voltage drop.

The more ac voltage that is available to the mixer (within the specified limits, of course), the louder the amplifiers will be

able to play and the more peak output power will be available for cleaner, punchier bass. The complaint of “poor bass performance” is often a result of a skimpy ac supply to the amplifiers.

AC Power Distribution

The majority of ac outlets encountered in homes and clubs (in the U.S.) are served by a 240 Vac center-tapped service entrance transformer. This provides two 120 V legs, or phases of ac power on each side of a neutral conductor.

If lighting is used in a show, it is preferable to power the lights from one leg of the service, and power the audio equipment from the other leg. This will help minimize noise from the lights coupling into the audio, particularly when SCR light-dimmers or switches are used.

When setting up for a show, oftentimes you are plugging into a wiring system you know nothing about. You may even be faced with 2-wire outlets that are missing the third safety ground pin. It's a good idea to have a three-wire ac outlet tester in your toolbox so you can check the outlets yourself to make sure they are wired correctly. These testers will tell you if the polarity of the hot and neutral wires is reversed and if the safety ground is disconnected. Don't use an outlet if it is wired improperly! This is to protect yourself as well as your equipment.



If you must plug into a two-wire outlet, use a two-wire to three-wire adapter (and this is the *only* proper use for such an adapter!). These adapters have a metal tab or a wire with a lug attached, which you should secure underneath the screw that holds the ac outlet faceplate in place. **NO CHEATING!!** This center screw is grounded (it's the law—the electrical code) and will provide a safety ground for your mixer. Verify proper grounding by connecting the adapter to the outlet and then plugging in your handy-dandy ac outlet tester.

Electromagnetic Interference (EMI and RFI)

The PPM mixers have been designed to reject electromagnetic interference floating through the air. You won't hear taxi cab radios or TV broadcast coming from your

PA speakers, but sometimes EMI can come from some unexpected sources and cause annoying buzzes in your system.

Neon signs are frequent offenders. If the club stage is right behind the front window, and there's a bright red neon “BEER” sign flashing on and off, find another place for the mixer other than on a table right behind the sign.

Another common source of EMI is a light dimmer. If you're using lights with dimmer controls, keep your lighting cables separated from your mic and speaker cables, and try to plug the lights into a different circuit than the amplifier. Some cheap wall mounted dimmers found in homes are pretty noisy when the lights are anywhere between full on and full off. While you aren't likely to encounter one of those on a gig, you might have one in your basement rehearsal space.

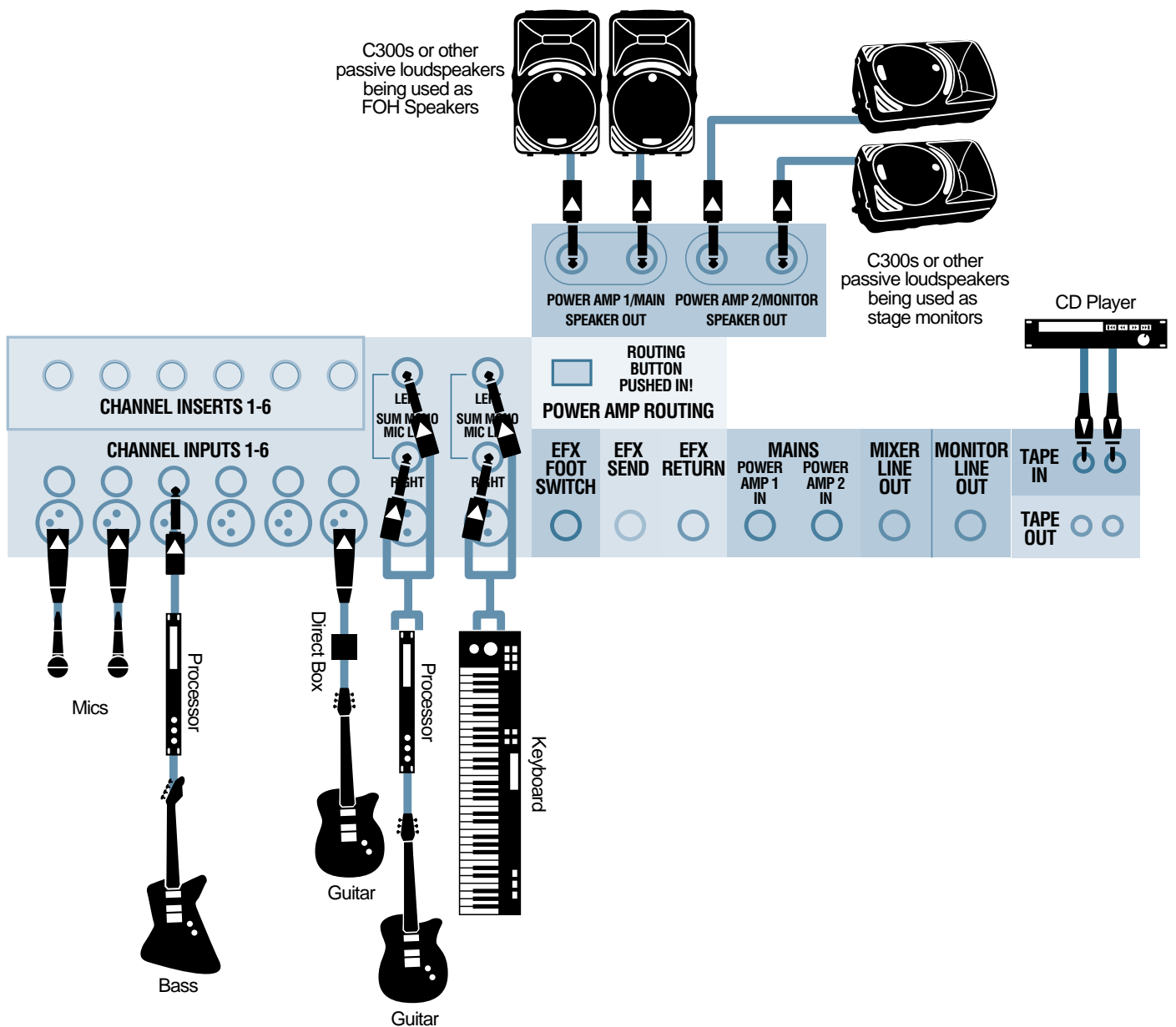
Your Roadie Toolkit

So you won't be caught short-handed at a gig, be sure to pack:

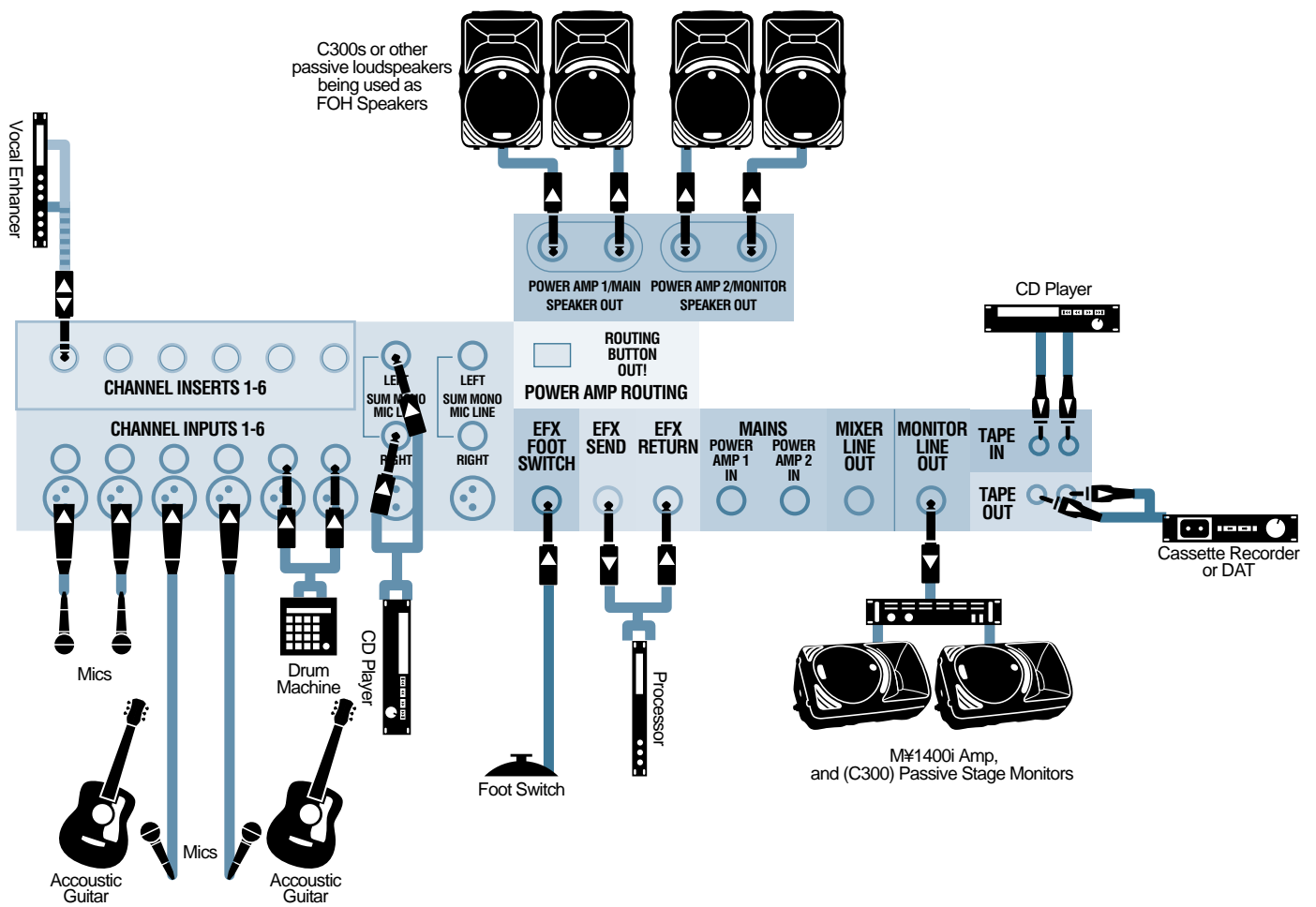
- A two-to-three wire ac adapter
- A common and a Phillips screwdriver (to ground your 2-wire adapter)
- A real heavy big orange extension cord, 14 gage minimum, 12 gage preferred.
- A multiple outlet strip—you're bound to have more than just the mixer to plug in.
- A volt-ohm meter (VOM) for checking ac line voltage and finding shorted or open cables.

You'll find several applications on the following pages which should cover most of your working situations, or at least give you a good start.

408M Small Club Mono PA



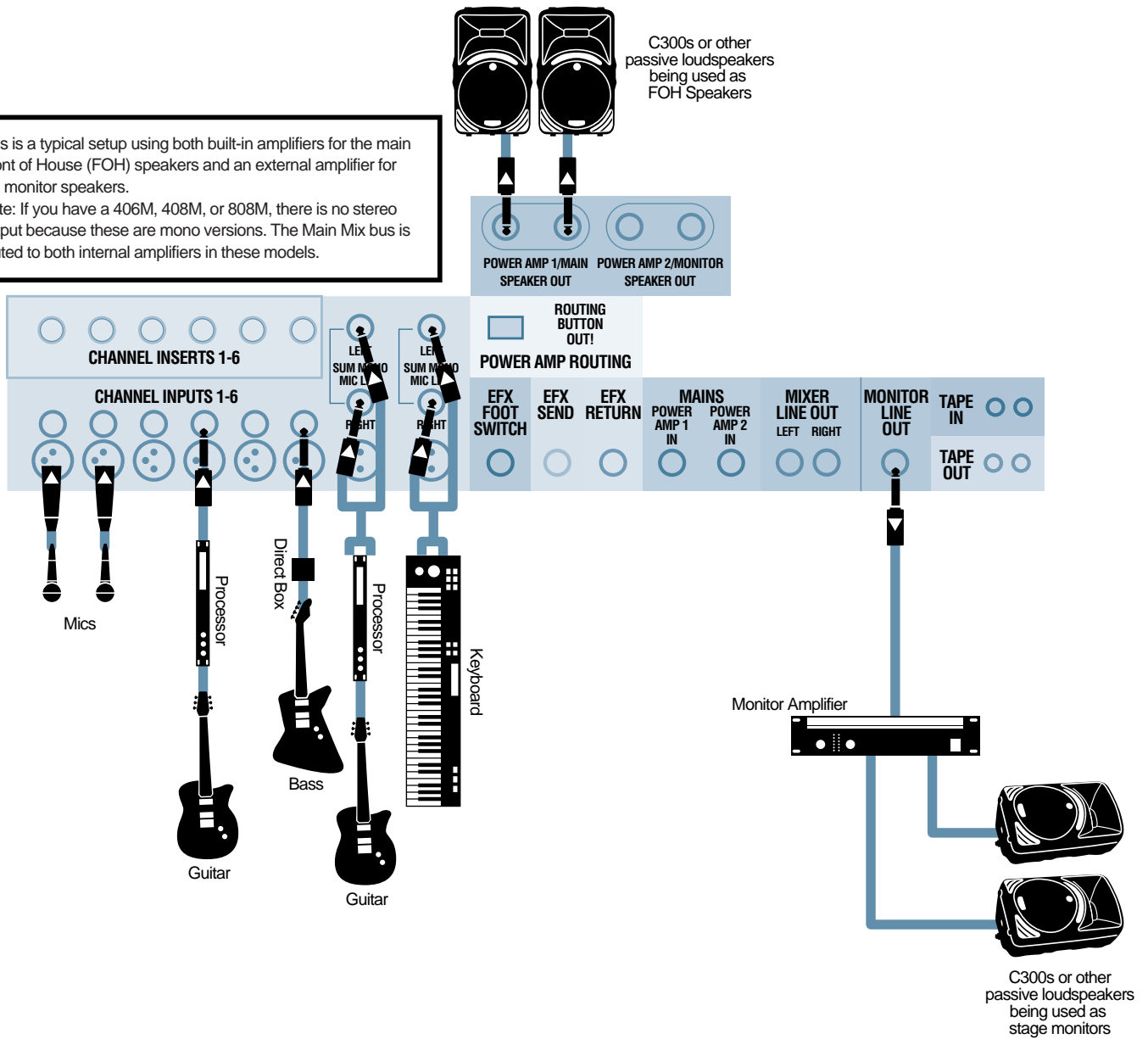
808M Large Club or Auditorium Mono PA



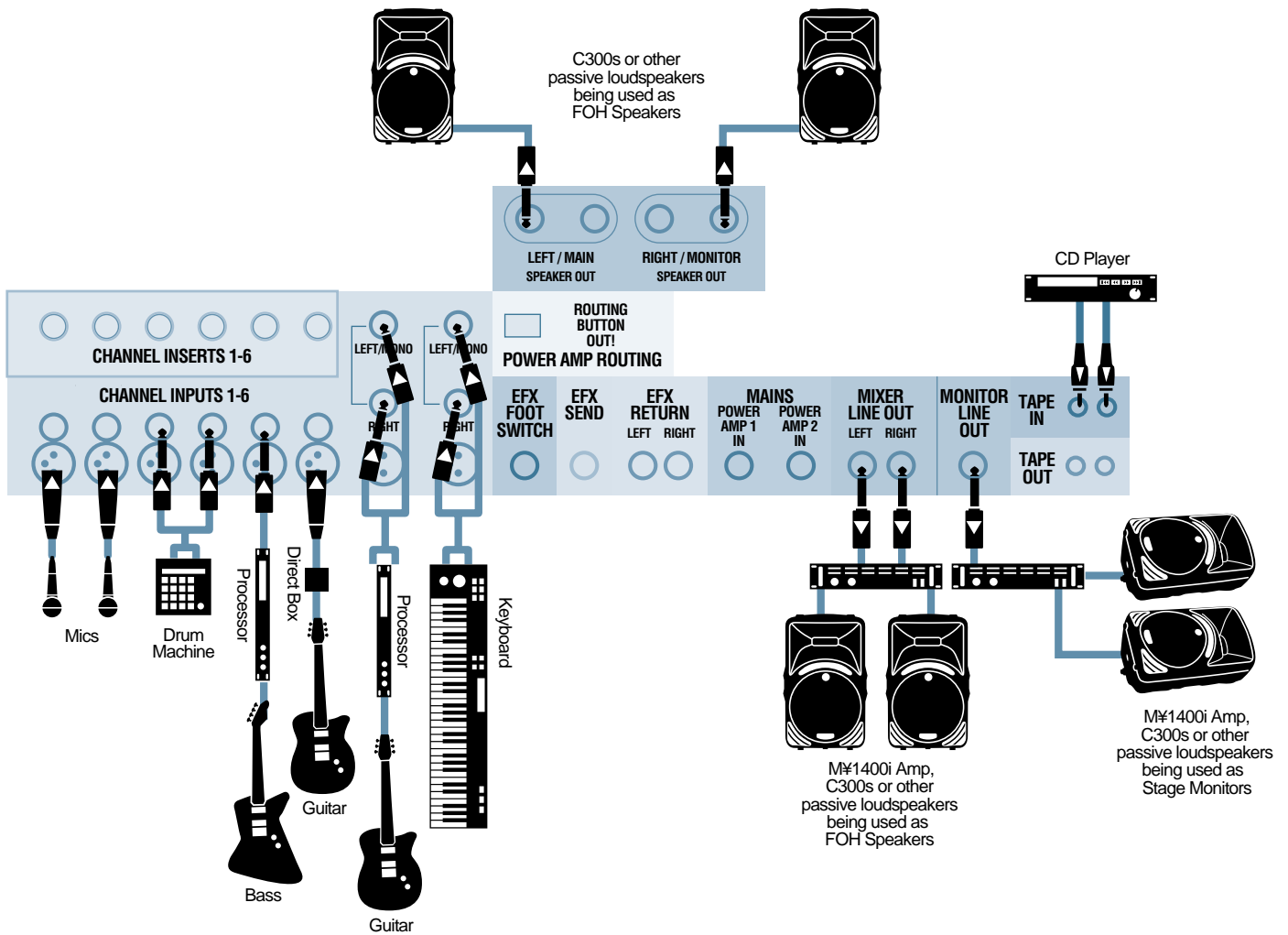
408S Small Club Stereo PA

This is a typical setup using both built-in amplifiers for the main Front of House (FOH) speakers and an external amplifier for the monitor speakers.

Note: If you have a 406M, 408M, or 808M, there is no stereo output because these are mono versions. The Main Mix bus is routed to both internal amplifiers in these models.

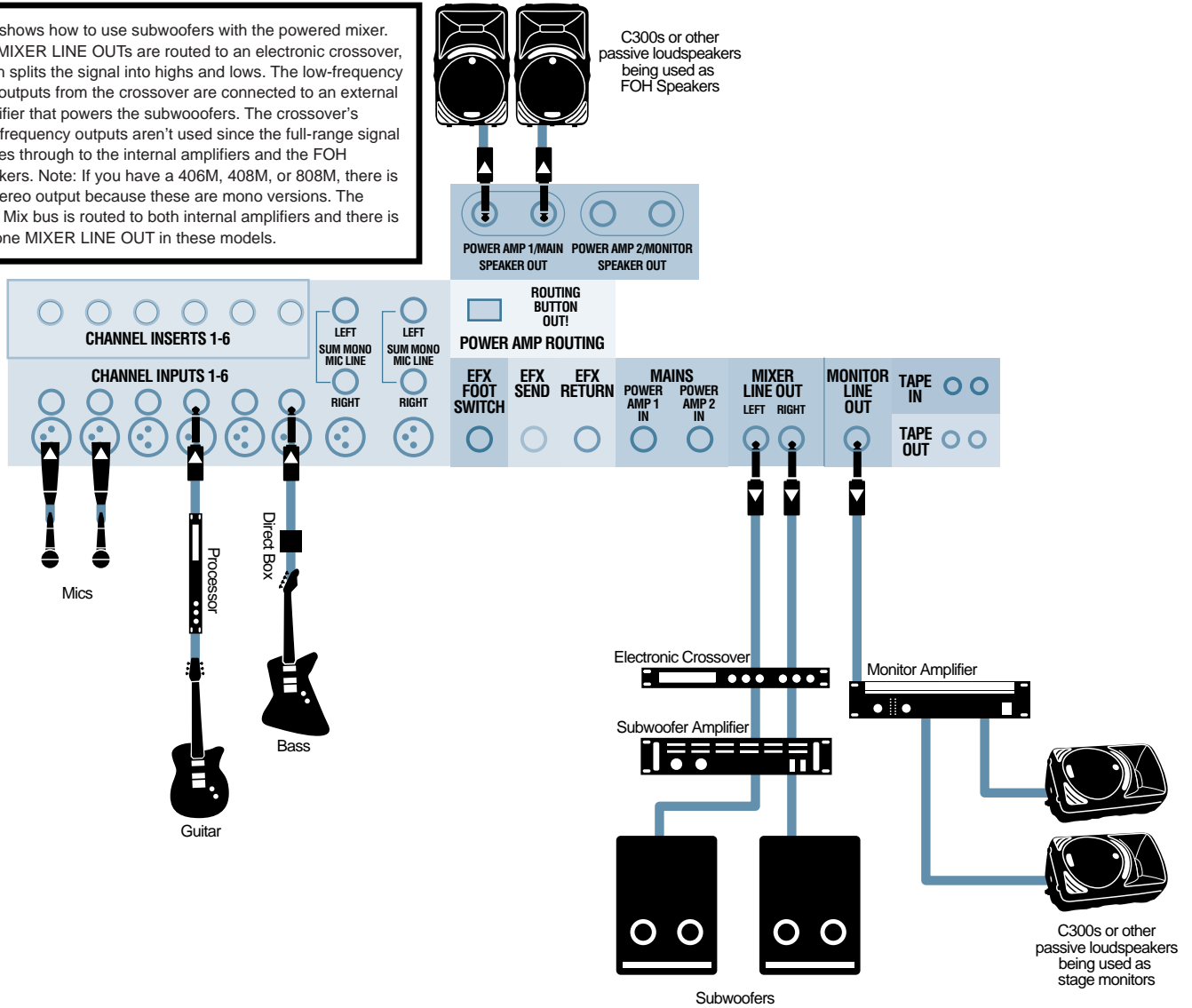


808S Large Club Stereo PA with Auxiliary Amplifier to Reinforce Front of House



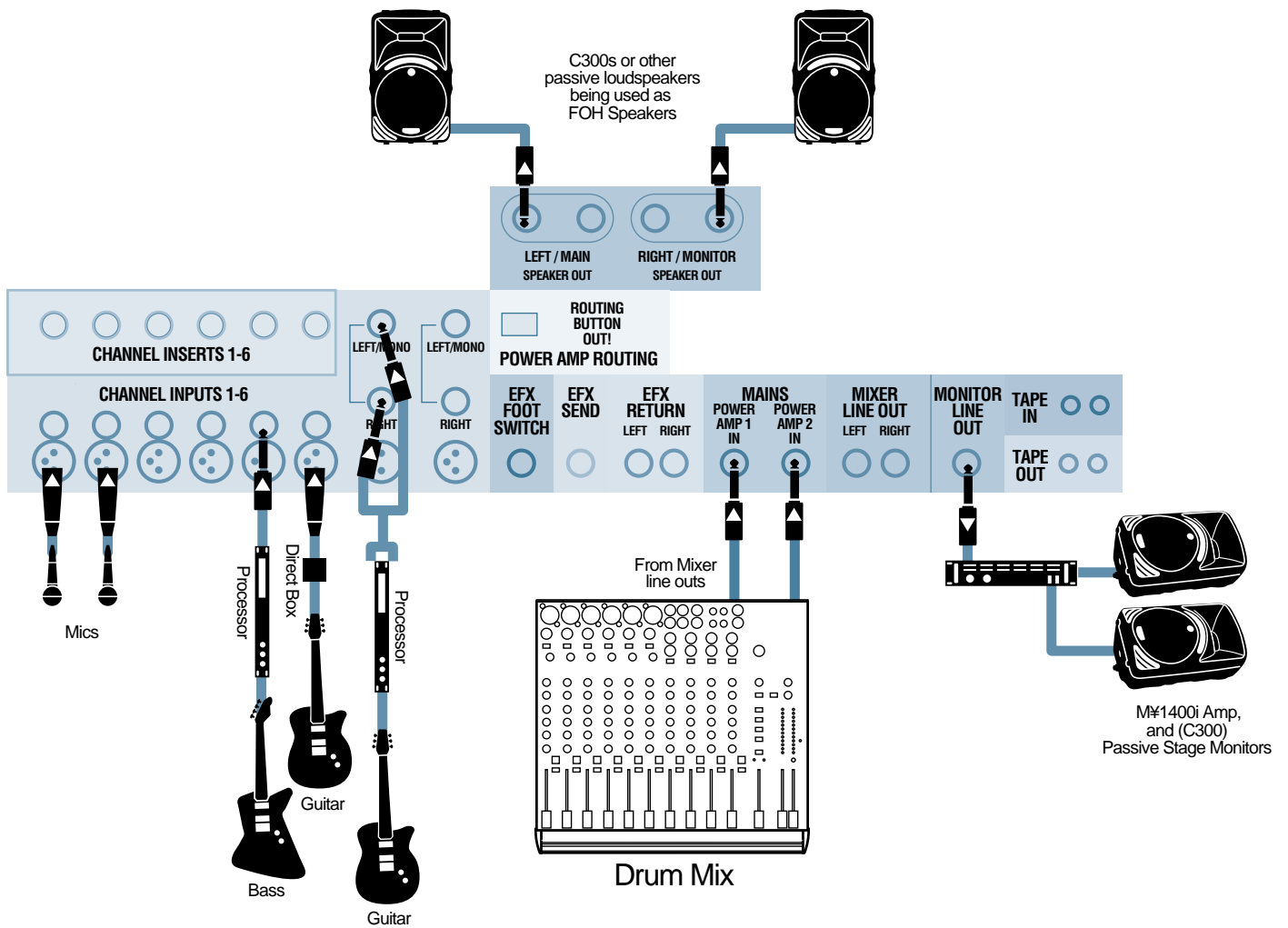
808S Club Stereo PA with Subwoofers (Using Electronic Crossover)

This shows how to use subwoofers with the powered mixer. The MIXER LINE OUTs are routed to an electronic crossover, which splits the signal into highs and lows. The low-frequency (LF) outputs from the crossover are connected to an external amplifier that powers the subwoofers. The crossover's high-frequency outputs aren't used since the full-range signal passes through to the internal amplifiers and the FOH speakers. Note: If you have a 406M, 408M, or 808M, there is no stereo output because these are mono versions. The Main Mix bus is routed to both internal amplifiers and there is just one MIXER LINE OUT in these models.



Adding More Inputs to the PPM

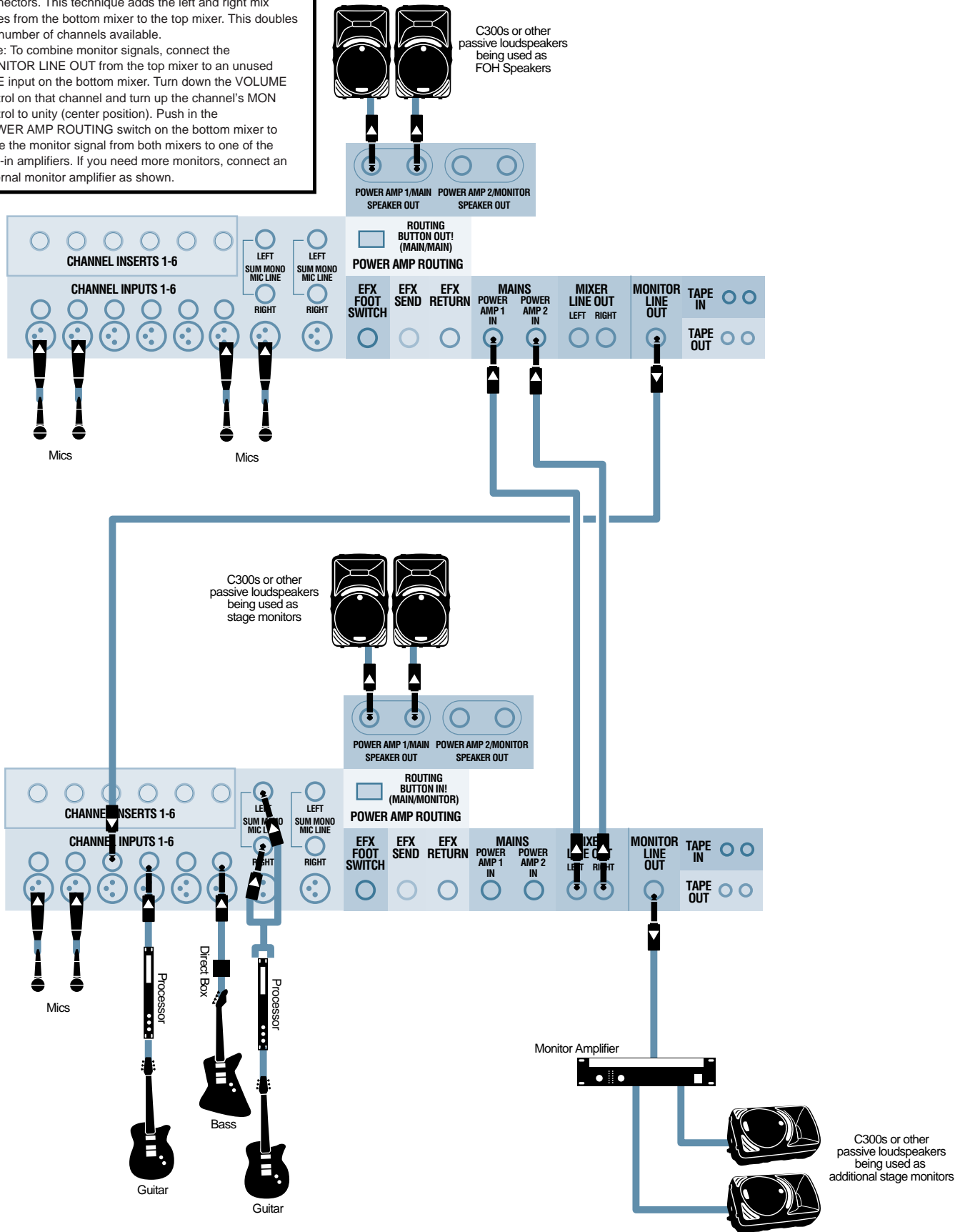
808S With an External Mixer



Two 808S' Stacked for More Inputs

This shows how to connect two powered mixers together using the MIXER LINE OUT and POWER AMP IN connectors. This technique adds the left and right mix buses from the bottom mixer to the top mixer. This doubles the number of channels available.

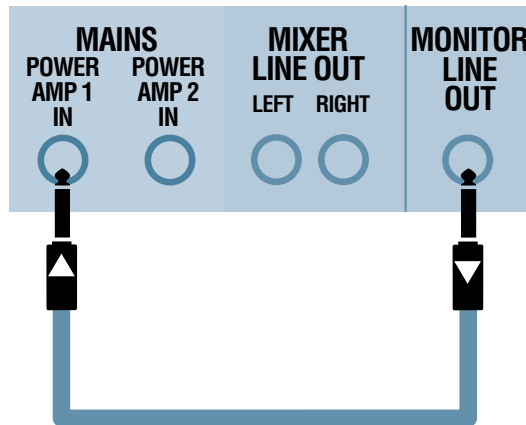
Note: To combine monitor signals, connect the MONITOR LINE OUT from the top mixer to an unused LINE input on the bottom mixer. Turn down the VOLUME control on that channel and turn up the channel's MON control to unity (center position). Push in the POWER AMP ROUTING switch on the bottom mixer to route the monitor signal from both mixers to one of the built-in amplifiers. If you need more monitors, connect an external monitor amplifier as shown.



Another Approach

In the previous example, the Left/Main power amplifier is unused and an external power amplifier drives the second set of monitor speakers.

By patching the MONITOR LINE OUT on the lower mixer to the L POWER AMP IN, you'll send the monitor signal to the LEFT power amplifier. A second pair of monitors can be connected to the LEFT/MAIN SPEAKER jack. The POWER AMP ROUTING switch should be in the Main/Monitor position (*in*).



This illustrates the flexibility of the PPM series.

Power Amp INSERTs

On mixers later than Serial Number *****, the POWER AMP IN jacks are wired in the “normalled through” configuration allowing you to use an *Insert Cable* for patching in an outboard 1/3 octave graphic equalizer, limiter, or use the PPM in a bi-amplified speaker system with an electronic crossover.

Earlier mixers can be modified to convert the “input only” POWER AMP IN jack to a normalled through configuration which will accommodate an INSERT configuration. Contact Mackie Tech Support for details if you wish to modify your own mixer.

Chapter 25: PA Setups: DFX Series PA Applications

This application is very similar to the 1604VLZ-PRO PA setup. The PA is set up in stereo, but here are some tips for scaling your system down in size if that's appropriate for your venue.

Scaling Down—Mono PA

When working with a small mixer in small clubs, you may find it advantageous to operate the PA in mono rather than stereo.

A common setup is to use one channel of a stereo power amplifier for the main speakers and the other channel for the monitors. Since there's only one pre-fader AUX send on these smaller models, you're likely to be using only a single monitor mix anyway.

Mono MAIN Output

The simplest way to get a mono output to the main speakers is to pan all the input channels to one side and simply use that one MAIN output to feed the speakers. You could also pan all inputs to the center and use either output.

You can also use a Y-Cable to combine the mixer's LEFT and RIGHT MAIN OUTPUTS (either the XLR or 1/4" jacks) without harming the mixer.

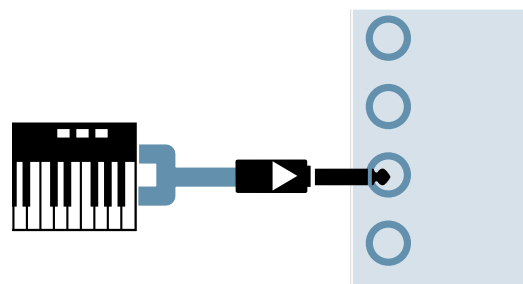
Stereo Sources in Mono



True stereo sources can be a bit of a problem. Turning the PAN knobs fully to one side will send only one channel of the stereo source to the panned output. That could mean that you won't hear one end of a keyboard or won't hear the normal balance of drums from a drum machine. Panning to the center and using only one output will provide a mono mix of stereo inputs.

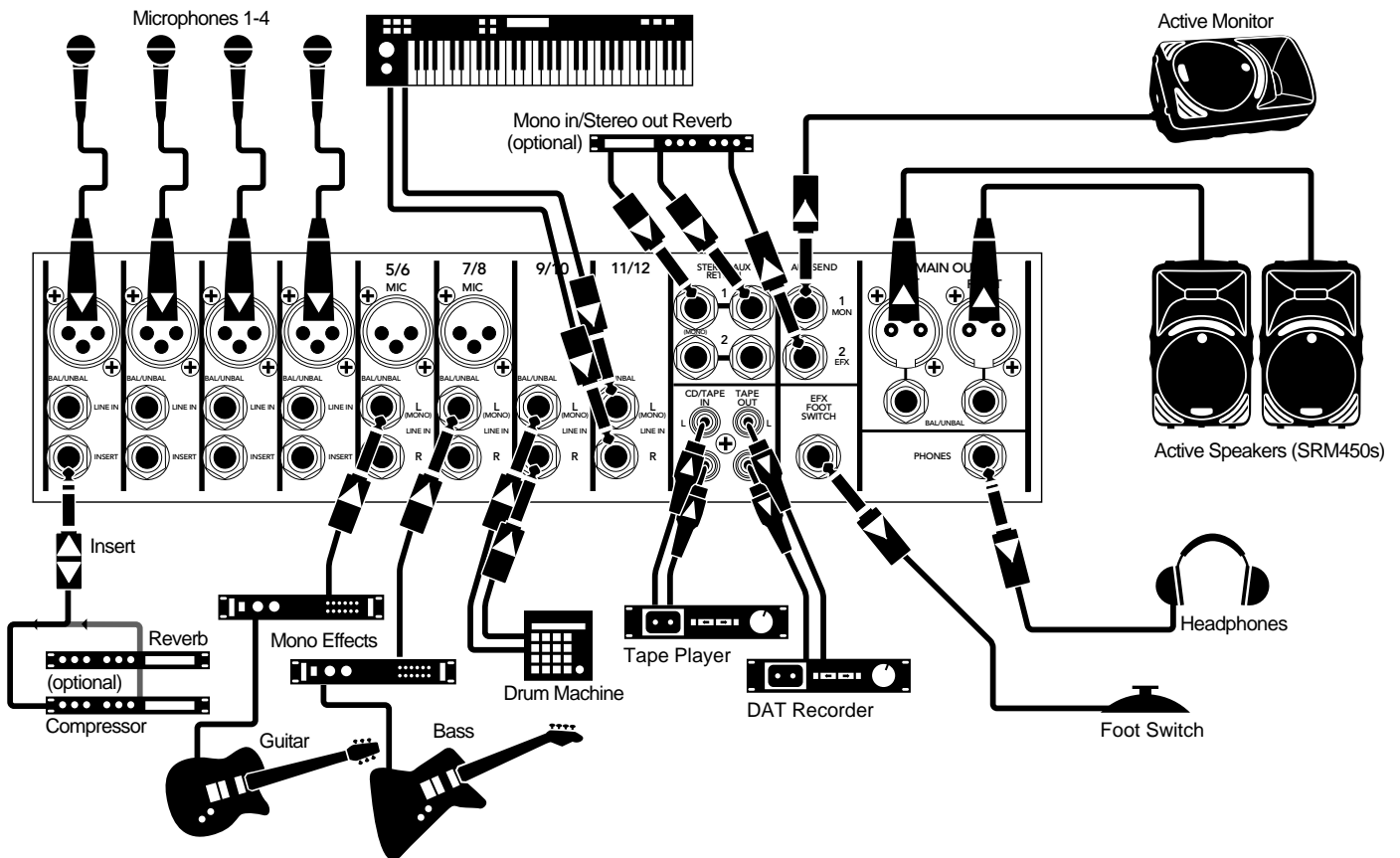
Fortunately, most stereo instruments provide a mono output by plugging into only one of the outputs. Check the manual or the label on the jacks. If you can get a mono output from your stereo keyboards or drum machine, simply plug it into the LEFT input of a stereo channel. Then you can pan the mono source wherever necessary.

If the instrument doesn't provide a mono output from one jack, you can use a Y-cable to parallel the two outputs.

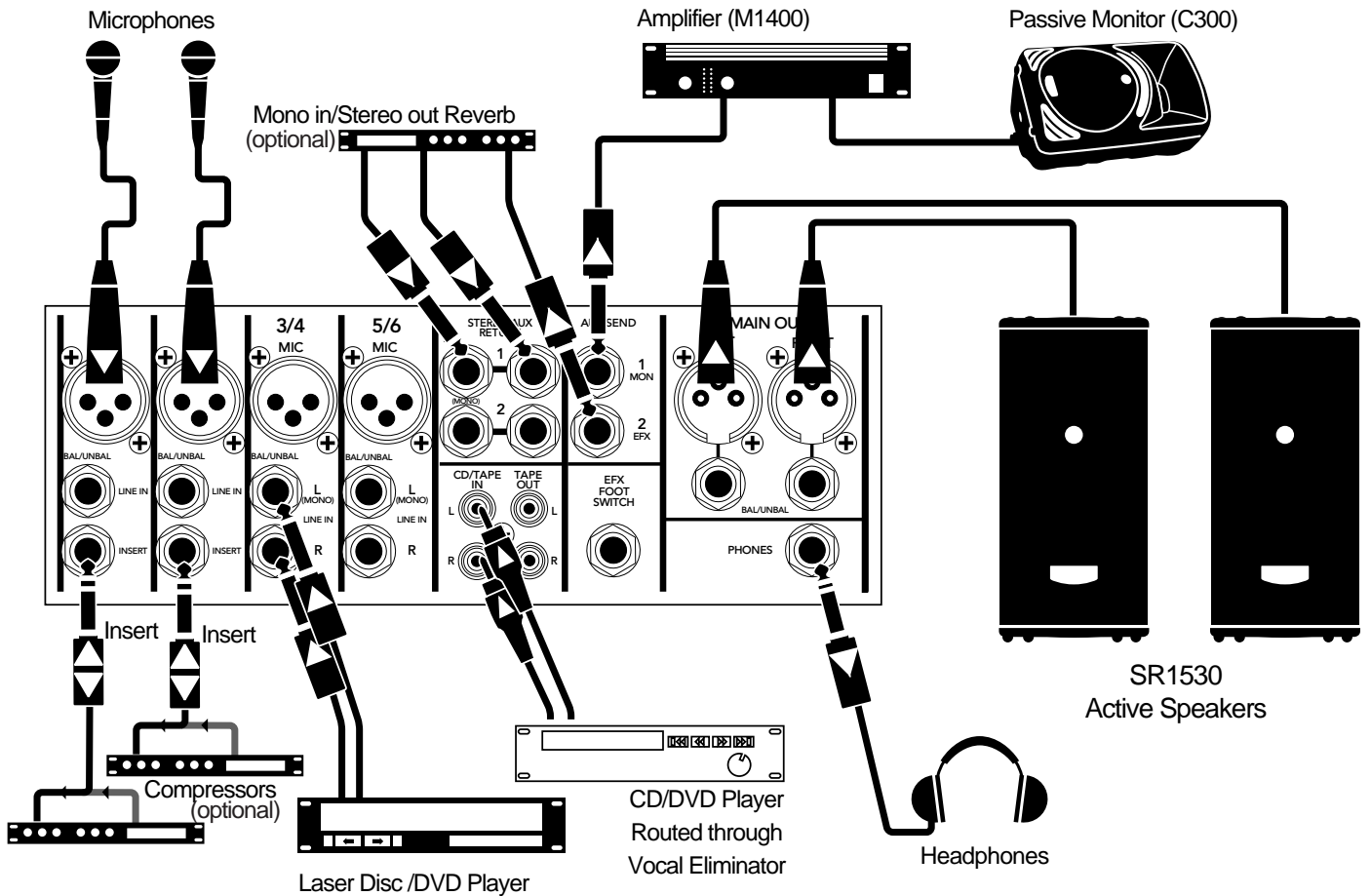


Listen closely to your synth patches if you're running the PA in mono. Some synths have stereo effects or patches that rely on out-of-phase material in the two channels. They sound gloriously spacious in stereo, but the effect, or even the entire sound, can disappear when the two channels are summed to mono.

DFX•I2 Small Club Gig

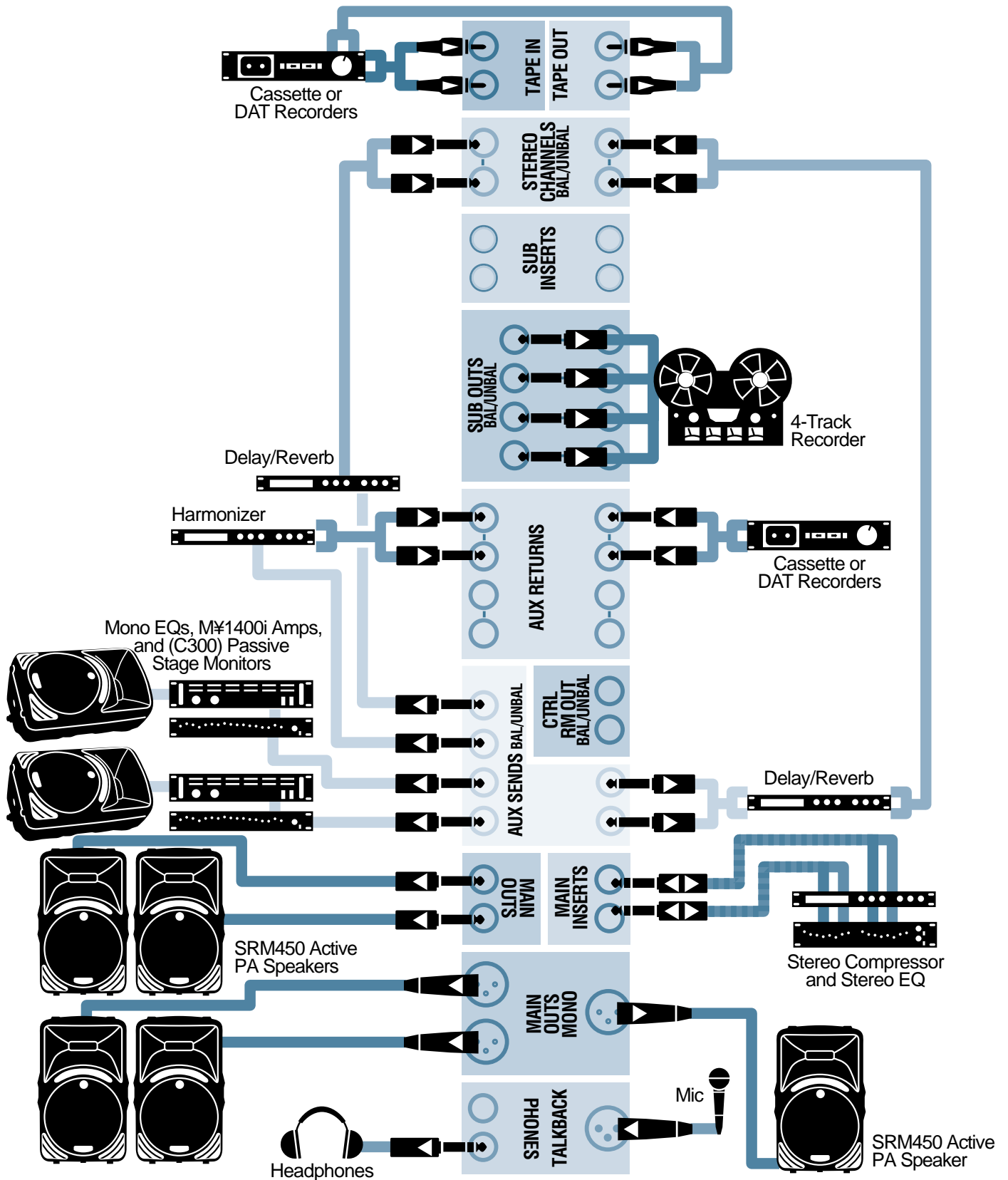


DFX•6 Karaoke Set Up



Chapter 26: PA Setups: SR Series Large PA Applications

Club PA



Inputs

For clarity, we've not shown the inputs on this diagram. Here's a typical setup for a rock band showing the sources, channels,

inserted processors, and channel output assignments:

Chan	Source	Input	Insert	Assign
1	Kick	Mic		1-2
2	Snare	Mic	Gate	1-2
3	Hi-Hat	Mic		1-2
4	Floor Tom	Mic		1-2
5	Lo Rack Tom	Mic	Gate	1-2
6	Hi Rack Tom	Mic	Gate	1-2
7	Drum O/H L	Mic		1-2
8	Drum O/H R	Mic		1-2
9	Bass Amp	Mic		L-R
10	Bass DI	Mic	Compressor	L-R
11	Lead Guitar Amp	Mic		3
12	Rhythm Guitar Amp	Mic		3
13	Acoustic Guitar DI	Mic		3
14	Piano Low	Mic		L-R
15	Piano Hi	Mic		L-R
16	Keyboard L	Line		L-R
17	Keyboard R	Line		L-R
18	Vocal 1 (Lead)	Mic	Compressor	L-R
19	Vocal 2	Mic		4
20	Vocal 3	Mic		4
21-22	StereoReverb 2	Line		L-R
23-24	Stereo Delay	Line		L-R

The PA system will be run in stereo, with drums and keyboards panned to fill the stereo field. Bass, guitars and vocals will be centered in the stereo field, and effects will be returned in stereo. This will give some stereo spread to the sound without losing important parts of the mix to audience members who are seated off center.

Guitars are routed to Subgroup 3 by assigning them to 3-4 and setting the PAN knob full left. The lead singer is assigned directly to the main L-R mix and panned to the center. He gets a compressor all his own

through his channel INSERT. Backup vocals are routed to Subgroup 4 by assigning those channels to 3-4 and setting the PAN full right. A compressor connected to the SUB INSERT 4 jack rides herd on the backup singers.

Keyboards and the piano are assigned L-R and panned a little off center but not fully left and right. Drums are assigned to Subgroups 1 and 2 and panned to taste across the stereo field. Some individual drum channels have gates patched into their channel INSERTs, and the full drum mix is

compressed with a stereo compressor connected to SUB 1 and 2 INSERTs.

Subgroup Assignments

All of the subgroups are assigned to the MAIN MIX by engaging their L/R ASSIGN buttons. We want the guitars and vocals to be centered in the stereo mix, so the PAN pots for SUB 3 and 4 are centered. Since panning of the drums is established by the position of the PAN pots on the drum input channels, setting SUB 1's PAN fully left and SUB 2's PAN fully right will give the full stereo spread. If you find that you've spread the drums too wide, bring them closer to the center using SUB 1 and SUB 2 PAN controls.

Main Speakers

These are fed from the MAIN outputs. We've shown Active (powered) speakers, but a power amplifier and passive speakers will work just fine.

A secondary power amplifier and speaker fed from the MAIN MONO output can be placed to fill a dead spot in the main room, or placed in another location such as the lobby or bar area.

Monitors

AUX 1 and AUX 2 are used for the two stage monitor mixes. AUX SEND MASTERS 1 and 2 control the monitor volume. Pressing their adjacent SOLO buttons allows you to check the monitor mix in the phones and read the level on the meters to assure you don't overdrive the monitor amplifiers when the singer yells "More monitor" over the din.

Effects

In this setup, we have two reverbs and a stereo delay fed from AUX SENDs 3-6. In live sound applications, it's fairly common to return effects to channels rather than AUX RETURNs so there will be more control available and, most importantly, they can be muted at the touch of a button. Reverb 1 outputs are connected to AUX RETURN 1, while Reverb 2 and the delay outputs return to the mix through the two stereo channels 21-22 and 23-24.

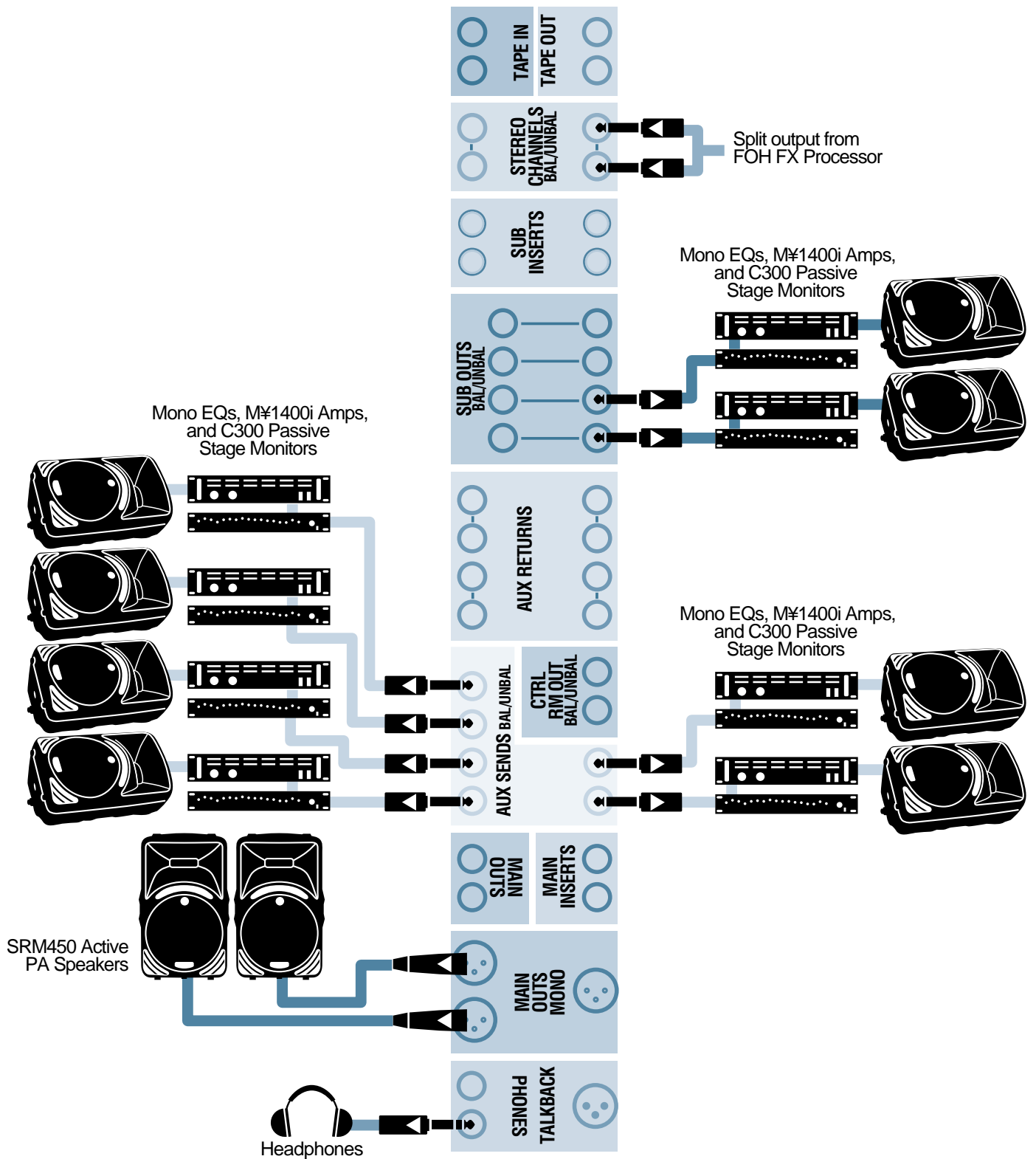
Tape

A recorder is connected to the TAPE OUT jacks to record the stereo mix. The record level follows the MAIN MIX fader. A playback deck or CD player for intermission music is connected to the TAPE IN jacks. Playback volume is controlled by the TAPE RETURN knob.

Talkback

You can talk to the stage or make announcements to the house using a mic plugged into the TALKBACK MIC jack. Pressing the AUX 1-2 button in the Talkback section sends the mic, through the TALKBACK LEVEL control, to the stage monitors. Pressing the MAIN MIX button sends the talkback mic to the main speakers.

Stage Monitor Mix



The purpose of a monitor mixing console is to provide a large number of different mixes from the same set of sources. Some of those mixes may be quite similar, differing mostly by what's not included in one mix or the other. Others mixes may be really minimal—just the singer and kick drum to keep him in time. Monitor mixes are rarely for anyone's enjoyment, they're to allow the players to hear what they need in order to give the best performance. This is usually an unbalanced mix, favoring either one's own voice or instrument, or a voice or instrument with which the performer needs to blend or get the song's timing.

About Monitor Mixing

A successful monitor mix engineer anticipates, or better yet, *learns* what each artist wants to hear in his monitor mix. It's equally important to know what they *don't* want to hear. A singer needs to keep time and hear pitch, so he'll probably want a little kick and snare drum in his monitor, but not the drum kit at full blast. It's really difficult to hear pitch with a chorused keyboard as a reference, so most singers just trust that the keyboard player will do his thing when it's time and get their pitch reference from a bass or rhythm guitar.

As we've seen in the general PA applications, monitor mixes are usually created from pre-fader AUX SENDs, allowing you to create completely independent mixes. The SR series consoles are equipped with up to four pre-fader sends (two are permanently post-fader). In this application, we've employed those for four of the monitor mixes, plus we've made two other mixes using the channel faders routed to the SUB outputs.

Inputs

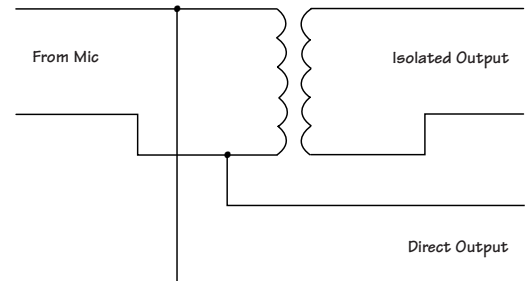
In nearly all cases, the inputs from the stage are shared by the monitor and front-of-house (FOH) consoles by using an outboard splitter. For this application, we'll assume that's the case.

Mic Splitting and Phantom Power—Who Supplies It?

It's becoming fairly common to use condenser mics on stage, so phantom power for those mics will be necessary. You and the FOH engineer will have to agree on who's

supplying the power for the mics and connect the splitter accordingly. The splitter system may be designed so it can be wired only one way—that makes the decision easy.

Most splitters isolate the two consoles with a transformer to eliminate ground loop problems.



dc won't pass through a transformer, so only the console which has the direct connection to the microphones can power them. With a transformer-isolated splitter, it doesn't hurt if both consoles have phantom power turned on, but only one console can actually power the mics. If that console, or its phantom power supply fails, both the house and monitor systems lose the powered mics. (but of course this won't happen with your Mackie)

Some inexpensive splitters are nothing more than Y-cables, occasionally with resistors in line, and those should not have phantom power applied from both consoles. Slight differences in voltage between the two consoles can cause one console to draw current from the other, and may not leave enough for the mics.

Outputs

AUX SENDS 1-4 feed four stage mixes. Engaging the PRE buttons make AUX 3 and AUX 4 pre-fader, the usual preference for monitor mixing.

We're using two SUB outputs for additional monitor mixes. Those are by nature post-fader, but changes that you make to these mixes using the faders won't affect the other mixes. It does mean, however, that those two mixes aren't completely independent. In this application, we're using one SUB mix for drums and the other for a downstage vocal monitor so the lead singer can hear himself when he moves to the edge of the stage.

Sidefills, which generally have a fairly well balanced mix, are fed from the MONO MAIN output. There's rarely room (or requirement) for stereo in monitor mixes

except for some very fussy performers, so all the monitor mixes in this application, including the sidefills, are mono.

Powering the Monitor Speakers

Each mix output gets its own graphic equalizer and power amplifier. The equalizer serves several purposes. It fine tunes the frequency response of the speaker, it serves to notch out feedback frequencies, and it also is used to adjust what's coming from the monitor speaker to blend it with other sound on stage. A performer who's standing close to the back of the main FOH speakers will hear a lot of low frequencies coming from them, so you'll need to take a lot of lows out of his monitor so he doesn't hear an over-bassy mix.

Effects

It's nice to be able to send some reverb or delay to the monitor speakers, particularly for singers, so we've included two effect processors in this example, fed from post fader AUX SENDS 5 and 6. By returning the effect processor outputs to channels rather than AUX RETURNS, we have the flexibility to assign (or not assign) the effect to any individual mix. Since nothing in the monitor mix is stereo, there's no point in taking a stereo output from the effect processor, or for using both sides of a stereo input. By connecting to the Left channel only, the effect return channel becomes mono.

Cue Speaker

With all the commotion on stage, all the amplifiers turned up to 11, the drummer torture-testing his sticks, and everyone shouting "MORE MONITORS" at you, you won't be able to hear a doggone thing in the headphones. Give it up. There's hope, however, since the CONTROL ROOM OUTPUT duplicates the headphone output. By connecting it to a power amplifier and speaker, you can compete with the din and hear what you're sending to the monitors.

The CONTROL ROOM output normally carries the MAIN L-R mix, but pressing any SOLO button interrupts that feed and sends the SOLO bus to the CR outputs. Want to hear what the snare drum sounds like? SOLO the snare channel. Want to hear what Monitor Mix 3 sounds like? SOLO the AUX 3 MASTER. To check one

of the mixes fed from a SUB, SOLO that subgroup.

You'll get the most accurate picture of the monitor sound if your cue speaker is the same type as your stage monitors. You'll never exactly re-create the stage sound in your cue mix since there's a lot of other sounds beside your monitor mix going into the artist's ears, but by hearing what you're sending, you can anticipate an artist's needs and hear the effect of your adjustments.

Assigns

Here's where you need to get creative. Of course any channel can be sent to any of the pre-fader AUX SENDS, but since we're using the MAIN and SUB outputs for additional mixes, we need to assign channels to those busses.

Start with the sidefill mix (the MAIN MONO output), which is generally a fairly well balanced full band mix. Assign anything that you want in the sidefills to the L-R bus. The position of the PAN pots is arbitrary at this point since we're feeding the sidefills in mono. Set the channel faders for a well balanced mix. Don't be afraid to use the channel EQs to help balance and clean up the mix.

Since SUB 1 is a drums-only mix, assign all the drum channels to 1-2 and set their PAN pots full left. Now SUB 1 will contain the drums, mixed as you mixed them in the sidefills. You may have to make a compromise here if the drummer wants more kick in his monitor than the rest of the band can stand in the sidefills. Before you raise the fader on the kick drum channel, try using the graphic EQ in the drum monitor signal path to emphasize the bottom end and the beater attack frequencies. That will usually get the job done without changing the balance in the sidefills.

Assign the channels that you want in the downstage vocal mix (SUB 2) to 1-2 and pan them fully right. You can add a little of the drums to this vocal mix by turning the PANs on the drum channels slightly toward the center from their full left position. And if the drummer wants to hear some vocals, turn some of the PANs on the vocal channels a bit toward center from full right.

Talkback

The talkback mic input allows you to talk to AUX 1-2 and the MAIN mix (sidefills). If

that gives you enough coverage so that you can yell at whoever you need, use it. If you want to be heard in all the monitor speakers, plug a mic into a spare channel. You can then send it to any (or all) mix outputs by using the AUX and ASSIGN controls on that channel.

Protocol

While this isn't exactly about mixers, here's a good place to mention that, as the monitor engineer, you're the one the band is going to glare at if things aren't right and they'll love you if the sound on stage is great. If you cooperate with the band, they'll help you to do your job. If you fight them, they'll fight back.

You also need to make friends with the FOH mixer, because what you do with the stage monitors can affect what people hear in the house. Remember, since you're sharing mics, your feedback is also his feedback.

Because of the high sound levels involved, with everything running right on the edge of both feedback and clipping much of the time, mixing monitors can be the most stressful job in the sound business. But if the monitors are great, you'll be the first one the band pats on the back when they leave the stage after a great set. That's a nice feeling.

Chapter 27: Multitrack Recording with the 8-Bus

Since you've bought a pretty large mixing console, we expect you'll occasionally be doing pretty big sessions. This section will describe at least one way of approaching a large session—one with a full drum kit, bass, guitar, piano, vocal, horns, and synthesizer. We'll go through recording the basic tracks and then doing overdubs.

Setup

Let's assume you have a 24-track recorder (we hope it's a Mackie HDR24/96) with its outputs connected to the TAPE RETURNS and its inputs connected to the SUBMASTER/TAPE OUTPUTS. The 8-Bus console triples the outputs of the submaster busses (see "double bussing"), so the inputs to recorder tracks 1, 8, and 17 are duplicated, making it possible to send any input channel to any track.

Effects inputs and outputs are patched to a couple of SEND/RETURN pairs.

Organizing A Multitrack Session

Decide where everyone's going to stand or sit in the studio, then draw a diagram of your studio setup, showing mic positions. Then assign an input channel to each mic.

Making logical microphone channel assignments now will help avoid confusion later. Group similar instruments together. If there's a left-to-right relationship to a group of mics (like drums mics or a vocal group), keep the same left-to-right sequence on the console. It's OK, in fact good practice, to leave some unused channels between groups, so if you decide you want to add one more drum or amplifier mic, you can put it near the others in its "family" rather than all the way at the opposite end of the console.

Plan your basic track assignments the same way. It's very confusing to have sources randomly strewn across the console when you're mixing. Don't put the sax track in between the floor and rack tom tracks.

Now, normal your console (also called "zeroing"). This means check the position of every switch and every knob to be sure they are in the normal, or expected posi-

tion. Your normal may be different from someone else's, but generally it means all switches off or up, all knobs either all the way down or at their Unity or centered position.

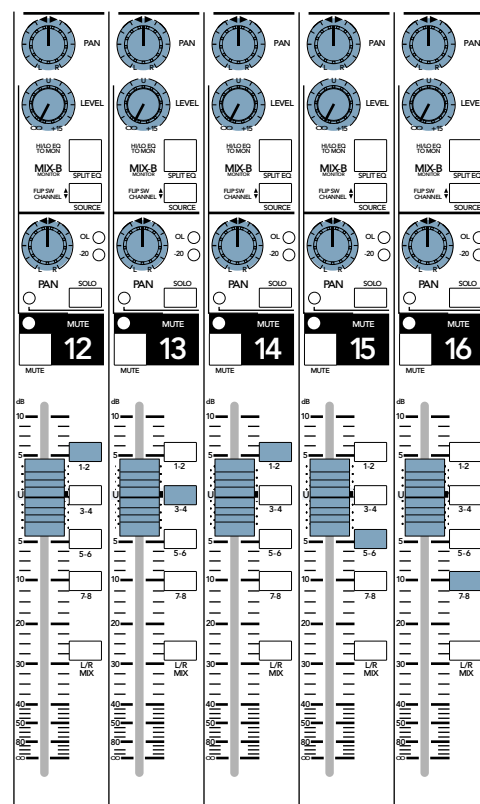


This is a good time to review the first part of this book to refresh your memory about the functions of some of the controls. A button in the wrong position or a level turned all the way down can cause a lot of hair-tearing when you expect to hear something and don't.

If a control has two available modes, for example, AUX 3-4/5-6 SHIFT or AUX 3-6 SOURCE (which you might want for tracking, but we'll get into that in a bit), now is the time to set them all.

Label and Document

Label all your input channels—do it as you plug in each mic, DI, or synth. It'll save a lot of spaghetti handling later. Put a strip of 1/2" or 3/4" white paper tape below the channel faders and write on it with a non-smearing pen.



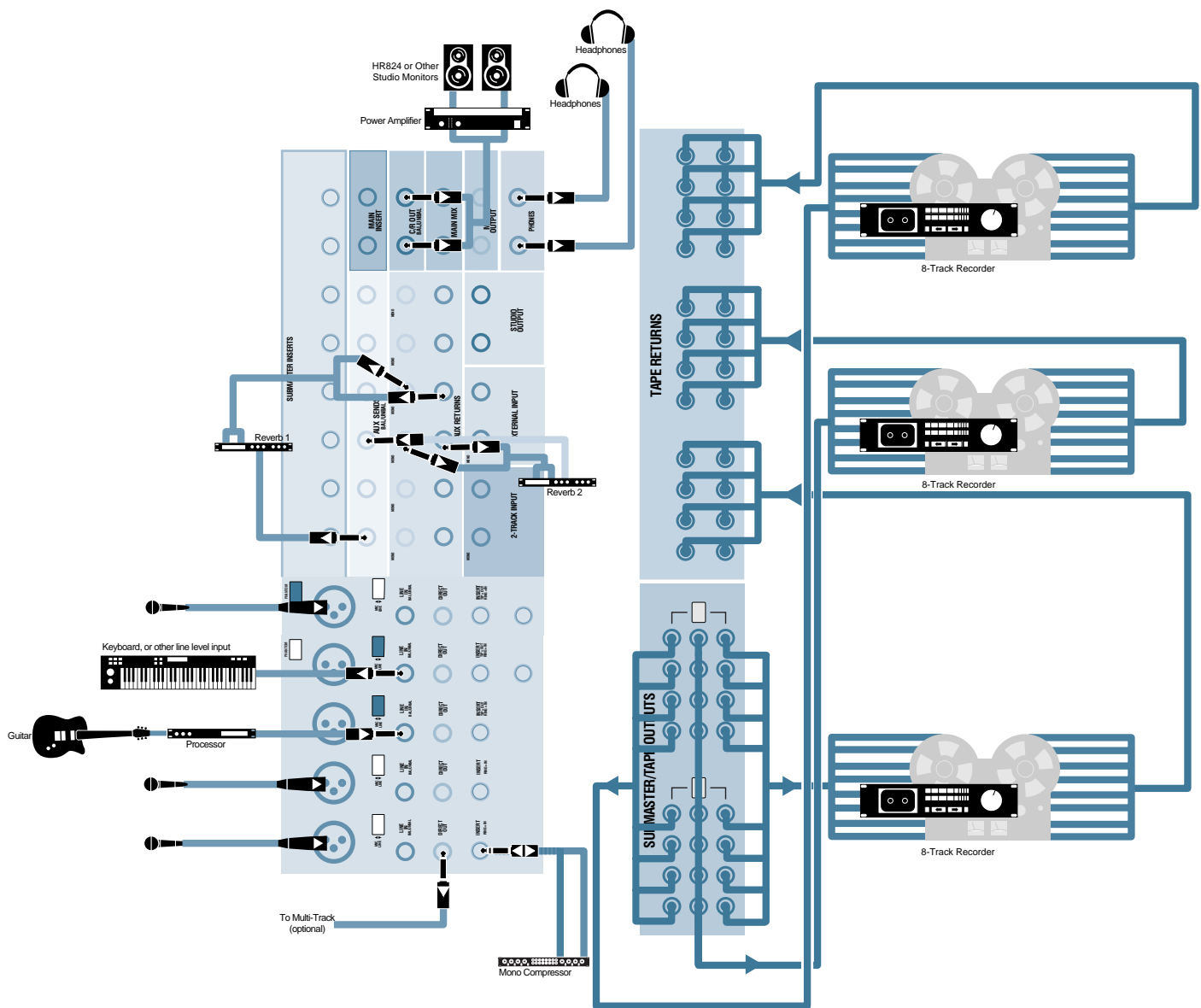
Avoid masking tape or duct tape. It will slime your console. Take a trip to an office, art, or recording supply store and get some low-tack (non-gummy) artist's or correction

tape. We like the Post-it™ brand tape and Sharpie™ pens.

Depending on how you bus and patch channels to recorder tracks, the MIX-B channel names (where the tape returns are coming in during tracking) may not correspond directly to the input channels. Prepare a label strip for the MIX-B section and label each track as you record it. You'll need some narrow tape to squeeze labels in between the top of the faders and the MUTE buttons, but it'll save you some confusing later. Here's an example:

Since, unless you've crossed up some cables, the MIX-B channel numbers correspond to recorder track numbers, when it's time to mix and you've FLIPped the channel inputs from MIC/LINE to TAPE, you can just move that MIX-B scribble strip down to the main faders and you'll have them labeled for mixdown.

Refer to the illustrations on the next two pages for the basic hookup and a bird's eye of the controls you'll be using.



[illegible]

TRACKING PART 2: THE BOARD

Using Buses

If you need to combine two or more inputs into one or two outputs (here we're recording the toms and drum overheads on two tracks), you must assign the inputs to a common bus or pair of buses.

If you have only one source going to one tape track, you have a choice of how to get it there: you can assign the channel to the bus that's connected to that track, or you can patch the recorder's input to the channel's DIRECT or INSERT Out. Take your pick.

Reasons for using a bus:

- It's there, it's easy, it sounds fine.
- You meet interesting people on the bus.

Reasons for using a DIRECT Out:

- You've already assigned all eight of your buses to other duties.
- You're a purist and you think you can hear the additional circuitry used in a bus.



Using the INSERT Out is actually the cleanest signal path if you're recording without applying any EQ to the track. Use a cable with a 1/4" TS plug inserted to the first click of the INSERT jack.

MIX-B Monitoring

The most convenient way to monitor during a tracking or overdub session is by listening to MIX-B.

Routing the Recorder to MIX-B

Set both the FLIP and MIX-B SOURCE buttons to their *up* positions. (Put this on your "zeroing the console" checklist) This connects the TAPE RETURN inputs to the MIX-B level control.

Select MIX-B as the MONITOR source in the Control Room or Phones matrix. With the recorder connected as shown on the hookup diagram, MIX-B will be a mix of the recorder outputs, exactly what you need to hear.

Recorder Monitor Switching

Just about every multitrack recorder made in the past 20 years is equipped with automatic input/playback monitor switching.

With the recorder in the Auto Input Monitor mode it will automatically switch its output between source and playback as you go between stop, play, fast wind, record and so on. That way, you're always hearing your instruments after they travel through the multitrack deck, regardless of whether or not the tape is rolling. With the recorder stopped or in Record, you hear what's going to the track on the tracks which are armed for recording. In Play, you hear what's recorded on the track.

MIX-B Monitor Mix

MIX-B allows you to set up a custom mix of the recorder tracks independent of the input channel output levels. You can adjust level, pan, reverb and even EQ in your monitor mix while you record or listen to a playback without disturbing the settings on the input channels. This way, if you decide to make another pass, everything is set and ready to roll. You can even connect a cassette recorder to the MIX-B outputs to record a rough mix of the session.



When using MIX-B to monitor the tape returns, be sure that none of the input channels are assigned to L-R. Otherwise, you'll be monitoring a channel from two sources (the input and the tape return) when recording and only the tape return on playback. The difference in sound will confuse you.

Headphone Cue Mix

Once you have your input channels assigned to recorder tracks, set the Phones 1 and 2 SOURCE to MIX-B. This way, musicians wearing headphones plugged into the console Phones jacks will have something to listen to as they tune up and rehearse.

Putting the recorder in the All Input Monitor mode will assure that everything routed to the recorder gets into Mix-B (don't forget to turn the MIX-B levels up!). Remember to switch the recorder back to Auto Input Monitor when you're ready to start tracking, though. Otherwise you'll panic when you hit Play and don't hear the playback.

Alternate Cue Mixes

Using the Phones on Mix-B might work for the entire session, but usually the players will want one or more custom mixes in their headphones. Like stage monitor

mixes, these aren't necessarily what you'd like to hear in the control room, but they help the players keep in time and in tune. The bass player and drummer may want bass and drums to be the loudest thing in their cans so they can stay in the groove, but that would annoy the vocalist to no end. Most likely, no sooner will you get a pretty good sounding mix than someone will ask for a different one.

AUX SENDs 3-4 and 5-6 can be used to set up two different cue mixes from source that's feeding Mix-B which in this case is the recorder returns. Set AUX Sends 3 and 4 to Pre-fader (their PRE buttons *down*) and select MIX-B as the AUX 3-4 Source by pressing the SOURCE button at the bottom of the AUX SEND section *down*. Note that with MIX-B selected as the AUX 3-4 source, "Pre" refers to the Mix-B level controls, not the channel fader.

With the console configured in this way, you'll be able to create a custom mix on either AUX Sends 3-4 or 5-6 depending on the setting of the AUX SHIFT button. It's almost like having another Mix-B section (Mix-C?).

When AUX 3-4 or AUX 5-6 is selected as the Phones SOURCE, sends 3 and 5 feed the left headphone and 4 and 6 feed the right headphone. When both knobs are in the same position, the sound will appear centered in the headphones. You can pan the source to the right or left by offsetting one or both knobs. It's not as convenient as a pan pot, but it works.



If you find that you still need more different headphone mixes, consider making the AUX 1-2 source modification (see Modifications in the Appendix). That will allow you to use all six AUX sends for Mix-B (tape returns) cue mixes.

Wet or Dry Monitoring?

Usually, you don't record multitrack tracks wet (with reverb) unless it's an integral part of a sound, because you can't undo it later. It sometimes helps to hear a little reverb in the monitors as you're working, however, and with the 8-Bus, you have the option of wet monitoring.

Reverb Sends

Since you're monitoring MIX-B, that's where you should get the signal that you send to your reverb unit. When the SOURCE button below Sends 3-6 is depressed (MIX-B) but the PRE button is not, these sends are fed after the MIX-B level control and make great wet monitor reverb sends.

RETURNS 3 and 4 are a convenient place to connect the outputs of the reverbs that you want to use in your monitor mix. Once you have your sends happening, you can send RETURNS 3 and 4 to the headphones by pressing the Phones 1 or Phones 2 buttons next to those Return level controls in the STEREO AUX RETURNS matrix. Even if you are using Send 6 to feed the reverb, there is no reason not to patch the reverb output into Return 3 or 4. The Send and Return numbers do not have to match.



But what about the extra cue mixes you made using AUX 3-4? Well, engineering, like life, is full of compromises. There are only so many knobs and jacks and even on the largest consoles, eventually you run out and have to seek a different path. You could set up your additional cue mixes on AUX 1-2 using a combination of the sends on the input channels for the sources you're recording, and the tape returns (press the LINE/TAPE switch *down*) on the recorded channels.

Reverb Returns to the Monitor

Getting reverb into the control room monitors is a little more involved since there is no return assignment switch to MIX-B. You have three choices:

- Assign MIX-B to L/R Mix in the MIX-B master section. Then, instead of monitoring MIX-B in the control room, select L/R MIX as the monitor source. Use Stereo AUX Returns 1-6 to bring the reverb into the L/R Mix buses. There is no disadvantage to this patch, unless you are already using the L/R Mix buses for some other function.
- Patch the returns into unused tape returns using the Tape In jacks on the rear of the console. Then route them into MIX-B, just as if they were additional tape tracks. This is a darn good reason to have purchased a console with more channels than you have recorder tracks. You did, didn't you?

- Patch the returns into unused channel strips using the Line In jacks. Then you can route the reverb anywhere your heart desires.

Let's Record!

Here's one way to set the board up (another option is shown in the hook-up drawings. Either works just fine):

Kick Drumto Channel 1 Direct Out to Track 1
 Snare Drumto Channel 2 Direct Out to Track 2
 Cymbals Leftto Channel 3Bus 3 to Track 3
 Cymbals Rightto Channel 4Bus 4 to Track 4
 Tom 1to Channel 6 panned between Buses 3 & 4 to Tracks 3 & 4
 Tom 2to Channel 7 panned between Buses 3 & 4 to Tracks 3 & 4
 Tom 3to Channel 8 panned between Buses 3 & 4 to Tracks 3 & 4
 Bass Ampto Channel 9Bus 5 to Track 5
 Bass Directto Channel 10Bus 5 to Track 5
 Scratch Vocalto Channel 11Direct Out to Track 11
 Guitar near to Channel 12Bus 6 to Track 6
 Guitar farto Channel 13Bus 6 to Track 6
 Piano Lto Channel 14Bus 7 to Track 7
 Piano Rto Channel 15Bus 8 to Track 8
 Trombone Ito Channel 16 panned between Buses 1 & 2 to tracks 9 & 10
 Trombone IIto Channel 17 panned between Buses 1 & 2 to tracks 9 & 10
 Flugelhorn to Channel 18 panned between Buses 1 & 2 to tracks 9 & 10
 Trumpetto Channel 19 panned between Buses 1 & 2 to tracks 9 & 10

Your monitoring and cue signals come from the MIX-B inputs corresponding to the tape tracks:

KickTrack 1Tape Return 1MIX-B
 SnareTrack 2Tape Return 2MIX-B
 Drums LTrack 3Tape Return 3MIX-B
 Drums RTrack 4Tape Return 4MIX-B
 BassTrack 5Tape Return 5MIX-B
 GuitarTrack 6Tape Return 6MIX-B
 Piano LTrack 7Tape Return 7MIX-B
 Piano RTrack 8Tape Return 8MIX-B
 Horns LTrack 9Tape Return 9MIX-B
 Horns RTrack 10Tape Return 10MIX-B
 Scratch VocalTrack 11Tape Return 11MIX-B



Notice in this example that the kick, snare, and scratch vocal channels are patched directly from the channel DIRECT OUTs to recorder inputs since those tracks contain nothing but a single source. The piano also gets a track for each of its mics, but we've chosen to use the busses to feed the recorder just for convenience. Feel free to patch the piano direct.

The bass amp and bass DI are both assigned to Bus 5 and recorded on Track 5. By adjusting the balance between the amp and

direct signal you can get a wide range of tones from the bass. Since the two sources are mixed to a single track, you'll have to live with the bass tone you establish at tracking time. Keep in mind what else will be going into the mix and blend the two inputs for a tone that will fit. The two mics on the guitar amplifier (one close, one distant) are also recorded on a single track, so blend them for the tone you want while you're tracking.

Drums and the horn section are each recorded to a pair of tracks, setting their levels and panning with the channel faders and pan pots while listening to the stereo mix of those inputs from MIX-B.

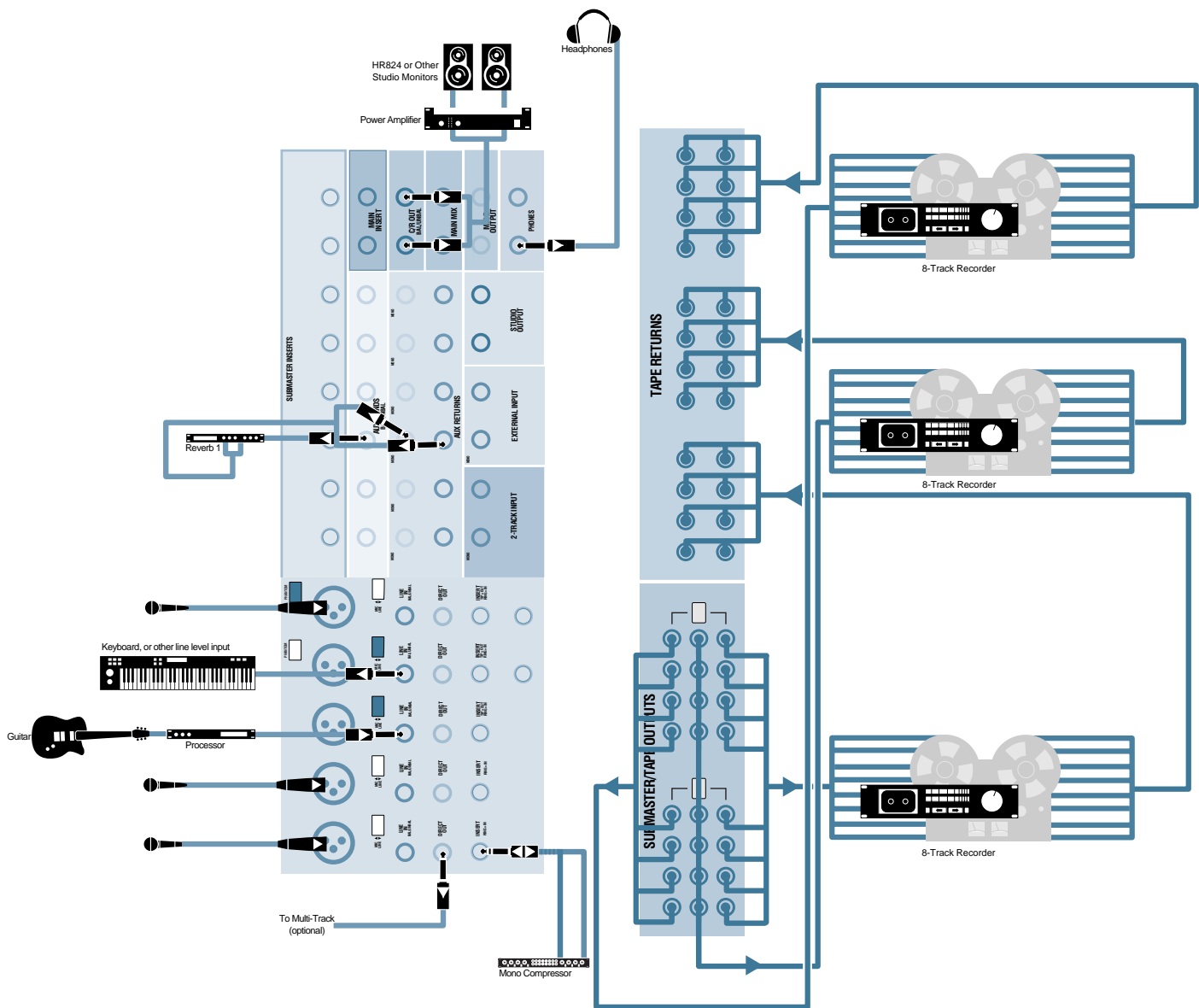
Once you're patched, assigned, and made some notes, recording the basic tracks should pretty well take care of itself. Keep on top of the players—be sure they're in tune, keep them tight (but not *too* well lubricated). You'll have great basic tracks before midnight.

Document! Document!

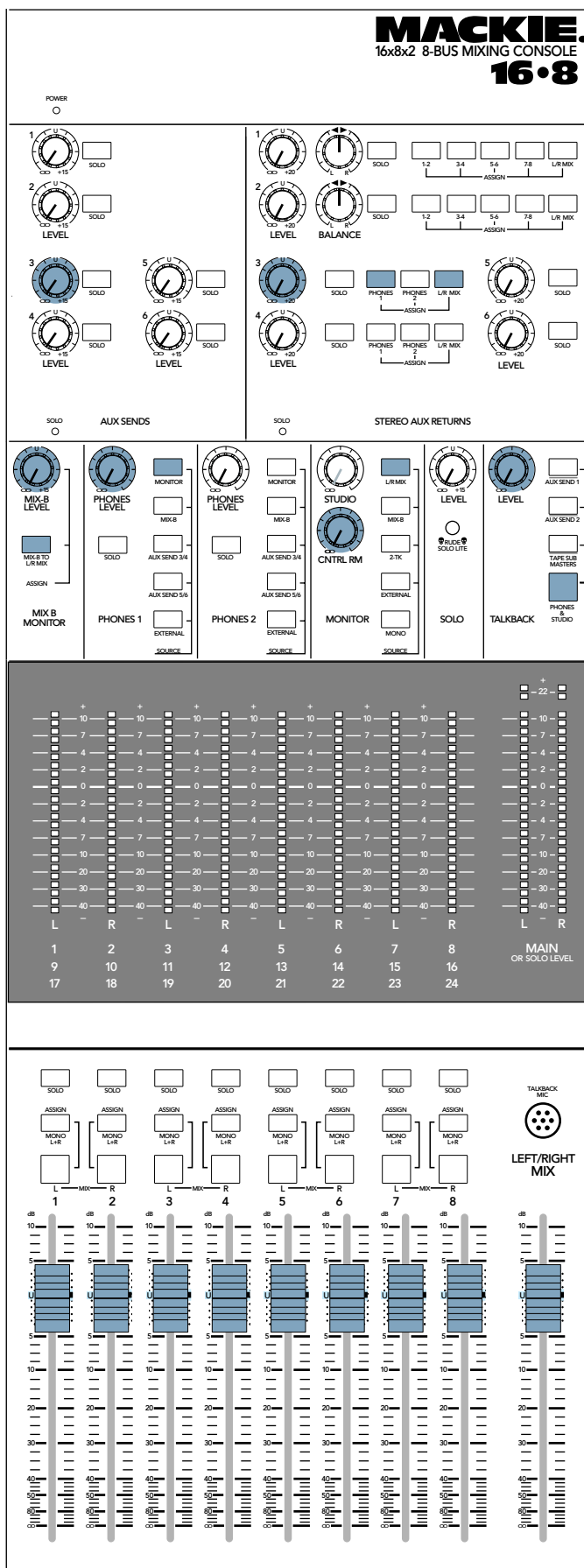
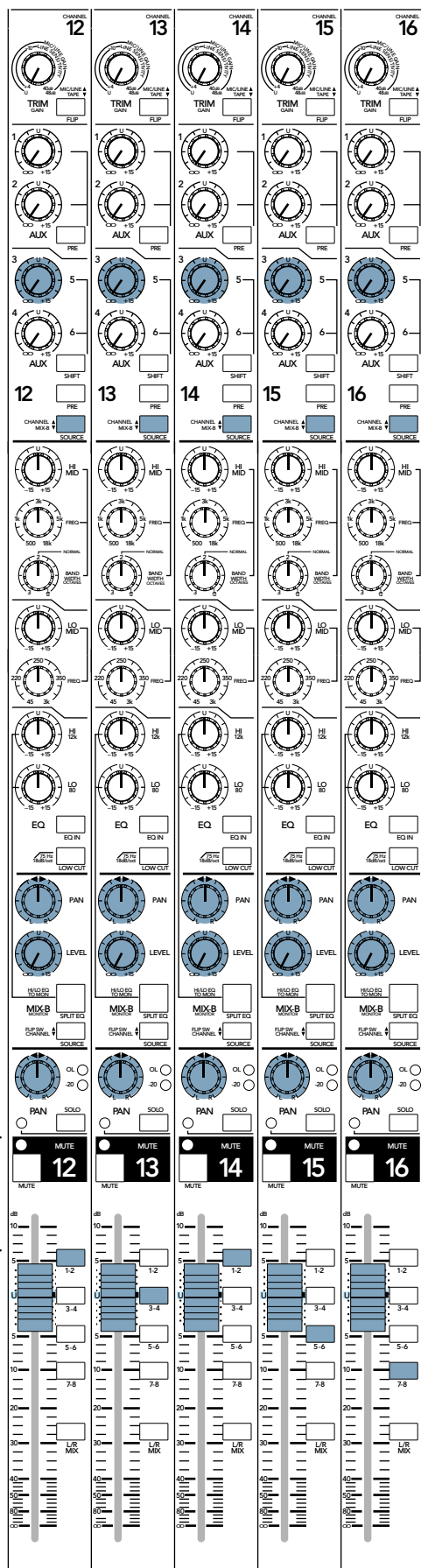
Once you've got your basic tracks down, take a few minutes to log all your settings, right down to headphone send levels and outboard compressor thresholds. You may be back next week doing it all over again, and you won't remember it all. Some engineers photograph the console with a Polaroid™ camera. Others like to grab a mic and a cassette, and just read the console settings into the recorder as if they were telling a story. To reset the console, listen to the playback and just follow your instructions.

One easy way to document your console is to copy the 8-Bus panel layouts from the end of this manual and mark your settings on them in color.

After you're happy with the basic tracks, it's time to start overdubbing the instrumental solos and vocals. The next two illustrations show typical hookups and the console controls.



Assign channels and adjust panpot to the appropriate bus for the desired tape channel input.



Tinted knobs are used in this application

Tinted buttons are pressed IN

OVERDUBBING PART 2: THE BOARD

Overdubbing

Start your overdub session by normalizing all the channels that you'll be using for instruments or vocals. If you're doing guitar overdubs with the same setup as you used for tracking, you can leave those as you've set them for tracking, though you might want to EQ a guitar differently when it's playing lead than when it was part of the rhythm section. Reset EQ, sends, anything that does not affect your MIX-B monitor and cue mixes. Do not change the FLIP switch settings yet.

Unless you have a string section or the Tower of Babel Choir coming in, overdubs are generally recorded one instrument or singer at a time, so you won't need all of the input channels. Pick a convenient input channel or two and use them as your inputs for overdubs. As you record on different tracks, just reassign the channels to the appropriate bus outputs to get them to the tracks you want, no sweat. With the triple busing feature, unless you're a purist, you won't have to re-patch anything to feed up to a 24-track recorder.

Since your monitor and cue mix setups haven't changed (or you've reset them accurately according to your notes), during overdubbing, you'll hear the basic tracks just as you recorded them, using the same set of controls to adjust the cue and control room monitor mixes. At the end of the night, you can run the monitor mix into a cassette or DAT and take a rough mix home.

Flipping Out Now

If you're going to start mixing right after you finish your overdubs, you have another setup option you might find appealing. Pick an input channel or two for your overdub mics that's out beyond the number of recorder tracks you have. If you're using a 24-track recorder, choose channels 25 and 26 for the overdub source inputs. Those are the channels you'll send to the recorder tracks using the Assign switches and pan pots.

Now, depress the FLIP buttons on channels 1-24 (including channels that you'll be using for overdubs). This will route the TAPE RETURNS to the main channel faders, and you can begin working on your mix with the full set of channel controls while monitoring the L/R Bus. You can work on

EQing the horn section while the guitarist is on his 23rd take of the big solo.

Headphones can still be fed via the AUX Send of your choice, or by selecting MONITOR as the Phones source. When you're done overdubbing, mixdown will be just about done, or at least you'll have a good idea of how it's going to sound. Amaze the band!

Bouncing Tracks

You never have enough chocolate, mixer inputs, or tracks. Bouncing, a technique for making your tracks go further, is a combination of mixing and overdubbing. Recording the four background singers four times in stereo gave you a really solid chorus, but the eight tracks you used didn't leave the guitarist much room to layer his part. So, after those multiple vocal parts are recorded, set up a mix on a pair of subgroup busses consisting of just the vocals, and record that mix on another two tracks.

Listen to that mix *really well*. When you're comfortable with the blend and you're satisfied that you've ducked all the clunkers, grit your teeth and erase the original 8 tracks, then tell the guitar player to get to work. Now you have six tracks available and haven't lost your huge chorus.

Mixing Overview

Recording and overdubbing requires some tender loving care on the part of the recording engineer, but the focus during tracking really is on the performance. It's important to get a good sound, but it's more important to keep the musicians really happy with what they're doing, keep the energy up, and snag that killer track when it happens.

Good mixing, however, focuses solely on your creative and technical engineering skills. In tracking, you've taken the song apart and reduced it to a bunch of individual elements, all meticulously recorded. Now it's up to you to blend the tracks so they sound like music again. On the technical side, you must take into account such things as the sound of home and car speakers, mono compatibility, how human perception changes under different listening conditions, matching your recording to the sound to similar products in your market,

not to mention tonal and level balance between songs, and meeting the technical requirements for mastering and replication.

Tune Your Ears

Get a CD or DAT of a song with a sound that you'd like your mix to approximate, and patch the player into the EXTERNAL jacks on the console. Then you can compare your mix against this sonic model at the flick of a switch to see how closely you're approaching your desired sound.

You'll be surprised at how easy it is to focus on a particular sound in your model, say the snare, if you listen to the CD and then listen to just the snare in your mix. This technique can help you zero in on a sound that you're trying to achieve.

Mixing Setup

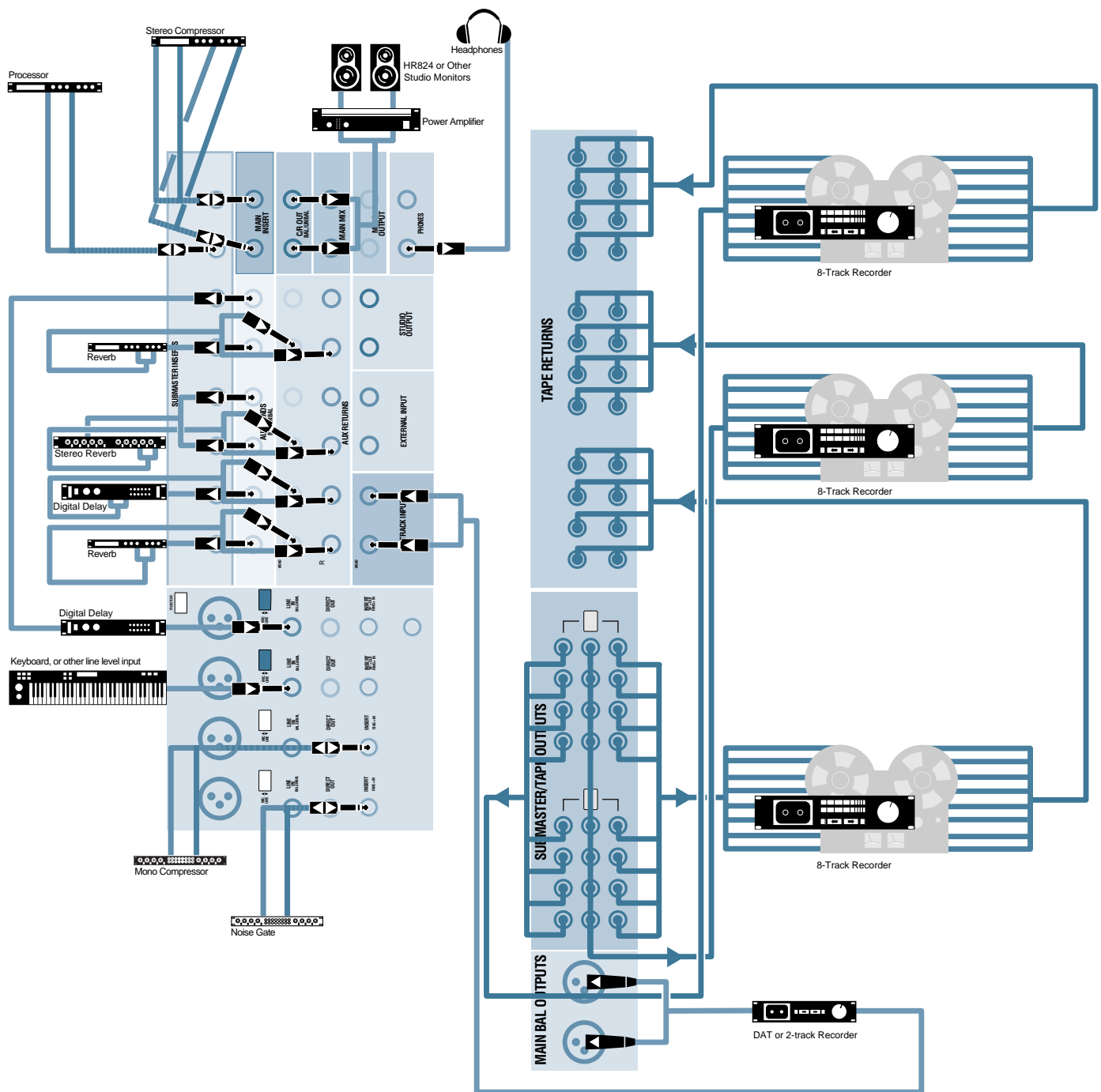
Clean and align your tape deck according to the manufacturer's instructions. If you're using a digital recorder, sacrifice a full floppy disk in its presence to ensure smooth operation during mixdown.

Try to get your inputs organized so that all your vocals will be on adjacent channels, all the drums on adjacent channels, and so on. If you were diligent in tracking, you should be pretty close. As overdubs tend to grow with the number of available tracks, you may want to re-patch some of your tape returns to get things re-organized. There's no technical reason why the output from Track 23, the last guitar overdub that you saved has to go to mixer channel 23 when the other "keeper" guitar parts are on tracks 12 and 13. They're just signals.

Re-label the faders. You've probably dripped sweat all over the old labels and can't read them any more, and besides, they now represent the recorder tracks rather than mics.

You might also find it handy to put 1/4" strips of tape vertically along each fader to mark levels on channels that you'll have to ride during the mix.

Your setup will look something like this:



CHANNEL 10 CHANNEL 11 CHANNEL 12 CHANNEL 13 CHANNEL 14 CHANNEL 15 CHANNEL 16

TRIM COARSE FINE MIC/LINE FLIP PRE SHIFT AUX SOLO MUTE

EQ IN EQ IN EQ IN EQ IN EQ IN EQ IN EQ IN

LOW CUT LOW CUT LOW CUT LOW CUT LOW CUT LOW CUT LOW CUT

PAN SOLO MUTE

10 11 12 13 14 15 16

Drums

Bass Guitar

Guitar

Keys

Vocals

Keys

Vocal Delay

MACKIE
16x8x2 8-BUS MIXING CONSOLE
16.8

POWER

AUX SENDS

STEREO AUX RETURNS

MIX B MONITOR

PHONES

LEVEL

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

LEFT/RIGHT MIX

Drum L

Drum R

Rev 1 L

Rev 1 R

Vox L

Vox R

Keys 1

Keys 2

Tinted knobs are used
in this application

Tinted buttons
are pressed IN

CONTROL

MULTITRACK MIXING PART 2: THE BOARD

Assigning Inputs and Outputs

Engage the FLIP switches and assign all input channels to L/R (unless you are using the submasters to group channels—stay tuned). The FLIP switch sends the TAPE RETURNS to the channel inputs and the ASSIGN sends the channels to the MAIN mix. Lastly, select L/R MIX as your Monitor Source.

Level Management

There is a tendency for levels to creep upwards as you add more and more tracks to your mix. One way to keep a handle on this is to start mixing with the L/R master fader set a few dB above its Unity Gain position. As you get closer and closer to your final mix and the level creeps up, ease the master fader down toward Unity Gain, which is where it should be. This will keep the monitor level fairly constant, so your ears don't get fooled with too loud a mix.

Pull all the channel input faders down and you'll be ready to start mixing.

Consider Compression

You can mix an entire project without a lick of compression; many engineers do. The dynamic range of a CD can certainly handle it. But consider—most people will be listening to your mix under less than studio conditions. There's background noise and road noise, distractions, and many people won't listen as loud as you mix (though others will listen considerably louder).

A little gentle compression, whether on individual tracks or on the entire mix, can reduce the dynamic range a bit and help to pull your mix together. Also, if you want to simulate what your mix will sound like over the airwaves, you can compress the heck out of it, like they do. They use very fancy compressors, but any compressor will give you an idea of what will happen. This is good for checking things out, but not for your final mix.

Doing the Mix

Nearly all multitrack recorders have some sort of *locator* that will allow you to rewind to the beginning of the song and park there until you're ready to start a mixing pass. Set up a locate point at the beginning, and perhaps other locate points at significant

places in the song so you can get to them quickly.

If your recorder will do it, get into loop mode so it will just play the song over and over. This is a timesaver when you're starting out and just want to get a feeling for the controls.

Start by mixing a group of tracks that run throughout the song, usually the drums or the rhythm section.

Set the panning, level, EQ, reverb and delay, and bring in more tracks as the mix begins to gel. Don't make any level marks on your fader position tape yet, but as the mix comes together, try to note which sections work without a lot of fader moves. Look for what appears to be the loudest part. Sometimes turning the Control Room Level way down, so the mix is very quiet, will reveal what sounds are clearly louder than others.

Now take a listen to the recording that you're using as your reference model. Make some adjustments in your mix to get it closer to the model. When you think it's getting reasonable, go to that loudest section that you located, and pull the master fader back to get the main mix meters reading close to 0 VU. Then go to the section that works by itself and start making little marks on the fader strips. You're getting close.

Listen to the model again. Start making more marks for the moves. Repeat until it sounds like a hit. Serves four adults.

Using "Outboards"—External Signal Processors

Compressors, gates and equalizers are generally inserted in line with the signal path. We call them "serial" processors since they're in series with the signal path. All of the signal goes into the device, then out and back to the mixer's signal path.

Reverb, echo, and spatial enhancement gizmos and are usually set up as send/return or "parallel" devices. Some of the signal is "borrowed" from a channel via an AUX Send, sent to the device, processed and returned to the mixer as a new, wet signal via the AUX Returns, to be mixed with the original, dry signal.

Delays, chorus and flanging devices, and, aural exciters can work in either the serial or

parallel configuration, depending on what they're doing. If you're delaying just one guitar, use a delay in a channel INSERT. If you're applying reverb to the all of the voices in the background vocal section, set it up in parallel.

Insert Devices—Where?

A compressor/limiter after EQ will compress differently than one inserted before the EQ. A compressor/limiter inserted before a fader will limit consistently at any volume setting, but one after the fader will effectively have its threshold moved by the fader level. None of these choices are right or wrong, they just sound different—as the engineer, it's your choice.

There are three common points for inserting processing devices in the signal path during mixing or tracking:

- In a pre-EQ channel insert point (for one channel only). This works on the channel input signal. When tracking, limiting here will prevent overloading when the singer bellows. When mixing, compression will allow you to raise the average level of the track, but you may still need to “ride” the fader to pick up soft words.
- In a submaster INSERT point. This affects a group of signals assigned to that submaster.
- In the L/R MAIN INSET point. This affects the whole mix. A compressor here is generally used to increase the apparent loudness without overloading on peaks.

Compressors and equalizers are “straight through” devices—you want to return *only* the processed signal to the channel, not a portion of the unprocessed signal. Reverbs and other “coloring” effects are usually very much over-the-top when the fully “wet” output is used in a mix. That's why that breed of effect processor has a wet/dry mix control which adds some of the unprocessed signal back in to the processed signal. When patching a reverb in-line with a single channel, adjust the reverb's Wet/Dry control for the desired amount of “wetness” to the signal going back into your mix.

Send/Return Devices

Since you usually don't need multiple headphone mixes during mixdown, you will normally have all six AUX sends available. Use a couple as your primary reverb

sends, perhaps one for a bright plate and the second one a chamber with a slap or pre-delay. That leaves you four for special effects. You can also use the MIX-B outputs or an unused subgroup bus if you need additional sends. Keep the sends post-fader unless you don't want the effect to follow the fader moves.

If you're only using an effect on a single channel in the mix, you can use the channel Direct Out as a send. Since the DIRECT or INSERT output doesn't have a separate level control, use the Input Level control on the effect device to adjust the “send” level to the effect.

On the 8-Bus console, the DIRECT OUTPUT is post-fader (and EQ, too) but the channel INSERT is pre-both.



Remember that here is no rule against sending a signal out to a processor on AUX 3 and patching its outputs to RETURNS 5 or 6. If your reverb has a mono input (usually one of a pair of inputs is designated Mono), try feeding it a mono signal, using just one AUX Send. You'll still get a good stereo reverb sound and you can make your AUX SENDS go further this way. Stereo inputs are useful when you want the reverb to appear in the same position in the stereo field as the source, but that's something that rarely occurs in nature.

If you're using a mono effect or only one channel of a stereo effect, connecting its output to the Left input jack of a return will place the effect in the center of the mix, or if you're using Returns 1 or 2, it can be panned anywhere.

If you plug the effect output into the Right jack, the effect will be placed on the right side in your mix. To put the effect on the left side only, patch the return into the Left jack, and put a dummy plug into the Right jack. That will defeat the left-goes-to-center normalling and send the signal directly to the left bus.

Of course you can also use empty channel inputs as effect returns. Simply patch the reverb output into a Line Input. Bonus: you now have console EQ available on your reverb return.



Feedback Alert! Be sure the AUX Send that feeds an effect processor is turned fully down on the channels being used as effect returns from that processor.

Using Subgroups

There will be times during mixing that you will want to set up subgroups within your mix. A subgroup allows you use just one or two (for stereo) faders to control a larger group of tracks (say drums, horns or background vocals). This makes for easier mixing, and also allows you to patch a single (or a pair for stereo) EQ or compressor on the set of tracks.

The Mackie 8-Bus console offers you several options using subgroups, depending on the situation.

To assign channels to a subgroup during mixing, first un-assign the channels you want to subgroup from the L/R MIX. Then choose the bus or pair of buses you'd like to use as a subgroup and assign the channels to that subgroup. Subgroup busses come in pairs, but you don't have to use both.

For a mono subgroup, the channel PAN pot selects to which of the two busses the channel signal is sent. For a stereo subgroup pair, the PAN pot controls the position of the channel source in the stereo field.

For example, if you want to use Bus 1 as a mono subgroup, assign the desired channels to 1-2 and pan those channels fully left. To group channels in stereo on SUBs 3 and 4, assign the desired channels to 3-4 and set the panning of each channel so its image appears where you want it in the mix. Trumpets on the left, trombones on the right.

But you're not done yet. Those subgroup busses need to go somewhere. You have three easy choices and a more flexible one that's a little more complicated.

There are two ASSIGN buttons associated with each SUB fader, a MONO (L+R) and either a LEFT or a RIGHT button depending on whether it's an odd or even numbered subgroup. If you want all the saxes you've assigned to SUB 1 to go in one heap on the left, press the LEFT button above the SUB 1 fader. If you want them all in the center, press *both* the LEFT and MONO buttons. If you want them on the right, take two steps back. Assign them to Bus 2 by swinging the Pan pots fully clockwise, then press the RIGHT button above the SUB 2 fader.

A stereo subgroup will have one odd and one even numbered fader. Assign one LEFT, the other RIGHT, and the channel

PAN pots can then be used to position the grouped sources across the stereo field.

So how do you move that pile of saxes somewhere off center, but not fully left or right? You'll need to send the SUB output to an input which can be panned anywhere in the L-R mix. If you have an unused channel, patch the SUB output to the channel LINE INPUT, assign that channel to L-R and pan to your heart's content. Bonus: You get to use the channel's EQ for overall equalization of the subgroup, and can use the AUX sends to put effects on the subgrouped channels.

If you've filled up all the main input channels, this would be a darn good use for all of those tape returns going to MIX-B that have been loafing during mixdown.

Finding More Inputs: MIX-B to L & R Buses

Your unbridled creativity will find ways to use up every jack, whether you are routing the vocals through a pair of Leslie speakers or keying a gated set of reindeer bells with the snare signal.

The Mackie 8-Bus consoles can't give you the infinite number of channels you dream of, (unless you buy an infinite number of console expanders—we'd like that) but you can double the number of mixdown inputs by using MIX-B. How's that for a deal?

If you're mixing using the setup from a few pages back, you engaged the channel FLIP switches to re-route the recorder returns into the main channel fader and EQ. FLIP also switches the Line Input jacks to the MIX-B inputs (it's those two sets of inputs that are flipped), and the line inputs can now be used as extra inputs to the MAIN L/R mix.

Since you can assign the MIX-B to the MAIN L/R bus, when FLIPped, the Line Inputs now go to the MIX B level controls, the MIX-B PANs, and then on to the L/R bus. Check over in the MIX-B MONITOR section above the Sub meters and you'll see the MIX-B TO L/R MIX ASSIGN button. That's the one that does the trick. Voila! Twice as many inputs!

You can get an AUX Send for the extra inputs by using the SOURCE switch in the AUX Send 3/4/5/6 area, and you can SPLIT the channel EQ if you need to EQ

the MIX-B input. You'll have to do some clever planning since there really are a finite number of resources, but we've provided the controls to route them to a few extra useful places.

As we mentioned above, you can use the AUX Return inputs as additional inputs to the mix, too.

Monitoring and Listening Levels

Check your speakers and amplifiers to be sure that playback levels are balanced left-to-right and the speakers are mounted symmetrically to your mixing position. A 2 dB shift in monitor balance will produce a 2 dB shift in the opposite direction in your mix.

Also, check your speaker polarity (sometimes inaccurately called phase). This is a basic thing we all know about, but it's amazing the number of times we've found studio speakers (especially near-field monitors, which are connected and disconnected regularly) wired with opposing polarity. You should train your ears to notice out-of-polarity conditions instantly. It's easy to hear (to us it sounds like a combination of not hearing enough bass and feeling like our eyes are slightly crossed), and getting polarity right will save you much grief in mixing.

Remember that your mix needs to sound good on anybody's system. Be sure you have some real-world listening speakers available in addition to the monitors you like so well, and check back and forth frequently. There's nothing like making a cassette of your mix and playing it in your car to tell you how your mix will sound when . . . well, when it's played on a car cassette player. Try it at several speed (driving speeds, that is, not cassette speeds) so you can get a sense of what parts of your mix get masked by road noise. An outboard switch box is handy for listening on alternate monitors in the studio, but you can also use the Studio Outputs of the console to feed alternate monitors.

Check your mix at different volumes, too. A mix that sounds great loud may not necessarily sound good at a low volume. Listen at a barely audible level from time to time. You should still be able to hear the essential components of the mix.

Also, check your stereo mixes in mono regularly during the mixing process. Much television and radio is still heard in mono, and your mix has to sound its best in mono as well as stereo.

Take a hint from the film mixers and set your dialog or lead vocals to about 85 dB/c at the mixing position. This is a moderately loud but still normal volume. If you have a sound pressure level (SPL) meter available you can take a measurement to get a feel for how loud 85 dB is. Radio Shack® has an inexpensive and remarkably accurate one (you're an analog engineer—get the analog model, not the digital model). No self-respecting set of ears should be without one.

Monitoring at a known and consistent volume will keep you honest, and keep your mixes balanced for playback. Sure, listen at very low levels, and crank it from time to time to remember why you're in this line of work, but stay at the moderate 85 dB C-weighting setting most of the time. You'll save your hearing and also make better mixes.

A Word About Automation

There are some outboard MIDI controlled fader/mute systems which you can add to your console to provide automation for your mixes. If you feel the need to automate, shop carefully and learn about the system you choose. For those of you without automation, there is hope. Billions and billions of great mixes have been done on non-automated consoles. Here are a few tips:

- Use subgroups, discussed earlier.
-
- "Mult" tracks that need EQ or effect changes during the mix to two channels, alternating between the two channels with the MUTE switches. *Multing* means connecting one output to two or more inputs. You can make mult boxes or just use "Y-cables".
- Enlist several sets of hands.
- Bounce a track that needs more moves than you can handle in a pass to a new track, riding the fader or EQ during the bounce. This allows you to concentrate on just the one track that needs attention, pulling up a word here, ducking a gasp for breath there, and boosting a couple of dB at 4 kHz when the singer

turned away from the mic for a word. When you have the perfectly tweaked track, you can just put it up in the mix and leave it alone—one less hand and about a million fewer brain cells needed to concentrate on the mix.



You might worry about a loss of fidelity or increase in noise resulting from copying another generation down. This is a valid concern if you're using an analog recorder, and digital users agonize over running the signal through another Analog/Digital conversion, but you'll just have to decide using your own ears whether it's a real problem. The degradation that you think you can hear if you really concentrate on one track gets buried when you have 23 other tracks covering up the little blemish. Consider which is the bigger problem—the sonic loss in copying or the difficulty in fitting the track to the mix.

- And last, most terrifying, but most powerful and effective: edit between sections of your mix. This is particularly effective when you've integrated a computer audio workstation with your mixer.

If you've been wildly editing mixes for years and years, you know what we're talking about. If not, learn to do it. Whether you do it digitally or you use a razor blade, you can fix that tiny detail in an otherwise perfect mix; you can mix a complicated track in sections rather than like a marathon; you can go from 200 instruments to a single whispered vocal and back again in a heartbeat; you can even fix a mix weeks later without losing the original magic —you just remix the one chorus that needs fixing and cut it in.

Section 3: Tips

Section 3 Table of Contents

About This Section

If you've already read through the first sections of this manual you should have a thorough understanding of signal flow through a mixer, the inputs, outputs, and controls, and have a good sense of how to connect and use your mixer for various applications. In this section, we're addressing several topics that aren't specifically about mixing or mixers, but about which we think you'd like to learn more. Having a better understanding of these subjects will make you a more creative and knowledgeable engineer.

Chapter 28: The Ins and Outs of Gozintas and Gozoutas

Think of a mixer as the heart of your system. Just about every device in your studio or sound reinforcement system is connected to the mixer at some point, many starting and ending there. Plugs, jacks, inputs, and outputs come in all different shapes and sizes, and it's important that all the devices that you interconnect play together nicely.

Three things of concern (or not—you'll find out more later) when interfacing analog electronics are impedance, operating level, and topology (balanced or unbalanced). Some of these tend to go hand in hand, but there's no reason for that other than convention.

In this section, we'll explore these issues and fill in the blanks.

Quick Summary— Balanced to Unbalanced Connections

In most systems, there'll be a mix of balanced and unbalanced inputs and outputs among the various pieces of equipment. This usually will not present problems when making connections if you understand how inputs and outputs are wired.

A *balanced* input or output consists of two signal terminals, called *high* (or *hot*) and *low* (or *cold*). It's the voltage between those two terminals that does the work.

There is almost always a shield over the signal-carrying conductors of the cable which carries no signal current, and which is usually connected to the chassis (ground) of the device. Examples are an XLR (microphone) connector or TRS (tip-ring-sleeve) jack.

An *unbalanced* input or output also consists of two leads, but what makes it unbalanced is that one of those terminals is always at ground potential (*and impedance*). The other lead carries the signal, and the signal voltage is between the signal (high or hot) terminal and ground. Examples are TS (tip-sleeve or "mono") plugs, or RCA jacks.

When connecting a balanced output to an unbalanced input, be sure the high signal

connections at each end are wired to corresponding plug terminals, and that the balanced low signal is wired to the ground terminal of the unbalanced input. In most cases, the balanced output ground lead will also be connected to the unbalanced input's ground through the cable shield.

When connecting an unbalanced output to a balanced input, be sure that the high terminals on the source and destination ends are connected together. The ground terminal on the unbalanced end should be connected to *both* the low signal and the ground lead from the balanced input.

If you have hum or RF interference problems, you may have created a ground loop between the two units. In some cases this can be fixed by disconnecting the connection between the cable shield and ground at one end of the cable or the other.

Sometimes you will have to make up special adapters to interconnect your equipment. For example, you may need a balanced XLR female connected to an unbalanced 1/4" TS phone plug. You can find some of them off-the-shelf, but there are so many possibilities that you may not find exactly what you need at the store. Learning to solder is an extremely valuable skill, but be careful not to burn your pickin' fingers.

Special Mackie Connections

We anticipated that you'd be combining a mixture of balanced and unbalanced equipment, so most connections on Mackie mixers have been designed with this in mind. With exception of the TAPE OUTPUT and TAPE INPUT RCA jacks, the channel INSERTS, and several jacks on the 8-Bus series, all Mackie inputs and outputs are balanced, but are designed so that they'll accommodate an unbalanced connection simply by plugging in the appropriate type of plug. Plugging an unbalanced 1/4" TS plug into a 1/4" TRS balanced mixer output will automatically unbalance the output and make all the right connections.

On the 8-Bus Series all of the 1/4" jacks, whether carrying balanced or unbalanced signals, are actually TRS jacks wired so that a 1/4" TRS plug inserted into an unbalanced input will automatically tie the ring (low) terminal of the plug to ground, assuring that a balanced output is correctly connected to the unbalanced input without requiring a special cable. This is a very useful feature when connecting professional units with balanced outputs to 8-Bus inputs. The wiring of off-the-shelf balanced 1/4"-to-XLR cables is standardized (not necessarily true for unbalanced versions) so you're safe and smart using an off-the-shelf cable for that mixed breed of connectors.

When You're Hot, You're Pin 2

"Is Pin 2 Hot?". Sometimes it matters, sometimes it doesn't, but people sure worry about it a lot.

What's Hot?

A voltage is always measured between two terminals, one of which is the reference for the other. An audio signal constantly alternates from positive to negative and back again with respect to that reference terminal. We call the reference the "low" or "cold" terminal, and we call the one that's changing relative to it "hot". Since we're only looking for a difference in voltage between two terminals, it's arbitrary which one we designate as "hot".

So, why the fuss? Two reasons: In certain wiring configurations, one terminal is always at zero voltage—it can never be "hot". Also, we want to be able to maintain (or at least be cognizant of) the actual polarity of the signal throughout our chain.

When the diaphragm of the microphone picking up the sound we want to reproduce moves inward as a response to increased air pressure, we want the loudspeaker at the other end of the chain to move in the direction which creates a similar increase in air pressure at our ears. (that's outward)

Pin 2?

History lesson coming up! The Pin 2 we're talking about is Pin 2 of the three-pin XLR-type connector commonly used on audio equipment. Back when every connection was balanced, on an XLR and fed from a

transformer, life was simple. Pin 1 was where you connected the cable shield. The signal was connected between pins 2 and 3.

Attention to details of design and wiring assured that when Pin 2 was positive with respect to Pin 3 at the input, the same would be true at the output. Any voltage measured between Pin 1 and either Pins 2 or 3 was purely incidental, due to electrical leakage or capacitive coupling between the transformer windings and ground—neither side of the audio signal was ever intentionally connected to ground.

But then a devilish deception came along—an XLR connector that was actually wired to an unbalanced input or output. With an unbalanced connection, the voltage reference point is the equipment ground. To this end, one of the two signal pins is connected, along with the cable shield, to Pin 1. This causes the signal voltage to appear between pins 2 and 3 just as it would on a balanced XLR connector, however there is also signal voltage appearing between one of those two pins and Pin 1.

Which signal pin? The one that's not tied to Pin 1, of course. Now, suddenly, we have a significance to one signal pin being "hot". Ampex usually gets the blame (or the credit) for initially establishing an industry standard of Pin 3 hot. Many of their units were supplied unbalanced, with an optional plug-in transformer available to make it balanced. It was their decision, based on a certain amount of logic, to make Pin 3 hot.

But you know what they say about standards—gotta love 'em because we have so many. Since this wasn't really a published standard, other companies, notably those in Europe, wired their unbalanced products with Pin 2 hot with respect to ground rather than Pin 3.

Enter Mr. Mike

The point where most signals enter the recording chain is the microphone. Fairly early on, American microphone manufacturers adopted a wiring convention: Positive pressure on the microphone diaphragm caused the voltage on the mic's Pin 2 to go positive with respect to Pin 3.

While there were some disagreement (or simply careless assembly) early on, eventually, microphone manufacturers worldwide settled on the XLR Pin 2 positive convention. Balanced or unbalanced, if a system is

wired correctly, sound which pushes air against a mic diaphragm will cause the speakers to push air against the listener.

The Balanced and Unbalanced Mix

Where the Pin 2 Hot issue becomes significant today is when interconnecting balanced and unbalanced equipment. While we usually expect an XLR connector to be balanced (some old broadcast equipment notwithstanding), the 1/4" phone jack in common use on audio equipment today is available with either one or two signal conductors, so it can accommodate either balanced or unbalanced connections. The terminals on a 1/4" plug are called the Tip, Ring (only on balanced plugs), and Sleeve for reasons which are pretty obvious when you look at it.

In an unbalanced configuration, the tip contact is hot and the sleeve acts as both the low signal terminal and shield connection. In a balanced configuration, the tip and ring of the jack carry the signal, with the sleeve being used to connect the shield.

When an unbalanced plug is inserted into a balanced jack, the jack's ring contact touches the plug's sleeve, which electrically connects the ring (usually the low signal) terminal of the jack to ground. This is the equivalent of tying Pin 1 of an XLR connector to one of the signal pins. Which signal pin? Another standard to the rescue!

AES14-1992 established today's wiring convention of Pin 2 Hot. Since the tip of a 1/4" jack is always hot, it's the equivalent to Pin 2 of the XLR, with the ring (if it's a TRS) being the equivalent of Pin 3. Most modern equipment adheres to this convention, making it straightforward to interconnect balanced and unbalanced equipment without losing the signal or inverting polarity.

Why the Fuss About Plus?

Most engineers, having given little thought to the issue of absolute audio polarity, don't expect that it makes any difference to the listener. Most of us have heard the hollow out-of-phase sound which results when the leads on one loudspeaker are swapped, but few people worry about which wire on the speaker goes to which terminal on the amplifier as long as both speakers are connected the same.

Consider the sound of a kick drum as heard by the drummer and by the audience. They're each listening on opposite sides of the drum head. The audience hears the beater attack as air that is initially compressed (air pressure increasing momentarily), while the drummer hears the attack as air that is initially rarefied (decreasing air pressure). Our ears perceive these two different changes in air pressure differently. The same effect is true for anything that makes a sound, but the reason (and the effect) is most apparent for a simple vibrating membrane like a drum head.

It's easy to blame non-linear components for this phenomenon, but we can't easily replace the suspension mechanism of our eardrums or swap out op-amps in our brain. If we want our recording to sound to the listener the same as it sounded in the studio, we must preserve polarity, at least to the extent that we can.

Since you can't test every home playback system and rewire it if necessary, there will always be someone who will hear your recording with incorrect polarity, but as a responsible engineer, it doesn't hurt to get it right when you record it. It's not difficult to do, either. Check your mics and cables and fix any that are miswired. Check for polarity inversions in your equipment and for correct wiring when going between balanced and unbalanced connectors. Awareness is the biggest step. You don't have to worry about your Mackie—we took care to get it right, just make sure your cables are wired correctly.

Balanced Connections

It's well known that balanced connections are superior to unbalanced connections, all other things being equal. They provide better rejection of stray noise, (*common mode* rejection), they hum less, and they help prevent tooth decay.

The reason why a balanced connection is quieter is because the *input* is differential. The input stage amplifies voltage *difference* between the two input terminals. If the same voltage relative to ground is applied to both terminals simultaneously, as would be the case with a stray radio signal or hum from a power transformer, the difference between the two identical voltages is zero—the interference gets cancelled out.

Textbooks define a balanced connection as one with a source that has the same *impedance* between each of the differential inputs and ground. It turns out that this is a sufficient (and necessary) condition for a balanced input to reject noise that's picked up by both conductors connected to its input. So, it takes two to tango. You need both a balanced output and a differential input in order to have any of the advantages of a balanced connection.

Balancing Methods

The original balancing device for both outputs and inputs is a transformer. Many people believe this is still the best way to do it, even though high quality, low distortion transformers are expensive.

Transformerless balanced outputs and inputs have become popular because of their lower cost and physical weight. With good design, they can perform very well. All inputs and outputs on Mackie mixers are transformerless.

Differential Outputs

The MAIN outputs on most Mackie mixers are differential—during the portion of the cycle when Pin 2 goes positive with respect to ground, Pin 3 goes negative with respect to ground by exactly the same amount. The impedance to ground of both output terminals is identical, so we've fulfilled our requirement for a balanced source. In addition, the differential output can provide twice the maximum voltage of a similarly designed single-ended output—Twice as hot or twice the headroom.

Impedance Balanced Outputs

Most other outputs on Mackie mixers (those designated BALANCED/UNBALANCED) are *impedance balanced*, but not differential. In this configuration, only the tip of the connector carries the signal voltage. The ring is always at zero (ground) potential, but rather than being connected directly to ground (which would make it a Plain Jane unbalanced output), it's connected to ground through a resistor that's the same value as the impedance of the hot output. The ring terminal, therefore, presents the same impedance *when looking back into the output* as the amplifier connected to the tip terminal.

Most balanced inputs will think that they're being driven by a balanced source, and will happily reject common mode noise. Since there's no active circuitry driving the low side of the balanced output, you don't have to worry about shorting out anything out when connecting it to an unbalanced input.

Impedance balancing isn't unique to Mackie—it's very common these days. Most sound cards with balanced outputs are built this way, as well as many other mixers and signal processors, and even microphones.



Not all balanced inputs are designed alike. Certain input circuit designs *require* a true differential source in order to achieve lowest distortion and maximum headroom. With all the cool gear out there, you might occasionally run across one of these inputs. It'll appear that your mixer output isn't "hot enough" to drive the unit. This isn't a fault of the mixer, it's just an incompatible design. The easiest fix is to put a transformer between the mixer's output and the unit's input.

Impedance—To Match or Not

Impedance is the resistance to the passage of an alternating current. It's abbreviated with the letter Z, and the practical unit of measure is the ohm, just like dc resistance.

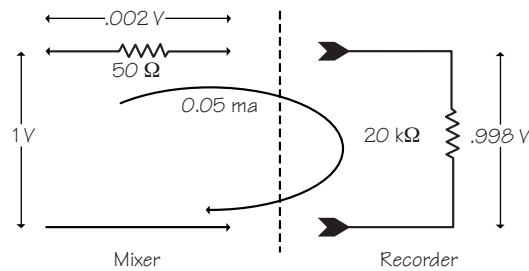
Ever notice, particularly in an older house, how the lights dim momentarily when the refrigerator or air conditioner switches on? If the wiring is old and thin enough, the lights stay a little dimmer all the time the refrigerator is running.

On a much smaller scale, the same thing happens when you interconnect any two pieces of audio equipment. No, the house lights won't dim when you connect your DAT to your mixer, but if you were to measure the output level of the console with a very sensitive voltmeter before and after you connected the recorder, you'd see a very small voltage change.

Unmatched Impedances

A simple but effective model of an output is a voltage source with a resistor in series with it. This could represent the output of a mixer, a signal processor, a recorder, or even a hydroelectric power generation plant. An

input can be modeled simply as a resistor connected between its input terminals. Undignified as it seems to represent your expensive recorder by a single resistor, it allows us to complete our simple circuit model:



Most modern audio equipment makes liberal use of operational amplifiers which, in the most common output applications, appear as a voltage source with a series resistance of around 50 ohms.

The input impedance of an opamp circuit in most common audio applications is in the ballpark of about 20,000 ohms (20 kΩ).

Because we have a complete circuit here, the same current flows through both the 50 Ω source and the 20KΩ load. Applying Ohm's Law,

$$I = \frac{1V}{(50\Omega + 20000\Omega)}$$

we see that this current is about 0.00005 amperes (0.05 mA)—not much current, but we're just pushing electrons around, not loudspeaker voice coils, so the mixer doesn't have to do much work.

Now, juggling the terms of the Ohm's Law equation and solving for voltage ($V=I \times R$), we see that our current causes a voltage of 0.998 volts to appear across our 20KΩ load, while the remaining .002 volts appears across our 50 Ω source. We've split the voltage between the source and the load, but managed to get nearly all of the voltage out of the mixer and into the recorder where we want it.

Older equipment worked by transferring power (voltage times current) from one unit to the next rather than voltage. 600 ohm inputs and outputs were common, and the outputs had enough power (real tubes, not tiny low power ICs) behind them to drive a 600-ohm load without breaking a sweat.

Modern studio gear is designed to work on the principle of maximum voltage transfer. This works just as well unless you're driving very long cables (thousands of feet) and saves cost, weight, and power. For maximum voltage transfer, so you *don't* want to match the source and load impedances. You want source (output) impedances to be low and load (input) impedances to be high. A good rule of thumb is about a 50:1 ratio minimum.

Since most equipment today is designed so that you'll have a much higher ratio than that, you can easily "mult" an output to several inputs without overloading the source. This is why you can double- or triple-bus subgroup outputs to two or three recorder inputs, or split a recorder output to two mixer input channels so you can set different equalization and effects in different parts of the song.

Matching Impedances

One instance where it's important to match source and load impedances is when connecting a loudspeaker to a power amplifier. Since a loudspeaker is a mechanical device and needs to do some hard work, (pushing air molecules around) it needs to receive high current as well as voltage from the amplifier.

Power is defined as voltage multiplied by current. Say you have an amplifier designed to work with an 8 ohm speaker. Juggle the numbers and you'll see what happens to the power when you vary the load impedance from above to below 8 ohms. You'll find that the speaker receives the maximum *power* from the amplifier when the source and load impedances are equal. We need less than a milliwatt of power to drive a recorder or a compressor, but we need several hundred watts to rattle the walls in the club and get the vocals loud enough to be heard over the drums.

The source impedance of a modern solid state power amplifier is typically considerably lower than the speaker for which it's rated, so in reality today's power amplifiers "voltage transfer" to speakers rather than "power transfer". The lower the source impedance, for a given amount of current through the speaker, the less power will be lost across the amplifier's output impedance, the higher the efficiency, and the cooler the amp will run.

The reason an amplifier which may have an actual output impedance of a few tenths of an ohm is rated to work with an 8Ω speaker is because the amplifier can only supply a limited amount of *current*—that amount of current which, through the rated load impedance, will equal the amplifier's rated power. A lower impedance speaker will try to draw too much current from the amplifier. Doing that for too long will release the magic smoke from the amplifier chassis and it will cease to operate.

So, when we connect an 8 ohm speaker to a modern amplifier's 8 ohm output, we're not really matching impedances, we're making sure that the amplifier remains within it's rated parameters.

Operating Levels

There are two common operating line levels, -10 dBV and +4 dBu. There are historical reasons for those numbers.

Back when Greg Mackie was still playing in the sandbox and hadn't graduated to the electronics laboratory, the Telephone Company was connecting all of their equipment using the "maximum power transfer" principle, with all input and output impedances being 600Ω. They decided on a standard power level of 1 milliwatt and called it 0 dBm—no dB difference from a milliwatt.

Then along came someone with a voltmeter and said "Hey, at 0 dBm, there's 0.775 volts coming out of this 600Ω output." Since this guy was talking about volts, he got out his slide rule and said ". . . hmmm . . . the ratio of .775 volts to 1 volt, log of that, times 20 is 2.2 dB.

$$dB(volts) = 20 \log \frac{Voltage(measured)}{Voltage(reference)}$$

If dBm is milliwatts, then I guess we'll have to call this voltage 2.2 dBV. Darn, I wanted a nice round number, too."

To get that round number, he defined 0 dBu as the *voltage* required to pump 1 milliwatt of power into a 600Ω load. So far so good.

Then someone came along and invented the VU meter to indicate the program level being sent to a broadcast transmitter. The best meters they could make back then weren't sensitive enough to move the pointer very far up scale with only 0.775

volts going into them, so they just cranked up the level until the meter reading looked .. well . . normal. It happens that this required boosting the 0 dBu signal level by 4 dB, and the standard operating level of +4 dBu was born.

This was back in the days when everything had tubes and transformers, and that combination was capable of putting out many more volts than the low power integrated circuits that are the heart of most of today's equipment. Professional studio equipment has always been able to comply with the +4 dBu standard operating level with plenty of headroom over that by hook or crook and a few power transistors at the output stage, but that was expensive.

The musical instrument industry never really had to deal with professional studio levels in the early days, so they set their own standards to match lower cost circuitry which was happier working at lower signal levels. They picked 1 volt as their nice round number and made their standard operating level 10 dB below one volt, hence the "consumer" or *semi-pro* operating level of -10 dBV.

Today it's pretty easy to find gear designed to work at +4 dBu, but -10 dBV designs are still common, so it's important to understand the significance of the different levels. You need to take care of business if you're trying to connect two pieces of equipment with different operating levels..

If you connect a +4 dBu recorder output to a -10 dBV console input, you'll run the risk of overdriving the input and introducing distortion. At best you'll wonder why you have to run your console faders or trim controls down at the bottom of their range to keep the console meters from hitting the pin. "My input is too hot" is the common expression of this interfacing problem.

Conversely, if you connect a -10 dBV mixer output to a +4 dBu amplifier input, you'll get a fairly low level out of the amplifier unless you push the console to near its maximum output level. This is where you say "My mixer isn't hot enough."

There are converters which adapt between operating levels, and sometimes purchasing one is the best solution to matching up your equipment.

Interrelationships

There's no reason for impedance, level, and balanced/unbalanced characteristics to be related other than by manufacturing cost, but that's really important, so a number of de facto relationships have emerged.

As a rule, -10 dBV inputs and outputs are unbalanced. They used to have higher output impedances, but today that's rarely the case. As a rule, +4 dBu inputs and outputs tend to be balanced. This used to be one of the things that distinguished pro and semi-pro gear.

Today, that distinction is blurred, but the folklore of "-10 unbalanced, +4 balanced" remains, and so, still, does the requirement for most systems to be able to accommodate both operating levels.

Chapter 29: Equalization Tips

Everybody knows you're supposed to EQ (otherwise why would we put all those knobs there), but where to start? And what do the knobs really do?

Simply, this is what an equalizer does—it takes our carefully designed ruler flat frequency response and mucks it up, boosting a frequency range here, lowering one there, all in the interest of making what comes out sound more musical than what went in, if not subjectively more accurate.

Initially, equalization in the studio was used to correct deficiencies. A loudspeaker or microphone weak in high or low frequency response could be made more “flat” by the use of an equalizer. It's only been in the last quarter century of multitrack recording that we've learned to use EQ as a creative tool, not just a corrective one.

EQ Function Follows Form

Equalizers come in many forms and functions—the EQ built in to your console, stand-alone equalizers that you can plug into INSERT jacks or between the console output and a monitor or front-of-house speaker system, the low-cut filter on your microphone, single-band, multi-band, graphic, parametric, active, passive, analog, and digital EQ. Even the bass and treble tone controls on the living room stereo or boombox are equalizers.

The operation of an equalizer is always described by its frequency response, sometimes in words, sometimes in numbers, but most often with a graph of amplitude versus frequency. The frequency response plot of an equalizer at work is anything but a straight line. Let's look at the fundamental building blocks that comprise an equalizer in terms of their frequency response curves.

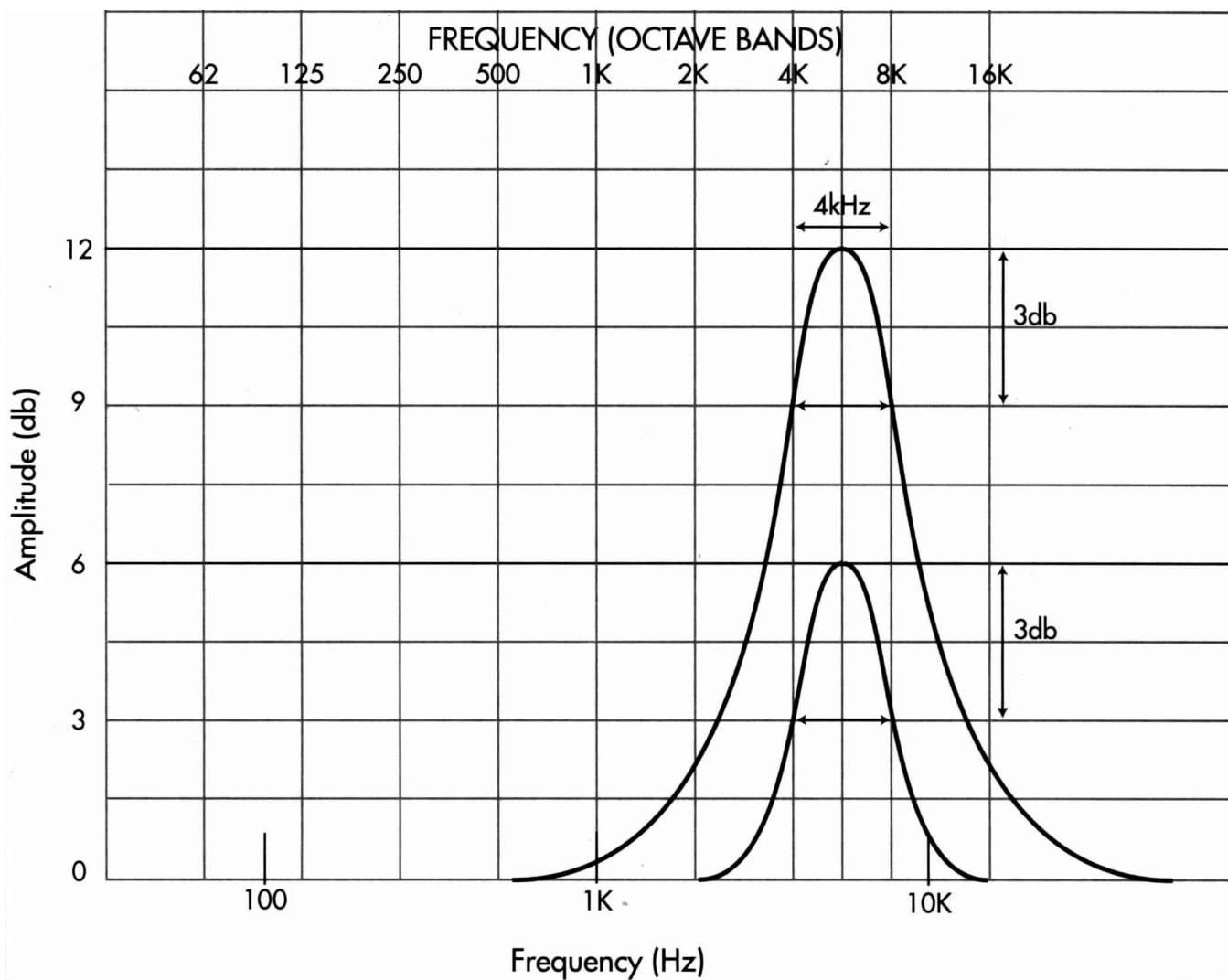
Logs and Octaves

Frequency response is normally plotted with frequency on the horizontal axis, using a logarithmic scale. Each decade (10 Hz, 100 Hz, 1 kHz, 10 kHz, etc.) is represented by a major division of the graph, with the intermediate frequency divisions scrunched closer together as they approach the next major division. This lets us represent a wide range of numbers while maintaining good

resolution in areas that relate to how we hear.

The vertical axis of a frequency response curve is normally plotted on a linear scale of decibels (dB) which are in themselves logarithmic, relating to the way we hear loudness.

Interestingly, when plotted on a logarithmic scale, frequency intervals one octave apart are equally spaced. A book on the mathematics of music theory will tell you why. This is handy because we often describe the working range, or bandwidth of an equalizer in terms of octaves.



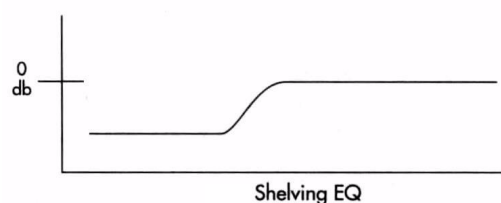
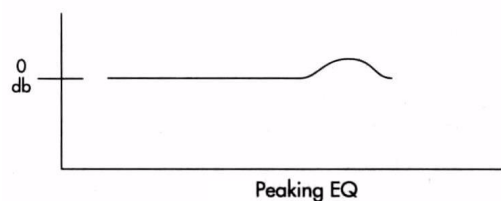
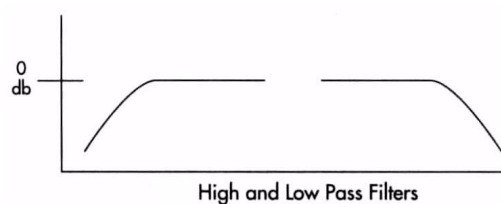
Compare the logarithmic divisions (100 Hz, 1 kHz, and 10 kHz) labeled at the bottom of the graph, with the linear scale along the top divided into intervals of one octave.

When looking at spec sheets, you'll usually see EQ curves plotted with the horizontal axis divided logarithmically, with 1 kHz in the middle of the axis. We'll use 1 kHz as our reference point for octaves, doubling the frequency each division above 1 kHz and halving it at each octave division below, rounding off decimals for sanity's sake.

Building Blocks and Dangerous Curves

The basic building block of an equalizer is a filter. A high pass filter (also called a low-cut filter) passes frequencies higher than its design frequency and attenuates frequencies which are below it. Similarly, a low pass (high-cut) filter allows frequencies below its design frequency to pass through unaltered while reducing the amplitude of frequencies above its design frequency.

A resonant filter puts a bell-shaped hump in the frequency response, flattening out on either side of its resonant frequency. A filter with a "shelving" characteristic only goes so far down along the attenuation slope and then the curve flattens out, allowing adjustment of a range of frequencies equally.



Because this is the real world, the frequency response of a filter doesn't just drop off like the face of a cliff. Its attenuation changes

with frequency at a rate determined by the filter's design. A simple filter design has a slope of 6 dB per octave, meaning that a signal one octave above a low pass filter's design frequency will be attenuated by 6 dB (one-half), a frequency two octaves above will be attenuated by 12 dB, and so on, until the signal finally gets lost in the noise.

Filters, by nature, are lossy—they only *reduce* signal's amplitude. But by combining a filter with an amplifier, it's also possible to boost a range of frequencies. With the exception of a simple low- or high-cut filter (generally an in-or-out switch like those at the top of the channel strip on several Mackie mixers), an equalizer can be adjusted so that, relative to "flat", the frequency response curve can be pulled either upward or downward. This is starting to look like a pretty useful gadget.

Knobs, Dials, and Switches

The various flavors of equalizer differ in the degree of control they offer in shaping the frequency response. The most basic EQ consists of low and high frequency EQ knobs. Their operating frequencies are usually fixed, typically around 120 Hz for the low EQ, around 8 kHz for the high. Those frequencies aren't magic, but tend to be useful when working with music—putting more "whomp" into a kick drum, removing rumble from a bass or guitar, or adding brightness to a guitar. Both bell-shaped (peaking) and shelving characteristics are found in this application.

Sometimes the operating frequencies of high and low EQ controls can be selected by a switch to increase its flexibility. Choosing whether the EQ is centered around 100 Hz or 250 Hz can be handy to bring a particular bass instrument out in a mix, while a choice between, say, 6 kHz or 12 kHz on the high end lets you add articulation to a vocal or shimmer to a cymbal.

As high and low EQ frequencies shift toward the center of the audible frequency range, the difference in sound between "peaking" and "shelving" shape becomes more apparent, and there are good applications for both curve shapes. Some low and high equalizers allow you choose between a peaking and shelving characteristic for greater flexibility.

Mid-Range EQ

Audio shaping is frequently needed in places other than at the ends of the spectrum, so it's common to include controls for tweaking mid-band frequencies. The fancier the EQ, the more controls you have. All Mackie mixers have at least one EQ control in the 2–4 kHz range.

Sweepable EQ

While basic, fixed frequency EQ is useful for overall trimming, when using an equalizer as a sound shaping tool, you'll quickly discover that the fixed frequencies are never quite in the right place. The *sweepable*, or "semi-parametric" equalizer adds another dimension of control. Here, rather than a fixed (or switch-selectable) frequency, a knob lets us tune the equalizer's center frequency over a fairly wide range to put the action precisely where it's needed. The hip term for this is "dialing-in the EQ".

Graphic EQ

A specialized form of fixed frequency EQ is the graphic equalizer. It consists of a large number of peaking filters with their center frequencies spaced one (or 1/2 or 1/3) octave apart, with the frequencies spanning the full audio range of about ten octaves. The boost or cut in each band is adjusted using a vertical slider, giving a graph-like representation the frequency response. The PPM and CFX series mixers are equipped with graphic equalizers, and many of the mixer hookup diagrams show an external graphic equalizer following mixer outputs that feed the loudspeakers.

Graphic equalizers aren't terribly common in recording applications, but they're frequently the EQ of choice for overall system tuning in PA applications where they're used to compensate for boomy room acoustics or to eliminate feedback by cutting gain at the howling frequency. Skilled live sound engineers have the uncanny ability to hear a squeak and immediately grab the right graphic EQ slider, even in the dark.

Parametric EQ and Parameters

Frequency, bandwidth, and amplitude of the boost or cut are the parameters which describe the curve of an equalizer. With a fully parametric equalizer such as found on the middle EQ section of the 8-Bus con-

sole, you can control all of them. We've been talking about these parameters in general terms, but let's take a close look at them.

Bandwidth is defined as the number of kilohertz between the points along the frequency response curve where the amplitude changes by 3 dB from the value at the top of the peak or trough of the dip. Take a look back at the graph showing the two peaks at 6 kHz. Notice that for both the 6 and 12 dB boost curves, the amplitude is 3 dB down from the peak at 4 kHz and also at 8 kHz. Since those two frequencies are an octave apart, we can say that this equalizer has a one octave bandwidth, independent of the amount of boost.

Sometimes "Q" is used to specify an equalizer's bandwidth. It's a much hipper sounding term than octaves, though the term is actually more useful to a designer than a musician.

Bandwidth or Q

Q is defined as the filter's center frequency divided by its 3 dB bandwidth. It's just a number, with no units. For the peak equalizer shown on the graph above, we have:

$$Q = \frac{6\text{kHz}(\text{center})}{8\text{kHz} - 4\text{kHz}} = 1.5$$

Where 4 and 8 kHz are edge-of-bandwidth points.

The bandwidth control on a parametric equalizer is often marked in octaves (usually something less than 0.5 to 3 or more) as it is on the 8-Bus, or it may be marked in Q, with a typical range of 2 to 10.

Notice that Q is independent of the amount of boost or cut, so a large amounts of boost affects frequencies beyond the center frequency to a greater extent. This is apparent if you compare the 6 dB and 12 dB curves boosts shown in the graph.

Higher values of Q mean a narrow bandwidth (steep slope) while lower values of Q give a represent a curve with a gentler slope. If you want 12 dB of boost at 6 kHz but don't want more than, say, 3 dB boost at 4 kHz, you must increase the Q of the equalizer.

A large amount of cut with a Q of 10 or more is useful for notching out a narrow

band of frequencies like ac line frequency hum or air handler noise. A Q of 1.5 with a small amount of boost is a good for gentle correction of spectral balance.

While some equalizers boast of “Constant Q”, not many of them actually work that way. Equalizers generally sound more musical when the Q varies with the amount of boost or cut so that “out of band” frequencies are affected in proportion to the bandwidth.

A Little EQ Philosophy

There are two principal uses for EQ—corrective and creative, and there’s no “right” EQ for any particular instrument or voice.

Creative EQ

When using EQ in a creative fashion, you intentionally want to change the sound, sometimes radically, creating a new sound—a rhythm guitar with practically no mid-range, a voice that sounds like it’s been recorded over the telephone—you get the idea.

When trying to create a new sound, there are no rules and no principles. Just turn all the knobs until you either get the sound you’re looking for, find something else that sounds really cool and you decide that’s just the sound you really wanted, or discover that there just isn’t enough to work with, either with the source or the controls provided by your equalizer.

The main thing to remember is that while you can take away or emphasize frequencies present in the original source, an equalizer can’t add frequencies that didn’t come into its input. (Well, it will if it’s driven into distortion, and that’s indeed a legitimate creative use.) If you want the bass track to sound like it was played on a flute, it ain’t gonna happen.

Corrective EQ

The proper starting point for a mix is with no EQ. Don’t begin mixing by reaching for the EQ knobs, at least not until you’ve developed a lot of listening and mixing skills. Don’t try to fix something until you know it’s broken.

How do you know it’s broken? When you can’t get a good mix just by balancing levels

and panning. Then, and only then is the time to start twiddling the EQ knobs.

A common piece of advice when EQ-ing is to first try cutting rather than boosting. If the basic tracks are well recorded, most mix problems are the result of two or more sounds trying to occupy the same portion of the spectrum.

This is common with modern multitrack recording techniques, where every instrument is recorded up close and personal, using the full bandwidth of the mic and the recording system. In real life, high and low frequencies drop off at different rates depending on how the instrument’s sound radiates, and we don’t usually encounter a situation where we hear every instrument in the ensemble right in our face.

Low-frequencies (in the 150-250 Hz range) tends to add up in the mix very quickly and it’s tempting to boost another frequency range to balance it. Rather than pile more highs into the mix, it’s better to cut some lows out of instruments that, even though they may sound great on their own, are muddying up the mix.

While it’s rarely a good idea to start out mixing by setting EQ knobs to where you think they should be without listening first, if all of your instrumental tracks have been close-miced and are likely to have excess bass, you might start out a mix with a couple of dB scooped out of this range on everything but the bass and kick drum, then add some back in judiciously to add some warmth.

Sometimes, corrective surgery is called for when an instrument booms at a certain note or a particular note is there, but it’s weak. Tuning in on that note and adjusting it with a narrow band parametric equalizer maintains much more of the instrument’s characteristic than simply limiting or compressing the track to even it out.

It’s very helpful to know the range of instruments in your mix (charts can be found in just about any recording reference book), but it’s also important to know understand the overtone structure affects the timbre of an instrument.

By dropping the fundamental frequency range of a lead instrument by a few dB but leaving its most important overtones unchanged, it will still sound like the same instrument in a mix, but by cutting a little, you’ve made room for another instrument

that was fighting for the same chunk of the spectrum. Generally an equalizer with a bandwidth of around an octave is appropriate for this sort of work. It's fairly common to use a high pass filter to clear away most of a guitar below 80 Hz or so to leave room for the bass and kick drum.

Warming Up a Track

In addition to making a mix work, EQ is sometimes used to modify the tone of an instrument or voice to add one of those obtuse characteristics such as “warmth” (around 250 Hz) or “air” (around 15 kHz).

Practice Safe EQ

Successful equalization is all about understanding what you're hearing. A skill that will make you a better engineer is the ability to recognize weakness or overemphasis in certain frequency ranges so you'll know which knobs to reach for, or if EQ is required at all. Sharpen your listening skills by running well mixed music through an equalizer and learn what happens when you boost or cut different frequency ranges. And keep your ears clean.

Here are a couple of charts to get you started:

Table I: Key Frequencies for Instruments

Instrument	Principal Frequencies
Electric Bass	700 Hz - 1 kHz = attack or “pluck” 60 - 80 Hz = “bottom” 2.5 kHz = string noise and slap
Kick Drum	60 - 80 Hz = “boom” (be careful of conflict with bass) 1.5 - 2.5 kHz = beater “smack”
Snare Drum	80 - 100 Hz = fullness 200 - 300 Hz = fat 1 - 2.5 kHz = crispness
Cymbals, Hi-hat	200 - 400 Hz = gong sound 7.5 - 10 kHz = “shimmer”
Rack Toms	200 - 300 Hz = fullness 5 kHz = attack
Floor Tom	80 - 120 Hz = fullness 5 kHz = attack
Conga	200 - 400 Hz = resonant ring 5 kHz = presence, slap
Electric Guitar	250 Hz = “body” (too much is “muddy”) 2.5 kHz = clarity
Acoustic Guitar	Same as electric, but there’s almost always too much 250 Hz
Piano	80 - 100 Hz = bass 2.5 - 5 kHz = “presence” 10 kHz = crispness narrow boost around 2.5 kHz = “honky tonk”
Brass, woodwinds	120 - 240 Hz = fullness 5 - 7.5 kHz = “shrill”
Voice	150 - 250 Hz = body 400 - 800 Hz = “nasal” 2 - 4 kHz = presence

Table 2: Characteristics of Frequency Ranges

Frequency Range	Produces This Desired Effect	When overused or when there's too much of this range
16 - 60 Hz	Sense of power, felt more than heard. <i>Caution! not reproduced on most small monitors.</i>	Makes music sound murky
60 - 250 Hz	Emphasizes fundamental tones of the rhythm section, changes musical balance making it fat or thin	Makes music boomy or thin
250 Hz - 2 kHz	Low order overtones of most instruments fall in this range. Changes an instrument's timbre, helps to separate instruments with similar fundamental ranges	500 Hz - 1 kHz creates "telephone effect", 1 - 2 kHz sounds "honky". Makes music tiring to listen to.
2 - 4 kHz	Vocal presence	Makes vocals "lispy", loses intelligibility of consonants (<i>merry, very, berry</i> will sound similar)
4 - 6 kHz	Clarity and definition of instruments, "up front" sound, small boost around 5 kHz makes a complete mix sound louder.	Vocal sibilance, listener fatigue
6 kHz and up	Brilliance	Sibilance, vocal harshness

Chapter 30: Using Compressors

Mackie doesn't make compressors other than the one built into the PPM mixers, but compression has become such an integral part of mixing that we thought if you needed to learn about it, you might as well learn about it from us.

Dynamic Range

To use a compressor effectively, it's important to understand the concept of *dynamic range*. Dynamic range is the difference between the loudest and quietest signal levels passing through the system chain. The span between barely audible and physical discomfort is about 120 dB, so we consider that to be the dynamic range of human hearing. Anything below the threshold of hearing will be lost, as will anything above the threshold of pain. But how much dynamic range do we need in our recordings, or, more importantly, how much can we really use?

Few people listen in a totally soundproofed room. A very well designed studio control room has an ambient noise level 10 to 15 dB above the threshold of hearing. Since we want to keep ourselves safe from hearing damage, 95 dB or so is about all the dynamic range we can use. But consider this: A very quiet living room has an ambient noise level 25-30 dB above the threshold of hearing. The inside of an automobile at reasonable driving speed is 60 dB higher.

Since most consumer audio systems aren't capable of producing painful sound pressure levels (with the exception of those cars that cruise by with the bass pumping loud enough to rattle the windows in your house), a typical listening environment can only support a dynamic range of 65 to 75 dB.

The electronics in our recording chain is much better than that, with digital recorders and analog or digital consoles capable of 105 to 115 dB of dynamic range. This means that the listening environment, not the gear, sets the practical limit on dynamic range.

Even in the best of control rooms, we have to squeeze as much as 115 dB of obtainable dynamic range into a 95 dB box. For the rest of the world, though, we have to

squeeze harder so soft passages don't get lost when your neighbor starts up his lawnmower, or when listening to the car radio at 65 miles per hour with the windows open. Therefore we can't record all the dynamic range that's available if we expect people to hear all the music we record.

What A Compressor Does

A compressor reduces dynamic range. When used properly, it's hard to detect the action of a good compressor, but a compressor can also be used creatively as a sound shaping tool. A compressor can be inserted into a single channel in the recording chain when recording or mixing a track, or compression can be applied to an entire mix or sub-mix. The hookup diagrams illustrate both applications.

Let's look at applying compression to a vocal track. Hard consonants such as the letter 'T' create a high initial sound level before settling down, whereas most vowels tend to be more even. The average volume level of a word may be fairly low, but because of an initial loud consonant, we can only raise the word's volume so far before running out of headroom. If there's music playing under the voice, even when boosting the vocal level as high as possible without distorting the attack, a word (or a syllable) may be far enough below the level of the music to become lost or misunderstood.

If we reduce the gain momentarily during the loud attack, then bring it back up when the distortion-risking blast is over, we can now boost the average level of the word so that it can be understood clearly over the music. What we're doing here is reducing the *dynamic range* of the word, the difference between the loudest and softest parts. By doing that, plus boosting the level, we've made the vocal track louder in the mix without reaching a volume at any point that will cause distortion.

Since to most listeners, louder equals better, another use for compression is to make a mix sound louder. Often there's a single sound—a snare drum for example—that's noticeably louder than anything else in the mix. A drummer hits the snare louder on some beats, and the loudest hit determines

the maximum level that can be recorded. By compressing the overall mix and sitting on the loudest hits, the average level of the song can be raised.

Basic Compressor Theory and Buzzwords

Threshold

We need to be able to set the compressor so that it will reduce the level of signals above a certain volume level and not affect lower level signals. The level at which the compressor starts working is called the *threshold*, and nearly all compressors have a control for it. Those which don't have a fixed internal threshold where you set the level where compression begins by adjusting the input level to the compressor.

The Threshold control is generally calibrated in dB relative to the nominal operating level of the unit (typically +4 dBu or -10 dBV), though it's rarely a precise calibration. If you wanted the compressor to work on everything that's 6 dB below normal operating level, you'd set the Threshold control to -6 dB.

Compression Ratio

Below threshold, a compressor has a linear gain characteristic, just like a good amplifier or a piece of wire. Whatever goes in comes out unchanged. When the signal going into the compressor increases by 6 dB, as long as that increase doesn't put it over the threshold, the output also increases by 6 dB.

Once the signal goes over the threshold however, the compressor goes to work and starts reducing its gain, or more accurately, adds attenuation. If the compressor's output changes by only 3 dB when an above-threshold input changes by 6 dB, we call this a 2:1 *Compression Ratio*. A compression ratio of 5:1 means that a signal that's 10 dB above threshold will come out only 2 dB above threshold.

Gain Reduction

We can also say that this action represents 8 dB (10 minus 2) of *gain reduction*. When expressing of the amount of compression in this way, we take an "eyeball average" since the actual amount of gain reduction at any instant depends on the input level at that instant. When someone says "I compressed

vocals 2 to 3 dB", they mean he applied enough compression so that most of the over-threshold peaks were attenuated by 2 or 3 dB.

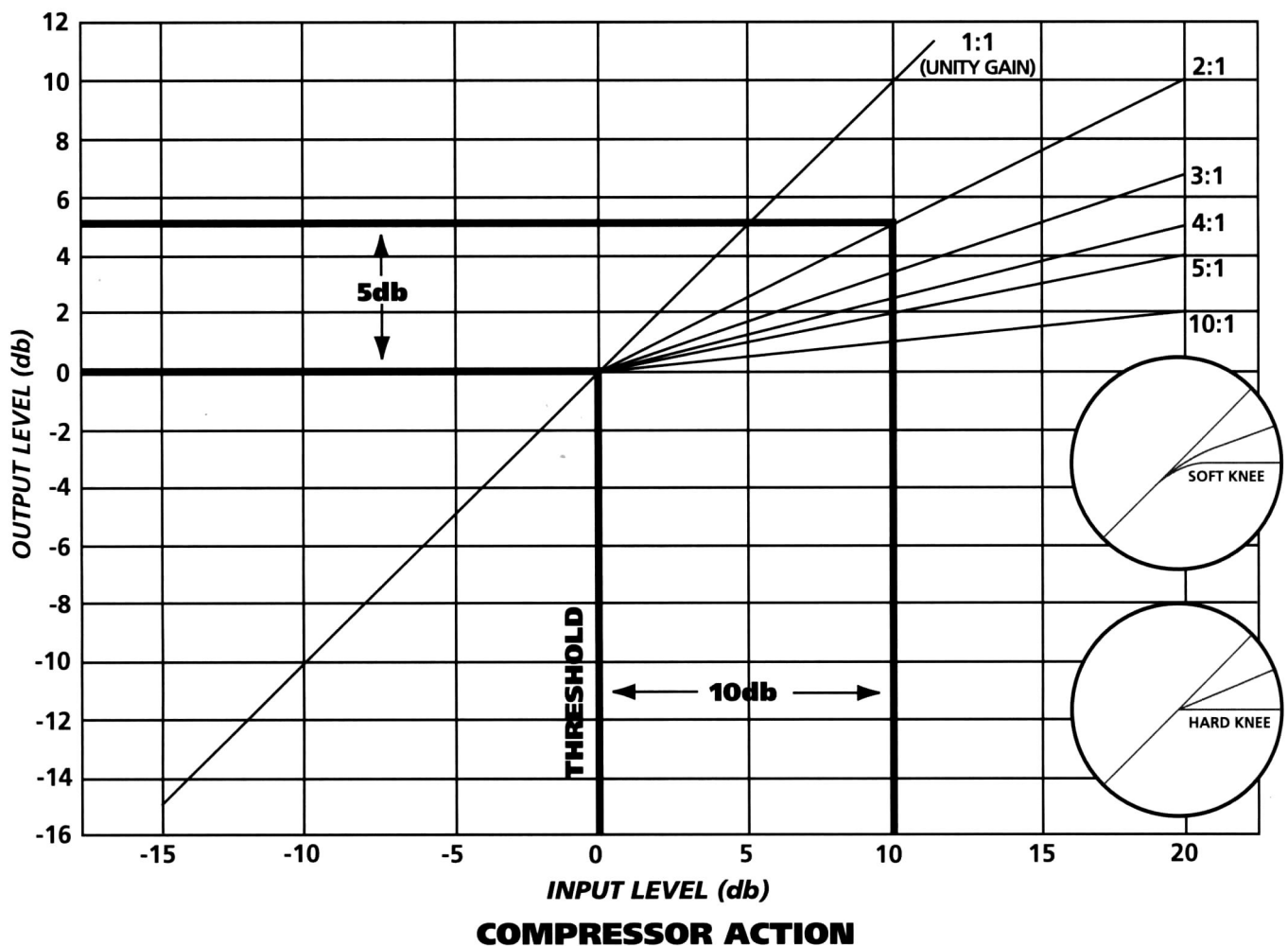
That's a typical amount of compression applied when tracking a singer with good dynamic control. It evens out sustained notes a bit and provides a small safety net against surprise overloads. A singer with less control may require 10-12 dB of gain reduction to level out his track.

Limiting

If we never want the output to exceed a pre-set level, the compression ratio must approach infinity (10:1 is usually close enough for rock and roll), so that a large over-threshold signal results in a very small change in output level. A compressor set to a very high ratio becomes similar in action to a limiter, as its output level is limited to the threshold level. Limiting the output of a PA system is a good way to protect your speakers and your audience's ears.

The graph below illustrates the action of a compressor for several ratios. The slope of the line represents the gain. Notice that for any compression ratio other than 1:1 (no compression) the line changes its slope at the threshold, which we've chosen to be 0 dB.

This change in slope is what makes a compressor compress. Below threshold, the gain is unity—the output change is equal to the input change. But above threshold, the gain is less than 1, which is actually attenuation. Look at the 2:1 line. Note that above threshold, we get only a 5 dB change in output for a 10 dB change in input—a ratio of 2 to 1.



The Compressor Knee

The point at which the slope of the gain line changes is called the *knee*. A compressor is said to have a “hard knee” characteristic when the slope changes abruptly at threshold. A “soft knee” characteristic is one in which the gain change is gradual over some range of input level. With a soft knee compressor, gain reduction actually begins a little below threshold and the full amount of gain reduction isn’t achieved until the input level is somewhere above threshold. A hard knee compressor tends to work better at catching transients while a soft knee tends to be less obtrusive on vocals. But these are only typical applications. Your voice or snare drum may vary.

Controlling the Gain

On the surface, a compressor is a fairly simple device. All the action takes place in the compressor’s gain control element with an amplifier on the front and back end to match up signal levels to the outside world.

There are several different devices used as the variable gain element, and to a large extent it’s the characteristics of these different devices that give each different compressor its “personality”.

The signal level must be detected so the gain reduction element knows how much gain to drop. Some compressors detect the RMS (average) level of the signal, others detect the peak, and still others detect both. In some compressors, the level at the input is detected and the gain control element decides what to do based on that. Another design measures the signal level at the output and feeds that information back to the gain control element. All of these things affect the sound of a compressor.

Today you can buy compressors with multiple personalities, where the gain element’s response curve shape with respect to time is selectable, thus emulating the action of different compressor designs.

Attack Time

An important compressor characteristic is *attack time*. Its definition is a bit loose. To some, it’s the amount of time it takes for

the compressor to reach full gain reduction when hit with an over-threshold signal, others define it as the time required to get most of the way (typically 67%) there.

Attack time makes a big difference in how a compressor affects the sound of the signal you're compressing. If the signal has a loud initial attack (like almost any drum), we may choose to allow the attack to get through unaffected even though it's louder than our desired average output level.

To do this, choose a compressor attack time setting that's longer than the instrument's attack time. On the other hand, if it's the transient that we want to squash, we need a fast attack (short time) so that gain reduction will begin as soon as possible after the input crosses the threshold.

Attack time is not the same as the shape of the compressor's knee, but their effects are related.

Release Time

Once the signal drops back below the threshold level, the compressor needs to return to unity gain, but this doesn't happen instantly. Instead, the compressor's gain rises gracefully (we hope) over some period of time. This period is called Release Time.

Some compressors have no adjustments for attack and release times. Either they're fixed by design or they're "program dependent" (automatic), which means the compressor decides how fast it should respond based on the envelope of the input signal. Each design has its place in the universe and you're not necessarily being cheated if your compressor is missing a knob or two.

Make-up Gain

Since a compressor reduces gain, it is usually necessary to amplify the signal following the gain reduction circuitry in order to get the output back to nominal operating level. Most compressors have an Output or Make-up Gain control which allows you to adjust the signal output level to match up with the next device in the signal chain.

The Gain Reduction Meter

Compressors usually have a meter which looks like a VU meter only it works backwards, indicating the amount of *gain reduc-*

tion rather than the signal level. A typical compressor's meter reads 0 dB whenever the input is below threshold and moves down scale as the input level goes above threshold and gain reduction starts.

Stereo or "Program" Compressors

A compressor is basically a single channel device, but there are stereo compressors, often called "program compressors" most often used to compress a full stereo mix.

The thing that differentiates a stereo compressor from simply patching one compressor into each channel is that the signal that controls the amount of gain reduction is shared by both channels.

Reducing the level of one channel of a stereo pair will cause the balance to shift to the louder side. We don't want the stereo image to wander around when one channel goes over threshold and the other one doesn't, so we connect the level detectors of the two compressors together. Now when either channel requires some gain reduction, that same amount of gain reduction is applied to both channels.

Compressor Artifacts

Two uncomplimentary terms often used to describe the sound of a compressor are *breathing* and *pumping*.

Breathing

Breathing is most noticeable on a solo voice and is often, in fact, the sound of the vocalist breathing. If release time is short, the gain rises quickly during pauses between words, just as the singer takes a breath. The increased gain makes the breath more audible.

Hearing a singer taking a breath may not always be desirable or dignified, but at least it's organic. Few recordings are made in an absolutely silent environment, however. Any ambient noise in the room will be boosted when the gain rises, creating an artificial "breathing" sound, perhaps carrying leakage from the singer's headphones along with it.

All compressors exhibit some breathing, but careful adjustment (which includes con-

trolling room acoustics and mic positioning) can minimize it.

Pumping

Pumping is another compressor artifact. It's most apparent when compressing an overall mix rather than a single track.

One instrument in the mix that's louder than the others will trigger the compressor into action. If that instrument stops playing, even for an instant, the level of the mix will increase noticeably. Each time the dominant instrument starts or stops, it "pumps" the level of the mix up and down.

Compressors that work best on full program material generally have very smooth attack and release curves and a slow release time to minimize the pumping effect.

Working the Knobs

If your signal has peaks up to +15 dB and you want to reduce those peaks to a more manageable +5 dB, you might set the threshold at -5 dB and compress using a gentle 2:1 ratio. Or if you want to use a stiffer ratio, say 6:1, you'd set the threshold at +3 dB. As an exercise, try plotting out a few combinations yourself. Then set up your compressor and listen to the differences.

Lowering the threshold while keeping the compression ratio fixed reduces the maximum output level since you're compressing over a larger portion of the range of the input signal. By keeping the threshold fixed but increasing the compression ratio, you'll reduce the output level by compressing only the loudest signals. There are no rules for this, let your ears be your guide, with the meters as a sanity check.

The Compressor as a Tone Modifier

Varying the attack and release times when compressing an instrument can change its timbre by rounding off a sharp attack or stretching out the sustain portion of the note's envelope. It's sort of like having an equalizer with a different parameter.

A drum hit can be "stretched out" by applying a long release time, a healthy gain boost, and fairly high compression ratio.

Compressing low frequency program material requires special care. The attack and decay portions of a kick drum are 60 to 80

milliseconds long, but a low pitched kick has a fundamental frequency of about 40 Hz. This means that only three of four cycles of the kick's fundamental are heard on each hit, much of that being in the decay portion of the envelope.

Compressing the drum with a fast attack time and high compression ratio will make more cycles of the attack portion of the drum audible, making its "thump" sound much more pronounced without having to boost low frequency gain with an equalizer. Try this next time you want a kick in the chest.

The beater attack is at a higher frequency (1 to 3 kHz) so a moderately fast attack will let a few cycles of beater through while working on the low "whump". Slowing down the attack lets more of the beater sound pass before being compressed, often allowing you to bring down the overall level of the kick in the mix without losing the impact.

Compressors with very fast attack time that often work well on vocals work poorly on a bass because the compressor actually tries to follow the individual cycles of the waveform rather than the envelope of the note. This characteristic can be used as a special effect, but usually it just takes all the life out of a bassy instrument.

The Bottom Line

A compressor can never be used by the book—not even this one. You need to listen carefully when you make adjustments. By understanding the effect of each of the adjustable parameters, you'll be better able to reach your goal quickly.

Chapter 3I: Gain Structure—Setting the System Levels

One of the most important things you can do to make an audio system sound good is to set up its gain structure properly. Conversely, improper gain setting throughout the system can really make it sound bad.

Gain Structure is the term we use for the collection of various gain adjustments throughout the system—the mic preamp, the fader, the main mix output level, the input gain of a power amplifier or recorder, and so on. Setting all of those gain/level controls to work together properly isn't difficult, but so often it gets ignored. You may run across the term *Gain Staging*. That's the process of setting gains throughout the system to achieve proper gain structure.

We hope that by now you've at least tattooed the **Level-Setting Procedure** on the back of your hand. Since that's where the sound enters your system, proper adjustment of the TRIM controls is a good start toward proper gain staging.

There are many ways of establishing optimum gain structure. Once you understand what you need to accomplish, gain staging will become second nature when you're faced with new equipment. To understand gain structure and gain staging, you need to understand a couple of closely related terms which we've sprinkled throughout this book—*dynamic range* and *headroom*.

Dynamic Range

Dynamic range is the ratio of the level of the loudest undistorted signal to that of the quietest audible signal, expressed in dB (decibels). Dynamic range can apply to a single piece of equipment or to a complete system.

For electronic equipment, the maximum output is ultimately determined by its power supply—If the power supply provides ± 15 volts to the integrated circuit opamps, the maximum possible peak-to-peak signal voltage is 30 volts. It will never be 31 volts—you can't generate what's not there, at least not without adding more components.

The noise floor limits the lowest audible signal we can use in the system. In general, the better the design, the lower the noise

floor. Reducing the noise floor increases the available dynamic range.

Typical professional audio equipment like your Mackie mixer can put out a maximum level of +22 to +28 dBu depending on the model and the output jack. Noise floor is a little harder to measure (there are several legitimate methods), but it's usually in the ballpark of -85 dBu with a bunch of channels assigned and set to unity gain. This gives a maximum possible dynamic range of 113 dB—pretty impressive considering that the dynamic range of human hearing (threshold of hearing to threshold of discomfort) is around 120 dB.

It's convenient that electronic system noise is usually considerably lower than ambient noise such as traffic and air conditioning.

Usable

dynamic range is rarely greater than 100 dB in a quiet studio and may be as little as 25 dB at a concert with an audience of 20,000 screaming teenage girls. (More civilized concert venues usually allow for dynamic range on the order of 55-70 dB.)

Headroom

Headroom is the ratio of the largest possible undistorted signal to the *average* signal level. Average level is subject to some interpretation depending on whether you're the sound reinforcement engineer at a concert or the promoter asking you to turn it up. At a loud concert, the average sound pressure level may be very close to the maximum possible level (very little headroom). If you're writing the spec sheet of a console, however, you want to be able to show the greatest possible headroom above the nominal level.

When it comes to electronics, the average level is generally considered to be the equipment's nominal operating level. If the nominal level is +4 dBu, our +28 dBu MAIN mixer output will have 24 dB of headroom.

Since there's little you can do about the system's dynamic range once you've chosen your equipment (short of turning off the air conditioner or gagging the screaming teen-

age girls), all you need to do in order to assure undistorted sound is to provide sufficient headroom. Sounds simple, but how much is sufficient?

Crest Factor

Crest factor is the ratio of the peak to the RMS average value of the signal. It's a micro-measurement though, not a long term average like sound pressure level or a sine wave voltage measurement. Crest factor is measured within the waveform cycle. As a simple example, the RMS value of a sine wave is 0.707 times the peak value. So its crest factor is $1/0.707 = 1.4$, or 3 dB.

Studies have shown that typical pop music has a crest factor in the range of 4 to 10, which translates to 12 to 20 dB. This means that we need to be able to provide 12 to 20 dB of headroom for peaks over the average level in order to avoid clipping. This is a requirement for every link in the chain.

Take it From The Top—Setting System Levels

Once you have the whole system hooked up, whether it's your studio control room or a concert sound reinforcement system, verify that you have all the proper connections and it will pass audio. Do this before you start setting gains so you don't get sidetracked with troubleshooting. Put on a CD, make sure it gets to the output(s), and then listen for hum and buzzes that will eat your noise floor for lunch. Once you're sure the system is passing signals cleanly and properly, you can start gain setting.

Pre-flight Check List

- Turn down all the power amplifier input level controls
- Turn the power amplifiers off (so you can run test signals through the system all the way to the amplifiers without driving yourself bonkers)
- Set all gain/level controls to their minimum gain positions.
- Bypass or zero out any equalization, both channel EQ and overall EQ on the outputs.
- Bypass any compressors or limiters, or set their threshold all the way up so they don't compress.

Console Gain Settings

All mixing consoles consist of a mic/line preamp stage, equalization, a channel fader, and channel routing controls (PAN pots and ASSIGN switches). There may or may not be a submaster stage depending on the model and whether a channel is assigned to a SUB or directly to the MAIN outputs. Finally, all channels are mixed together to various outputs (MAIN, AUX, Control Room, etc.) most of which have their own level control.

To set gain structure properly, you want to maximize the *signal-to-noise (S/N) ratio*. This requires some thought and care, as each stage along the path contributes some noise as well as gain. The pessimist looks at this and sees that each stage contributes to the degradation of S/N ratio—and, you know, he's right. But the amount of noise a stage contributes is fixed, so the higher the signal level is at its input, the better the S/N ratio at the output.

It's generally good practice (and this is as much a designer's issue as an operator's) to bring the input signal up to the desired operating level (say +4 dBu) as early in the signal chain as possible. If you need to amplify a microphone by 60 dB in order to get to the final desired output level, it's best not to do it in steps—20 dB at the preamp, 20 dB in the equalizer section, and another 20 dB at the output. You want to put as much of that 60 dB gain as possible right at the preamp stage and run everything else close to unity gain. That's why we have the famous Mackie **Level-Setting Procedure** and those little "U" marks on just about every control.

By soloing a channel, you send its preamp output to the meter. Adjusting the TRIM control for a 0 VU meter reading on peaks sets the preamp gain so that you'll have at least 22 dB of headroom before encountering clipping *in the preamp*.

This may seem like plenty, but remember, you'll be adding up a bunch of channels by the time you get to the main output bus.



Audio signals add as the square root of the sum of the square of their amplitudes. Since 0 VU on the meter represents a level of 0.775 volts, get 24 channels cranking all at once and you'll get 3.8 volts (which translates to around +14 dBu) out of the mixer. That's a

theoretical case—0 VU sine waves on every channel—but with real music, at any instant in time some channels will be peaking higher, others lower, so it all about averages out.

With the input gain set this way, as long as the rest of the console is at unity gain, you'll have at least 20 dB of headroom at the preamp of all channels and that extra loud scream won't clip at the front end. There's also at least 10 dB of headroom at the output with all the channels going. Under any reasonable conditions, the mixer won't be the source of clipping in a system.

Outboard Gear Levels

Many outboard units including power amplifiers are designed so they operate at “unity gain”, and have no level controls at all. Others have input level controls, and electronic crossovers, equalizers, compressors, or other in-line signal processors often have both input and output level controls.

In a perfect world, having no input level control would be OK. The console would add all the gain necessary and provide all the control. Often, however, particularly in live sound situations, it's necessary to turn up the output level of the console to the point where it's at risk of clipping in order to get sufficient volume at the speakers. This is not a good situation and indicates that the power amplifier isn't properly matched to the system.



It's not uncommon for a loud sound reinforcement system to be running very close to clipping most of the time. Very brief clipping is usually not noticeable nor harmful, but sustained clipping sounds awful and can damage your speakers.

Setting Outboard Levels



Begin by turning the power amplifiers off or setting their input gain controls to minimum. Otherwise, you'll be in for a lot of noise.

Set the console so that it's producing maximum output level. Play a CD or use an external tone generator. After assuring that the input isn't clipping (that ol' **Level-Setting Procedure** again), set the channel fader (or AUX SEND control

if you're setting up something connected to an AUX output) to Unity gain, then raise the MASTER fader (and channel fader, if necessary) to bring the output up to the maximum level before clipping as indicated by the VU meter. This is the hottest signal your mixer can send to that output.

If the mixer is connected directly to a power amplifier, skip to the paragraph about setting power amplifier gain. If there's a device or two in between, you'll need to set those levels before getting to the power amp. Keep reading from here.

When going into a device that has only an input level control, set its input control to the point where that device is putting out its maximum level before clipping. If it has a clip indicator or level meter, rely on that to tell you when you're there. If not, crack the gain on the power amplifier (hopefully there's one) just enough to hear and listen for clipping. It'll be pretty obvious.

If the in-line device has both an input gain and an output level control, it's important to set the *input* gain first. Do this by turning the output level up just enough to hear the signal, set the input gain to the point just before clipping, then finally set the output level so the output is just below clipping.

See what's happening here? We're squeezing the last clean decibel out of the console, then, with the knowledge that we'll never be driving it any harder than the maximum console output level, squeezing the last clean decibel out of the next unit in line, and so on. By feeding the outboard chain with the maximum clean console output level, we're assuring that we'll be getting the largest gain boost up front where it belongs, and the rest of the system will be loafing and not amplifying noise.

Setting Power Amplifier Gain

First, understand that the input gain controls on your power amplifier are *sensitivity* controls. They have nothing to do with the amount of power the amplifier can produce. That's fixed by the design. What the control (or the manufacturer, if there's no input gain control) determines is how much input voltage is required in order for the amplifier to produce its full rated power.

With the control turned up full, the amplifier might produce full power with a +4 dBu signal at its input, or it might take +20 or +24 dBu or even -20 dBu. Once

you understand that, it's pretty simple to set the amplifier gain properly. You want the power amplifier to put out full power when the rest of the system is fully cranked.

Setting the amplifier sensitivity too low means you'll never reach full power but, in trying, you can send a pretty darn loud clipped signal (which will be cleanly amplified) to the speakers and your audience. More sensitivity means that you can't take advantage of the full headroom in the console because you'll run out of headroom in the power amplifier first.

To set the amplifier input gain properly:

- Hold your ears and warn everyone else within hearing range. You've been sparing your ears for a while, but now it's time to make noise.
- Turn the amplifier's input gain controls all the way down.
- Crank up your music or test tone to maximum level below clipping as indicated on the console meters.
- If the power amplifier is off, turn it on. Turn up the amplifier's input gain control until you can just hear clipping or its clip indicator just goes on, then back it off a little bit. This is the proper gain setting. Leave it there.

If the volume is too loud (Hah! In your dreams!), it's fine now to turn down the master level at the console. You don't have to turn on all those meter lights just because you paid for them. If it's not loud enough, it isn't going to get any louder—the amplifier is undersized, the speakers aren't efficient enough, or they're placed incorrectly.

In The Studio

The same principles apply when setting up a recording chain. You don't want your recorder to run out of headroom before the mixer output that's feeding it. Otherwise you won't be feeding the recorder a signal with the best signal-to-noise ratio. Conversely, you want your recorder's input sensitivity to be high enough so that you can reach maximum recording level without the mixer clipping.

Many recorders today don't have input level controls, so to a certain extent you're at the mercy of the designers. Fortunately, you're usually safe if you connect inputs

and outputs with matching nominal operating levels.

Mismatched Components

Sometimes the best theoretical setup doesn't work in practice and you need to compensate somewhere. Suppose you've set up your system for maximum headroom and when the console meters are just hitting 0 VU, the sound is so loud that people are leaving. OK, so you turn the console main mix level down and they come back in when your console meters are peaking around -20 VU. But now the meters on the recorder that you have connected to the main outputs are barely moving and you're concerned that you'll get a noisy recording. And on top of that, the console faders are so close to the bottom that you don't have very much working range to get a good mix.

The proper thing to do in this case is to reduce the input sensitivity of the power amplifiers so that with the mixer running at its optimum level (around 0 VU on its meters), you're getting the right volume level in the house. True, you're compromising the gain structure a bit, but you do what you have to do. It's far less destructive to run the last stage in the chain at lower than optimum gain than the first stage.

Chapter 32: Meters—What They Tell You (and What They Don't)

In May 1999, the classic VU meter celebrated its 60th birthday. It served the industry well, and when properly interpreted, it's still a useful tool for indicating loudness.

We've previously addressed the 0 VU (nominal level) and peak (clip) points on the console meters, but there's a lot of useful information in between. Today's digital recording processes have brought us around to taking a hard look at the inadequacies of both the traditional VU meter and the peak meters for level management.

What's a VU?

The *Volume Unit meter* was originally designed to help broadcast engineers keep the overall program level consistent between speech and music. The classic VU meter has well defined mechanical rise-and-fall response rates for the pointer, rates which are actually a little too fast to accurately represent loudness but fast enough to show movement between syllables. Basically, the VU meter was designed to look like you thought it should when responding to speech, making it easy to tell with a glance at the meter when something wasn't working. Engineers learned that brief excursions to the +3 dB top end of the meter scale rarely caused distortion, nor did these brief peaks result in an significant change in perceived loudness.

RMS vs. Peak Audio Levels

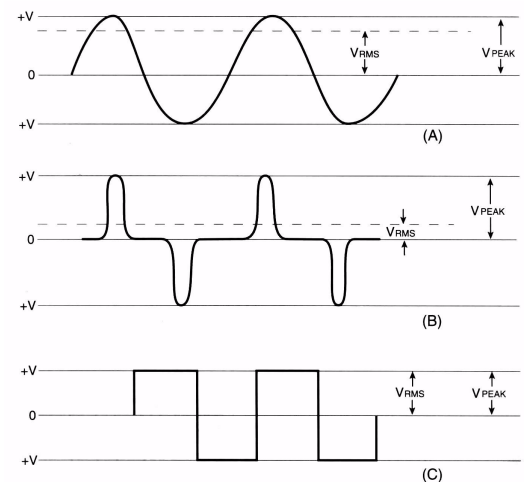
Once sound is converted to electricity, we can represent the audio level by measuring electrical voltage. The audio signal alternates rapidly between positive and negative values of voltage. A symmetrical signal such as a sine wave—a single-pitched note with no overtones or distortion—spends as much time on the positive side of 0 volts as the negative. The numerical average of the positive and negative voltages over many cycles is zero—not a very useful number when what we want to know is how loud a sound that voltage represents.

A method of measurement that corresponds fairly closely with how we perceive loudness is called the RMS (root mean

square) average, a mathematical expression of the amount of energy in the waveform.

The RMS value is calculated by squaring the voltage at each point on the waveform (remember, the square of a negative number is a positive number), summing the squares, dividing by the number of points measured (the more points, the more accurate the calculation), then taking the square root of that quotient.

For a sine wave, the RMS voltage is 0.707 times the peak voltage. You've probably heard of this numerical relationship (it's one-half the square root of 2) between peak and RMS before. Understand, though, that it holds true only for a pure sine wave (Waveform A in the following illustration).



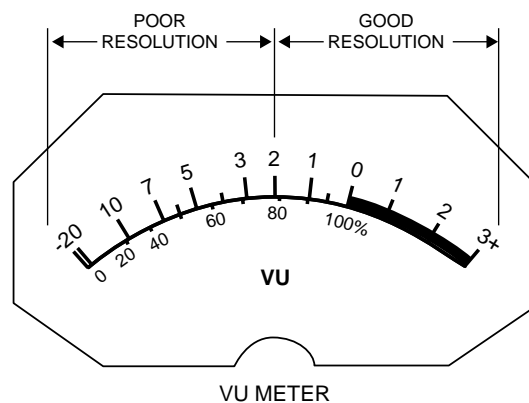
Look at the string of narrow pulses (Waveform B) in the figure. The voltage spends much more time at zero than at some positive or negative voltage. Even though the peak voltage of this pulse waveform is the same as the sine wave above, the RMS value of the pulse waveform is lower. It will indicate lower on a VU meter, and it sounds quieter than a sine wave with the same peak amplitude. This is why a drum may be the correct volume in a mix but appear to read low on the VU meter.

Since a square wave (C) spends all its time either at peak positive or peak negative level, its RMS and peak amplitudes are equal.

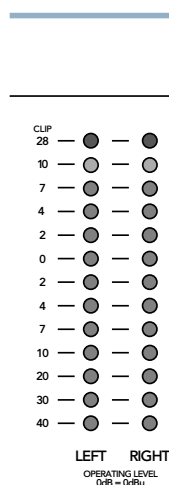
The Classic VU meter

Today we use a VU meter primarily to indicate an impending overload condition, but that was never part of its original function. But is the VU meter a good headroom indicator? Only if you're working with program material that's already consistent in level and doesn't require a generous allowance for surprise peaks. Just look at the meter scale.

While the meter scale has a range of 23 dB, fully half of the scale (the top half) represents only 6 dB.



This is excellent resolution when reading steady tones, but pretty wasteful when working with a recording medium that's capable of handling a dynamic range of better than 90 dB. There's virtually no usable resolution below -10 VU, so if we take it on faith that we have at least 10 dB of headroom above 0 VU even though we can't see it on the meter, we have an indication of only about a 13 dB headroom range.



An LED meter such as you'll find on most Mackie mixers provides a wider measurement range—four calibrated points above 0 VU plus a Clip indicator. However, since you don't know what the signal level is when it's in the cracks between the LEDs, resolution isn't very good above +4 or below -10 VU.

Why compress such a wide range on to a scale with this little resolution? Practicality. Loudness is a logarithmic function. It takes more than twice the signal voltage for something to appear twice as loud, so the indications that you can see still represent loudness fairly well.

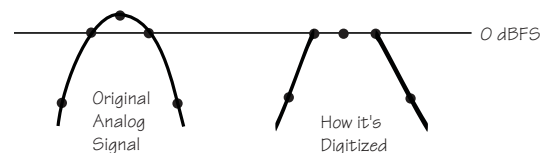
Digital Metering

In the analog world, we count on the designers to provide a reasonable amount of headroom above 0 VU. An analog tape recorder generally doesn't reach unacceptable levels of distortion until the level exceeds +10 VU.

In an analog system, we're not so concerned about how many dB above the nominal level we're running as long as the system gain structure is set properly so that everything peaks out at about the same point.

Digital metering is different, however, because *there is no headroom above 0*. The digitized value of the analog signal can never exceed the number represented by all the bits in the analog-to-digital converter turned on.

We don't call the 0 point on a digital meter 0 VU, it's called 0 dBFS (dB relative to full scale), and zero is all the way at the top of the meter, not in the middle. Any input voltage which exceeds the calibrated 0 dBFS level will be digitized as 0 dBFS, leaving a nice flat topped waveform looking quite unlike the original.



The digital level never exceeds 0 dBFS, but clearly there's something wrong, about which we would like to be warned.

The conventional way of determining a potential digital overload condition is by counting the number of consecutive samples at 0 dBFS, turning on the "Over" indicator when that number exceeds a preset threshold. A common standard for an "Over" indication in a 16-bit system is three consecutive full scale samples. This is reasonable since we can assume that the first sample of a such a string occurred while the waveform was on the way up, and the last occurred on the way down, therefore it must have tried to go over the 0 dBFS level some time in between.

Three consecutive 0 dBFS samples for an Over is pretty stringent. It could mean that only one sample tried to go over the limit (the ones on either side being correctly digitized to full scale). At 44.1 kHz, one sample is too brief to be detected audibly by most

listeners on most program material. With 24-bit systems, where we can allow a little more headroom, many meters indicate a single full scale sample as an “over”. There’s no standard, so you have to use your judgment and your ears.

Making the Meters Agree

It’s nice, when the console’s VU meter is reading 0, that the VU meter on the recorder connected to it will also read 0. Getting them to agree is akin to setting up the gain structure of a system.

In an analog system, it’s usually as simple as setting the Record Level control on the recorder. With a steady tone set so that the console’s meter reads 0, adjust the recorder’s input level control until its meter reads 0 and you’re there. That is, assuming that your recorder has an input level control. Today many don’t, and assume that you’ll provide the recorder with the proper signal level. It’s possible that you’ll never be able to make the meters agree, but hopefully they’ll not deviate by more than a few dB.

Calibrating Digital Meters

Since 0 is at the top of the scale on a digital meter, clearly you don’t want to set the recorder’s input gain so that 0 VU on the analog console’s meter corresponds to 0 (full scale) on the digital recorder.

You do have one pair of points which can correspond, however—the clipping point on the console and 0 dBFS on the digital meter. If your recorder has an input gain control, you can set the console so that it’s putting out a test tone just below the clipping level, and adjust the recorder’s input gain control so that its meter just hits 0 dBFS. That way, both the console and recorder will hit their clipping points together.

Generally recorders with fixed input gain are designed so that a signal at the recorder’s nominal input level (usually +4 dBu or –10 dBV) will give a reading somewhere between –20 and –12 dBFS. If the recorder’s nominal level is at –20 dBFS, this means that it will reach 0 dBFS at an input level of +24 dBu. Since some Mackie consoles can put out +28 dBu before clipping, the recorder can max out before the console does. If you’re using an unbalanced

output from the console, however (+22 dBu maximum output) you won’t be able to reach peak level on the recorder before the console clips.

If you follow the console’s meters and mix at around 0 VU (which is a good idea) you may find that the recorder’s meters keep comfortably below full scale. The recorder manufacturer has allowed himself a generous amount of headroom so that it won’t be his product that’s clipping. Understand this, and don’t worry about it. The meters are tools and are just a guide.

It’s a popular belief that if a digital recorder’s meters aren’t hitting full scale all the time, you’re losing resolution. This is a fact of life. A record level that never gets above –20 dBFS is probably too conservative, but if the level averages around –10 dBFS with peaks getting close to 0, that’s just fine.

Computer Workstation Metering

Something to be aware of with the on-screen metering provided by digital audio workstation (DAW) programs is that redrawing the meter on the screen is usually a low priority task. Some DAW programs don’t even bother—the meters only work when not actually recording, allowing you to set levels during a rehearsal, but not monitor the level during recording.

On-screen meters that function while recording often lag far enough behind when the A/D conversion takes place to allow you to get into trouble if you’re working close to the limit. While it’s difficult to drive a nominal +4 dBu digital input all the way to clipping from your Mackie mixer, many computer sound cards are designed to operate at a nominal input level of –10 dBV. If you connect a –10 dBV input to a +4 dBu output, you can easily drive it to clipping. Beware!

If your recording software provides record level controls, it’s a good idea to find a setting experimentally that allows you to set the proper recording level by watching the mixer’s meter, then leaving the software record gain setting fixed at that point. Check your sound card’s manual though. Some recommend setting the input gain to maximum since they attenuate following A/D conversion, causing a loss of resolution.

Hint: When using this method to calibrate your computer software to the analog mixer's meters, use live sources or some of your own recordings. Commercial CDs are usually heavily compressed and don't have peaks (which the digital meters will indicate) that exceed the average level (which the analog meters indicate) by more than a few dB. Real life, which is what you'll be recording, isn't so neatly packaged.

Practice Safe Level Setting

Consider keeping your peaks a dB or two below 0 dBFS. It won't hurt the signal-to-noise ratio enough to worry about, and if your recording will be tweaked by a mastering engineer, it'll leave a little breathing room for digital level adjustment or equalization.

Chapter 33: Grounds, Shields, Hums and Buzzes

One of the greatest benefits that modern audio equipment has brought us is the potential for wider dynamic range than ever before.

There's a downside, though. Low level hum and other noise that was formerly masked by analog tape hiss becomes audible once we have a really quiet recording medium. Even if you use balanced interconnections throughout your system, with any given mix of equipment, you may find your system noise level 10 to 20 dB higher than the theoretical limit.

Radio Frequency Interference

With electrical and electronic pollution creeping into our environment from devices which have become part of our everyday life, attention to shielding and grounding is more important than ever. Noise induced from electrical power lines (ac hum) is something we've lived with forever, but until recently, RFI (radio frequency interference), unless you were unlucky enough to build your studio next to a broadcast transmitter or factory, was usually down at the inaudible level.

Today, this has changed, and we find potent RF generators in the form of switching power supplies, computers, monitors, and cellular telephones scattered throughout the studio and control room. FCC type approval is one thing—real life audio problems are another. While your computer may be certified not to scramble the picture on your TV set, electromagnetic energy creeping into your audio equipment manifests itself as hisses, crackles, and buzzes that you don't want to preserve in full digital splendor.

Do Balanced Lines Cure All?

It's a common belief that balanced connections are better than unbalanced when it comes to rejecting outside noise, but strangely enough, there's quite a bit of equipment using balanced connections which isn't as noise-immune as we'd like. In fact, some unbalanced equipment that per-

forms the same function can turn out to be quieter.

This can justifiably cause one to lose faith in scientific theory, as well as faith in your friendly dealer or magazine columnist who advised you to spend the extra bucks for balanced equipment.

What's wrong with this picture? A few things, some of which you can easily fix, some are more difficult, and some you'll just have to live with.

You may think your system is properly grounded because all of your power plugs have a ground pin, but you don't know what happens to that green wire once it goes inside the box. There are poorly constructed cables and connectors, both home made and purchased. There's the haphazard scheme of ground-lifting that frequently results when trying to reduce hum and noise in a system.

And then there's what has been dubbed "The Pin 1 Problem", which describes how unwanted signals are introduced into audio equipment through cable shields, which, due to design or construction deficiencies, inadvertently become radio antennas. Don't worry, Mackie is very much aware of this issue and has taken care to design our mixers with good, solid grounding. We can't supply *all* of your studio equipment, however.

What's Ground?

"Ground" is one of the most common (pun intended) electrical terms. In electrical power systems, it means a direct and electrically sound connection to that dirty brown stuff under the house.

Electrical wiring always has one foot firmly planted in the earth to help reduce equipment damage resulting from lightning strikes and other power surges. In electronics, however, the term "ground" has several different meanings, only one of which is a connection to earth.

In the practical sense, ground is a very low impedance path for electrical current to return to its source. That source might be the unit's power supply or another unit that feeds it a signal.

Ground doesn't have to be referenced to the earth in order for equipment to work noiselessly (a portable battery powered recorder is an excellent example). But an inadequate ground connection can create a pretty good antenna, actually making the noise problem worse.

In fact, often disconnecting electronics from the power (earth) ground reduces hum. Surely at some time you've plugged in a noisy piece of equipment through a ground lift adapter and found that your immediate problem (the hum) was reduced. This proves that you have a problem, but it doesn't solve it. *Lifting power grounds is dangerous and should not be practiced.*

Pin 1

Pin 1 is the name we give to the terminal of any signal connector to which the cable shield is (or is supposed to be) connected. On an XLR connector, by industry convention, it's actually Pin 1. On a 1/4" phone plug, it's the sleeve. On an RCA connector, it's the shell. On a terminal strip or multi-pin connector Pin 1 is whatever the manufacturer designates, generally with a "ground" symbol or legend.

The Pin 1 Problem

For effective shielding, Pin 1 should be connected directly and solidly to the unit's chassis. When you plug a shielded cable into a connector on the unit, the shield becomes an extension of the unit's box so there's no break in the shielding and no place for RFI to get in.

Mackie uses a steel chassis with connectors attached directly to the metal, but you may find some gear in your collection that uses circuit-board mounted connectors and even plastic cases, creating a devious route from the cable shield to the unit's electronic ground. Further, that ground point may or may not be connected to the box.

The problem with this type of construction is that any noise current flowing through the cable shield doesn't stop short as soon as it enters the cabinet, but rather goes on to the unit's internal signal reference "ground" point. Once there, it pulls the ground point a little off zero voltage, allowing the noise to sneak into the device's output.

Ground Loops

The indirect electrical route between the shield and the chassis is a path that forms a loop, the famous *ground loop* we've all heard about.

Since this is a fairly low resistance path, any current induced in the loop from a nearby magnetic field (a power transformer or a ac power cord for example) can be fairly substantial. Noise current as low as 1 milliamperes in the shield has been known to cause hum as loud as 20 dB below nominal output in certain pieces of equipment, and noise currents as high as 100 milliamps have been measured in what appear to be straightforward installations.

Is Unbalanced So Bad?

Noise in an unbalanced system is usually attributed to the lack of the common mode rejection offered by balanced inputs, but more likely, it's a result of the Pin 1 Problem. The problem is exacerbated by the fact that the sleeve of PC board mounted 1/4" jacks (a common method of construction today, particularly for bargain priced effects processors) is almost always isolated from the chassis. This is often the result of a manufacturer's attempt to eliminate ground loops in the unbalanced equipment, but it also guarantees an indirect path, if one at all, between the incoming shield and the chassis, and that's not good.

What To Do?

Amidst all of this gloom and doom, there's some good news. Modern audio equipment manufacturers are beginning to understand grounding and shielding issues and are cleaning up their act. (Check out the jacks mounted directly to the steel chassis of your Mackie mixer.)

It's best simply not to buy any equipment which has a Pin 1 problem, but how do you know what's "Pin 1 clean"? It's fairly easy to test for pin 1 problems by inducing a current onto the equipment's ground and see if it appears on the output, but there's no established standard for how much leakage is acceptable.

One-End-Only Shielding

If much of the noise we're trying to eliminate comes in via the cable shield, then why not simply disconnect it? Indeed that's one

of the time honored methods of de-noising a system. The principle of one-end-only (OEO) shield connections, sometimes called telescoping grounds, has been with us for years.

Wiring in this manner was easy to do when engineers built their studios from stem to stern, but the use of custom-built cables with shields on one end and not on the other makes it difficult to accommodate “visiting” equipment or setting up portable systems. Never the less, this is a tried-and-true principle that can often improve the noise performance of equipment that is built with less than an ideal ground configuration. Nearly all equipment dating from the first generation of home studios (which were typically interconnected with RCA phono cables) can benefit from OEO wiring.

Today, however, many studios are wired with store-bought cables which are all built with their shields connected on both ends. This makes OEO wiring impossible unless you take the cables apart and modify them.

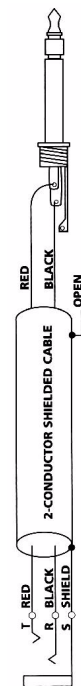
Since disconnecting a cable shield (which actually creates a leak in the shielding) can in some cases reduce system noise, we can logically conclude that in those cases, the ground problem is more serious than the RFI problem. Even with aluminum and plastic case construction, it seems that manufacturers are doing a pretty good job of keeping their equipment relatively immune to stray electromagnetic radiation.

Be Systematic

There are two approaches to OEO grounding. One sounds more fun than scientific, but may actually be more effective in the long run. The other is systematic and boring but generally leads to satisfactory results and is the one to use when building (or rebuilding) an installation from the bottom up.

The fun way is simply this. Start with everything in your system connected together using cables that have shield connection intact at both ends. Turn everything on on, crank up the monitor volume, and listen to the hum and noise. Now, pick a cable, choose an end, and cut the shield. If the hum gets louder, reconnect it. If it gets quieter, leave it disconnected. Then do it again, and repeat the process until either:

- the hum is reduced to an acceptable level (it’ll never go away entirely).
- something stops working because you’ve lifted shields at both ends of the cable and broken the signal return path.
- you get electrocuted (just kidding!).



If you’re using store bought cables with molded-on connectors, you don’t want to cut off one end and rebuild it without the shield unless you know it’ll help. Building a ground-breaking adapter and plugging it in line with a cable at either end will let you experiment without unnecessarily destroying your cables. A plug, a jack, a piece of cable, and a few minutes with a soldering iron and you’re ready to go on a ground loop witch hunt. If you’re using a mixture of balanced and unbalanced cables, build your adapter using a two-conductor jack and plug and two-conductor cable as in the illustration. It’ll work for either balanced or unbalanced connections.

The systematic approach requires that every cable have its shield connected at the source end (the end that’s plugged into an output) and disconnected at the destination end.

This may not always be the optimum configuration since pin 1 problems can occur on both inputs and outputs. You might have a piece of equipment that has a worse problem on its output than the unit to which you’re connecting it has on its input, but by being consistent, you at least assure that every shield is broken at only one point.

AC Power and Grounding

If all the world’s electrical equipment ran on dc (direct current) as Thomas Edison proposed when he first wired New Jersey, we wouldn’t have many of the hum and noise problems we encounter in our sound systems today. There are many very practical reasons why we adopted alternating current as a standard for power distribution, but we got some problems in return.

Probably the greatest source of hum in an audio system is unwanted voltages related to the ac power line creeping in where we don’t want them. This is mostly a result of

normal voltage differences that exist between the chassis of two interconnected pieces of equipment. Those differences are a result of different electromagnetic fields such as nearby power wiring, internal or external power transformers (wall warts) and electromagnetic hash conducted into the chassis through the power line. When there's a voltage difference between the two ends of a cable shield, current can flow through the shield and into our gear. Our goal is to reduce these stray paths as much as possible, and to swamp out whatever gets by despite our efforts at shielding.

Dedicated Power Feeder

The ideal studio or PA installation has a dedicated power feed from the main circuit breaker box, with no other electrical equipment tied to it that isn't part of your audio installation. This isn't difficult if you're building a studio from the ground up or are setting up a studio in a private house, but practically impossible if your studio is in an apartment or college dorm.

The advantages of a separate power feed for your studio are two-fold. First, a direct feed will present the lowest possible impedance to ground for stray noise sources. Your power line won't meander throughout the house acting as an antenna, picking up stray electromagnetic interference from the television set or cordless phone on its way back to the main electrical ground point.

Second, you can be sure (assuming the job was done properly, by a qualified electrician) that the new power feed will have its neutral and ground wires tied together at *one and only one* point—that point where the power feed is grounded when it comes into the building. Sometimes the neutral wire (which is supposed to be at ground potential) can inadvertently get grounded somewhere along its route between the main breaker box and your studio. This creates a ground loop in the neutral wire and assures that you'll be putting some line frequency hum on the chassis of your equipment.

Star Grounding

An excellent defense against ground-related hum problems is to provide a separate, low impedance *insulated* ground wire (14 gage minimum) for each piece of equipment, connecting its chassis (ideally very close to the internal Pin 1 ground point) to a central

ground point in the studio. All the ground wires radiate out from this common ground point like rays of light radiating from a star, hence the name "star grounding".

A good place to tie your ground wires is a copper plate or bar (you can buy one at a electrical supply house). The ground plate should be connected to the ac power ground at the point where the power feed comes into the studio. This sounds like an extreme measure but it's not expensive or difficult to do, and it really can prevent a lot of trouble further on down the road. Don't cheat. Ground everything with its own wire.



It might occur to you to provide your own ground for the studio by driving a copper stake into the earth and running a wire from it up to your central ground point. **DON'T DO IT!!** A nearby lightning strike (it doesn't even have to be a direct hit) can induce a substantial potential difference between two points on the earth, and that potential difference between your well planted ground rod and the ac power ground for the building will ZAP your gear. Guaranteed. It might also burn down your house and void your insurance since this, according to the National Electrical Code, is strictly not Kosher.

Built-in Grounds

Almost everything (including your Mackie mixer) that isn't powered by an external transformer has a three-wire power cord with one of those wires connected to the equipment chassis. It seems like this would suffice for a star ground if you plugged everything into the same outlet.

It does in concept, but not in practice. First off, the ground wire is there for safety, not electronic interference reduction. It's usually lighter than the 14 gage wire that we recommend for your signal ground path, and it's designed to hold up only long enough for a fuse or circuit breaker to blow if there's a certain type of electrical failure in the equipment. In this service, it doesn't matter if there's a couple of volts dropped across the ground wire while it's doing its job, but we want to keep voltage drop in the ground to well below microphone levels.

Second, our studios tend to have a lot of equipment plugged in. Let's face it—we're gear sluts. And what's the most convenient

way to accommodate a lot of ac plugs? An outlet strip. If you can get away with just one outlet strip plugged into your separate power feed, that will do as a star ground, but few studios today are that simple. Outlet strips tend to get daisy-chained, increasing the length of the indirect ground wire, and pretty soon you no longer have a star ground.

The direct run of #14 wire from each piece of equipment to the ground point will provide a substantially lower impedance ground path than the safety ground through the power plug, so it will carry away the noise you don't want.

Rack Mounting

Rack mounting gear is pretty common. Since equipment is attached to the metal rack rails with metal screws, all the metal cases in a rack are connected together electrically—sort of. Since there's usually some paint that you don't want to scratch which partially insulate the screws and the rails from the chassis, this isn't always a reliable, low impedance connection, so it's still important to individually star-ground each piece of rack mounted equipment in order to eliminate inadvertent ac voltage differences between rack mounted chassis.

Sometimes people try to isolate rack mounted gear by using wooden rack rails, plastic screws, and plastic insulating bushings. This is good in theory but pretty chancy in practice. Unless you take great care when installing the units, adjacent panels can touch, blowing your attempt at isolation.

Buzz Off!

Eliminating electrical noise in a studio is one of your biggest challenges, particularly when the studio evolves over time as your production requirements and gear list grows. What happens all too often is you'll bring home a piece this month, and another next month, and then one day you'll crank up the monitor volume and realize that you have a problem that's bigger than you realized and you won't know what caused it. Checking for problems with each addition will keep them from compounding.

In fact, although your patience will probably get the best of you before you're done, a

very good procedure to follow when you first assemble your studio or sound system is to power up the amplifier and speakers first, connect them to the mixer, and check for hum as you plug in each cable. If you notice hum when you plug something in, stop and fix the problem. This takes less time than plugging everything together, discovering that you have a hum problem, then having to take it apart one cable at a time until you find the guilty one.

Chapter 34: Building A Mix

You've probably been wondering when we were going to get around to telling you how to use your mixer for mixing. Well, mixing is an art, a craft, and a skill, and you can't learn how to do it from a book. In this section, we'll attempt to present at least one view of what goes into a mix and you can take it from there.

Mixing is much more than the mechanical process of combining a bunch of sources. You have to look at all the elements to the mix together, and see the big picture. Mixing means continually listening, re-evaluating what you have, and trying different things to get all the pieces to balance with each other. Balance is the key, in fact in Britain, they use the term "balance engineer" for what we in the US call a recording engineer.

There's a lot of personal taste involved—your taste, the artist's, the producer's, and sometimes the drummer's girl friend's. Sometimes you, as the mixing engineer, won't get to make all the decisions, but at least you should have an opinion, and you learn to form valid opinions from doing a lot of listening and a lot of mixing.

A lot of the work *should* have been done in the arrangement, but today we have a tendency, at least when recording, to throw in everything including the kitchen sink and then see if it fits when we finally have an opportunity to listen to the whole thing. The easiest tracks to mix are those where the arranger has thought the piece through and has a good idea of how they'll fit together at the time the parts were written or arranged. Think of the classical composers who never actually heard their symphonies until they handed out sheet music to the orchestra for the first time. They had to mix with pen on manuscript paper with just a little help from the conductor. You have a lot more control in the studio but you have to use it intelligently and most important, musically.

Enough philosophy, let's get down to business.

Space In The Mix

The most important thing to understand about a mix is that everything has its place. The dimensions you have available into which you must put all the pieces are mea-

sured in volume, pan position, and depth (foreground or background).

Even a mono mix need not be flat, it can give you a sense that some things are closer and others further away. A stereo mix with no other manipulation than panning will have no depth and will be boring. In order to give the mix a feeling of three dimensional space, you have to apply panning, effects such as equalization, reverberation and delay, and things have to be at realistic volume levels relative to each other in the mix.

Panning

This should be no mystery by now. Splitting a mono channel out to two loudspeakers and varying the proportion of the signal sent to each gives the sense that something is closer to the right or left speaker. This works pretty well when you're seated between the two speakers, but a listener off to one side or the other won't hear the same perspective. It's important to find panning that works for everyone even though not everyone will hear the same thing.

Depth

When you hear a sound followed by a long echo, your brain tells you that the sound source must be pretty far away. We create the sense of depth or distance in a mix by applying echo or reverberation. The more you add relative to the "dry" source, the further away it appears.

It's easy to understand this concept if you imagine yourself standing in a very live room or maybe even a large cave. All that space will have very little effect on the sound of someone talking right next to you, but when your friend calls to you from the other side of the room, you hear as much or more reverberation surrounding his voice as you hear of his direct, unaffected voice. Your brain processes this and tells you that he's further away than the person right next to you.

Panning Positions

Start your panning with the most important elements of the music. Pan them toward the

center so you'll have room at the sides to build up support around them. Jazz, folk, and classical music is usually panned the same way the group sets up to perform, but that's not a strict rule. With pop music, in which a recording often is often impossible to recreate in performance, you can be more liberal, but there are some important considerations.

Front and Center

The center is the dominant position of the mix, so that's where you want the listener to focus his attention.

For a typical rock song, putting the lead vocal, bass, kick and snare drums in the center is pretty standard. If the vocal is the only melodic element that's occupying dead center, there's less chance that other instruments will compete with it for the same frequency range.

In the vinyl record days (still important for certain forms of music), there are technical reasons for centering the low frequency instruments—bass panned far to one side can cause the needle to jump out of the groove. You may choose to pan a bass solo played up the neck off to one side a bit, but the power behind the mix really belongs solidly in the middle.

Far Out—Hard Left and Right

Usually full (“hard”) left or right panning is reserved for things that are recorded in stereo, stereo effects, delays to create a pseudo-stereo image, or special effects such as doubled or tripled parts, a “whispered” vocal track, or that incidental percussion instrument that you hear once or twice in a song. A hard panned source playing throughout the song tends to be distracting to the listener, taking the focus away from the lead parts.

Left- or Right-of-Center

This is where to put your guitars and keyboards, things that support the lead vocal. The more important an element is to the song, the closer it should be panned to the center. Of course they're all important, but think about what supports the vocal most. That's what should be adjacent. Those instruments that get their own solo space in the song, such as lead guitars, can be somewhat off center, giving the listener a chance to shift his focus a bit. You want to avoid a

position that makes an instrument stick out and sound unnatural, but using panning to separate instruments that occupy similar portions of the spectrum is a good thing. Fiddles and accordions, or trumpets and alto saxes on opposite sides of center are good, unless your concept is to make those sound like a single instrument—which can also be a good thing.

Analyze The Mix

Once you've made some initial decisions about how to pan things, start the playback, bring up the faders so that you can hear everything, and push the Play button.

HANDS OFF THE CONSOLE!

Just sit back and enjoy it one or two times through. Don't fiddle with the knobs or solo tracks yet. Listen to your first mix and analyze what you hear.

Once you've become familiar with the song in this state (you may already be familiar with it from tracking, but it never hurts to step back and give a fresh listen), start thinking about the dynamics of the mix. How does the song build? Does it start with a few instruments and build to the full band? Should it? Could it? What can't you hear? What do you hear too much of? Most important, is the focus *really* on the song or the melody if it's an instrumental?

When the guitar solo comes in, it should temporarily replace the vocalist as the dominant element of the mix. It will if it's just a little off center, but it won't if it's panned way off to one side—it'll sound tacked on like an afterthought. If the keyboard plays a hook in the chorus, you need space for it to be heard as clearly as the vocal. This might mean turning down the rhythm guitar during that hook. Experiment to find out what works once you've decided what you're after.

How does the song build, energy-wise? To be exciting, the energy level has to move, and while it can build and relax throughout the song, it should climax at the most important parts. Remember, you can't tell where the high points are if there aren't any low points for contrast.

Sometimes something just has to go, and this is the time to make those decisions. Having that same hot guitar lick constantly running throughout the song won't necessarily make it exciting. It will be cool the

first time, but after three or four blistering blasts, the listener (other than the guitarist's boyfriend) will wonder what other licks she's got. Start making some notes as to where those hot licks are effective and when to shut her up.

Mix Decisions

Here's where the fun starts. You really want everything to be up front, but it can't be. It's too much of a sensory overload for the listener. You've already decided which elements are the main focus and which take on a supporting role. In a rock song, the bass, kick, and snare are supporting, but they drive the song, so they're really important, too. The mix should center around the rhythm and the vocal, and everything else should fill in the spaces.

In contemporary rock music, it's pretty common for the drums to be just about as loud as the vocal. This is a judgement call, but it's probably what your listeners are accustomed to hearing. The relationship between the kick drum and bass is also important in driving a song, so you don't want to lose the bass either. The bass/kick balance is often the best place to start mixing.

EQ In The Mix Context

It's important to understand that EQ in mixing isn't about the sound of the instrument or voice on its own, it's about the sound in context of the mix. In the EQ chapter, we discussed using equalization to help blend sounds in the mix and carving out spaces in the frequency spectrum to make room for things. Now is the time to apply that knowledge.

When you initially record a track, you want to capture the full range of an instrument, and every nuance. You might have recorded the world's greatest sounding guitar track, but once you put it into the mix, that rich, full spectrum sound will clobber something else if you don't tame it. The same principle applies when you're mixing live sound, only there you have to make decisions in real time, and be ready to defend yourself when the bass player's girlfriend comes up to you and says "Jason's bass sounds *much* boomier than that when he plays it at home. Can't you turn up the bass on it any?"

You aren't quite ready to start riding the faders yet. First work with EQ so that you can clearly separate all the parts into their relative lead/support roles. The idea is to use EQ to constrain the different sounds so they aren't competing for space in the mix. If you're constantly having to turn something up in order to hear it, it's probably because something else is in its way. You might solve the problem with EQ, or maybe with panning, but reserve riding the faders to smooth out playing or singing once you have the basic levels set.

Good planning while you're arranging and tracking can go a long way toward making mixing easier. If the keyboardist is playing in the same register as the singer, suggest that he take it up or down an octave. That sort of thing can become a problem that you really can't fix very well in the mix.

What you want to end up with is a musical "notch" into which each instrument can fit neatly. Much of the time, you don't need to hear the full frequency spectrum of an instrument to recognize what instrument it is—a taste of its sound can often make the listener think it's all there. Understanding that principle and making it work without being obvious that you're tricking the listener can give you a lot of space in which to mix.

Consider EQ in relation to pan positions. You want to have a solid bottom to your mix, so it's reasonable to add some 100 Hz to the kick, bass, and snare which are panned to the center. The bass may need some presence boost around 800 Hz, and some zing in the strings around 6 kHz, but those won't get in the way of the on-center lead vocal who might get a little 3 kHz presence boost, and maybe some 250 Hz "warmth". Neither will a 1 kHz boost to get some crack to the snare. Boost any of those centered instruments in the 2 to 5 kHz region, however, and they'll start to crowd the vocalist's mix space.

Guitars and keyboards are usually panned somewhat off center, so it's OK to give them a little 3 kHz, or 7 kHz that you might also add to the vocal for a little "air". Leave the high boosts at 10 kHz or so for things that are panned out to the sides. Chances are they'll be off-axis to the listener so a little high frequency boost will help bring them back into focus.

Using Reverb to Create Depth

Once you've found a place in left/right and frequency spaces for all the instruments, you'll want to make them sound like they're not all lined up like targets in a shooting gallery. Adding the fore-and-aft dimension can also help reduce crowding in the mix. The more reverb you add (reducing the dry signal to maintain the overall balance), the farther back the instrument will appear.

Today's digital reverbs are so flexible that you can simulate just about any space as well as create some really unlikely ones. Start out your front-to-back mixing by using a reasonable sized room simulation that's appropriate for the style of music you're mixing, and experiment with sending different amounts of the background instruments to the reverb unit. You may choose to put the lead singer into a smaller, less reverberant room than the backup musicians since he's the one who's in your face full time. It's fairly common to use several different reverb units in a contemporary music mix with each one programmed to simulate a different space. Deciding what kind of reverb and how much to use is an acquired skill. Take your time and listen carefully.

Panning With Delays

While we tend to set pan pots "by eye" and figure that's where we'll hear the phantom image, it turns out that if you're listening in the sweet spot (centered between the speakers), the center image is remarkably solid. There can be as much as a 3 dB difference in level between the two speakers of a center-panned source before you'll hear it move off center. On the other hand, you don't have to move your listening position very far off center before the source sounds like it's coming from *just one speaker*. It kind of makes you wonder why you bother using anything but left, right, and center panning. In fact, many mix engineers routinely use only five pan positions (half-left and half-right in addition to the obvious ones).

There's a principle of psychoacoustics known as the *precedence effect* which causes us, when we hear something coming from two sources at slightly different times, to

perceive that it's coming from the source from which it first arrives at our ears.

Panning an instrument fully right and delaying it by 0.1 to 1 ms in the left speaker will make it sound like it's coming from somewhere between the speakers, and that position will be maintained pretty solidly over the full listening space. Adjusting the amount of delay will adjust the apparent position. It's an alternative to using the pan pots, and it can even be quite effective to give a stereo feel to a live sound mix. You can position something off center that won't get lost for a listener on the opposite side of the house since it will appear in both speakers at the same level, just at different times.

Fixing It In The Mix

Hopefully by now your mix will need only a little fine tuning. You may need to re-adjust some volume levels or reverb sends. You may want to boost up a word that the singer swallows or raise the level of the guitar during a solo. These are the things you should start noting.

You will almost always find some places in your mix where you'll have to "ride the faders"—that is, make dynamic adjustments as the music is playing. This is a performance just like playing an instrument and it requires the same kind of skills. You need to be familiar enough with the song to know when your cue is coming, you need to practice so you'll push the fader up just the right amount (and push the correct fader), and be able to re-establish the basic mix after you've made the move.

If there's one guitar track with both rhythm and lead parts, you may have decided that it needs both different panning and EQ when switching roles. A good way to handle this is to split the track out to two mixer channels, setting one up for lead and the other for rhythm. When you mix, cross-fade between the two or mute one and unmute the other. The guitarist did his job in the studio, now as the "console player", it's up to you to get the timing and levels right.

Adding Compression

Some people think a mix isn't done unless something (or everything) is compressed. A compression ratio of 3:1 with the threshold set so that there's 2-3 dB of gain reduction

on peaks can even out a vocal track nicely, but the track has to be pretty good to begin with. A compressor isn't a substitute for a vocalist with good mic technique and control, nor is it an automatic volume control so that you don't need to ride a fader.

It's pretty common to use light compression on the total mix (connected to the MAIN INSERT jacks) to bring out some of the quieter elements of the mix. A compression ratio of about 1.5:1 with fairly long attack and release times won't be very obtrusive. Some people prefer to mix with this kind of compression in place from the beginning, others like to add it at the end if it helps. Again, there are no rules.

As an aside, if you can't get your mix to sound like the commercial recording you're using as a model, it could be that you don't have enough tracks and you don't have enough signal processors. On a really busy project with 40 or more tracks going into a mix, it's almost impossible to mix without compressing nearly all of them to some extent. If your resources are limited, consider compression as you're tracking.

Golden Rules of Mixing

Mixing has a lot in common with playing a musical instrument. You don't learn to do either well overnight. With your mixer and your ears, you have all the tools you need. Here are a few things to keep in mind which will help you keep your focus:

- Decide what's most important and build your mix around that.
- Put up the faders and just listen for a while. Learn the song. Learn when all the solos and background parts come in.
- When you decide that an adjustment is needed, turn the knob far enough so that you can hear a change. Don't boost the keyboard 1 dB at 8.5 kHz because you read that a famous engineer did that. He most likely had a different keyboard or was using it in a different context.
- Listen to your mix on different speakers, take notes as to what you hear, and mix again. It's perfectly normal for a mix not to sound the same every place you hear it. What's important is that it sounds good every place you hear it, and that might take some tweaking.

Chapter 35: Taming The Wild Stage Monitor

Intuitively, a stage monitor setup seems like a really bad idea. The speakers are pretty close to the microphones, often pointing directly at the microphones—a perfect setup for feedback. Further, they muddy up the house sound since the audience hears a lot of bass coming from the back of the monitor speakers but no treble, since that's directional and is aimed in the opposite direction, toward the players. Yet, stage monitors have become an essential part of just about every sound reinforcement gig. Here are some tips to keep them under control.

What Goes Into The Monitor Mix?

Since the monitors almost always get a different mix (or mixes) from the front of house, you have some options as to what and how much to put into those speakers. An inexperienced band will say “We want to hear everything.” That's a sure sign you'll be in trouble. It's important to figure out what's NOT needed in the monitor and start building the mix from there.

Singers always want to hear themselves in the monitors, but it's often not necessary to put electric guitars or bass in the monitors since everyone can usually hear those OK. Unless the drummer wants to hear himself really loud (then you know you're in trouble) there's rarely a need for more drums in the monitor mix than is necessary to keep everyone in time.

Since feedback is usually the greatest problem with the monitor system, the fewer live mics you have with loudspeakers pointing at them, the easier it will be to keep feedback in check.

Speaker Placement

When you set up the stage, take a good look around. Where are the monitor wedges pointing? We almost always use directional mics in sound reinforcement, and a cardioid mic has less sensitivity from the rear than the front. By placing the monitor speakers so that they're aimed right at the null in the mic's pickup pattern, you can increase the gain before feedback by several dB.

Also, obvious as it seems, make sure the speakers are aimed at the players. A monitor blasting out at someone's pants legs won't help any of the players and will only make your job mixing the house more difficult. Sometimes moving a wedge a few inches or turning it a few degrees can make a big difference in how well it can be heard by the players (and can't be heard by the mics).

Pay particular attention to monitors for seated performers. Wedge cabinets are usually constructed so that they fire upward at about the right angle for a standing performer. Prop up the rear edge of the speaker cabinet with a 2x4 if necessary to aim it at the player's head, not over it.

Ringin' Out The System

Feedback occurs at very specific frequencies, the frequencies at which the total system gain from mic to speaker and around again exceeds unity. Using an equalizer (usually a graphic), we can reduce the gain at those troublesome frequencies so that it's below unity.

Start with the equalizer set flat, the master level control for the monitor system turned all the way down, and the Aux Send mix levels set about where you expect them to be. This is only a guess at this point but you have to start somewhere. Now slowly raise the level of the master send and pretty soon you'll start to hear the system howl at some frequency.

Back it off a little so the feedback just stops, and then start pushing the sliders on the graphic equalizer *up* one (and only one) at a time until you find the one that makes the system start to howl. That's your first feedback frequency. With experience, you'll start to recognize those frequencies and automatically reach for the right slider.

Once you've located the hot frequency, pull that equalizer slider *down* several dB and raise the master send level again. You should now find that you can bring it up a little further than before until feedback starts again. Hopefully the feedback this time will be at a different frequency—if not, you didn't take the first frequency down far enough. Move its slider down a few more dB and continue. A couple of times

through this exercise and you should be able to get enough level in the monitors before feedback occurs to satisfy your players.

There are a couple of problems with this process that you should understand, however. The feedback is caused by a very high Q resonance somewhere. That means that the bandwidth of the feedback frequencies is very narrow. The resolution of your equalizer may be such that the feedback frequency falls in between two bands and taming that frequency requires too deep of a cut at one (or even both) of the surrounding frequencies. That can take a pretty big chunk out of the spectrum. You can reduce the feedback, but when the singer does his sound check, he'll tell you that his voice doesn't sound right.

The reason why is that it's missing those frequencies that you just took out with the equalizer. A 1/3-octave band equalizer is better than a one octave band EQ for this application, and a parametric is even better. With a parametric, you can sweep the filter to nail the frequency, then narrow down the bandwidth so as to remove as little of the spectrum as possible.

Another problem is that as the stage fills up, the house fills up, and the air temperature and humidity change, the natural feedback frequencies will change, requiring you to make some adjustments during the show. This is why mixing monitors in a large show is almost always done by someone other than the house mixer, on a separate console. It requires a lot of concentration.

Be A Good Diplomat

To minimize problems, try to minimize what goes to the monitor speakers. It's almost impossible to get the band to turn down, but the softer they play the better they'll hear the monitors, and the fewer problems you'll have mixing the house. But frustrating as it is, don't fight with the band. You really have to give them what they want to hear or they won't put on a good show. A good portion of the job of mixing monitors is personality. Let them know that you want to know what problems they're having, and don't expect them to solve your problems for you.

Often you can make suggestions for the band's setup which will make them sound a lot better and hear better. Is the bass so

boomy that it's indistinct and has to be much too loud to be heard clearly? Help the bass player adjust his tone controls. Get clarity at the source and he'll thank you. Turning or tilting an amplifier or raising it up off the floor often gets the direct sound up to where it can be heard without being so loud, or without having to be put into the monitor mix.

Good Mic Technique

The better mic technique the singers have, the more level you can give them before feedback occurs. Vocal mics have a tendency to pick up everything if the singer isn't really close. When this happens, turning the singer's mic up in the monitor mix not only brings up his voice, but everything else on stage, and that doesn't help him to hear himself any better, he's just louder.

EQ for Other Reasons than Feedback

Try to minimize the frequency range that you're sending to a monitor. Think about what it's doing and get rid of the frequencies that you can't use. A 12" wedge isn't going to reproduce much below about 60 Hz, so use your graphic equalizer or low cut filter to take off some bottom end. It'll make more headroom in the amplifier for the range where you really need it. You can almost always cut some of the monitor signal in the 250-350 Hz area on the drums because there'll be plenty of that floating around the stage acoustically. You may find some other frequencies that you can cut without affecting the sound. The more low frequencies you can cut out of the monitors without the band sounding thin, the less trouble you'll have in the house.

Pflop the Phase

One old road dog monitor engineer trick is to reverse the phase, or more correctly, the polarity of selected monitor speakers. There are a couple of reasons why this sometimes helps.

First, in addition to having unity gain around the feedback loop, the microphone and speaker signals have to be in phase in order for feedback to occur. If they're out of phase, you need more than unity gain

before feedback will occur. Phase of an acoustic source isn't a simple parameter—it's different for every frequency and it changes every time a mic is moved relative to a speaker. But by reversing the polarity of the speaker or the mic, you may be able to put the offending frequency further out of phase in the feedback loop, giving you more available gain before feedback.

The other thing is a physiological one. When you sing, you hear your voice both from the outside and from the inside of your head as well, through the tubes that go up your throat and into your ears. If the speaker is pushing air against the outside of your eardrum at the same time that your voice is pushing against it from the inside, the two are going to partially cancel. If one is pushing while the other is pulling, your eardrum will move further and you'll hear the sound louder. Some people won't notice a change at all, but others will find it dramatic. You can't change the length of the singer's throat, but you can change the polarity of the voltage going to the monitor speaker.

Mackie compact mixers don't have polarity switches for the mics or outputs, but you can reverse the speaker polarity by reversing the two leads from the amplifier to the speaker. It might help. Chances are this trick will only work for one singer or one mic, but it never hurts to try. You might save the day.

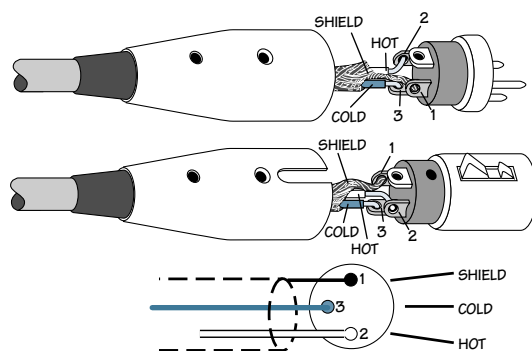
You can make a couple of microphone polarity reversing adapters to carry in your tool kit. A short piece of mic cable that's been "miswired" so that pins 2 and 3 are swapped on one end makes a handy mic polarity inverter. Just patch it in line with the mic cable and you've inverted the polarity of that microphone.

Chapter 36: A Connector Compendium

XLR Connectors

Mackie mixers use 3-pin female XLR connectors on all microphone inputs, with the industry standard wiring conventions: Pin 1 wired to the grounded shield, pin 2 wired to the “hot” (positive polarity) side of the audio signal and pin 3 wired to the “cold” (negative polarity) side of the signal.

Use a male XLR-type connector, usually found on the nether end of what is called a “mic cable”, to connect to these inputs.



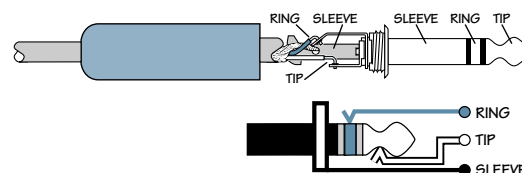
Mackie uses 3-pin male XLRs for some balanced line outputs on some models. These are also wired pin 1 ground, pin 2 high and pin 3 low.

1/4" TRS Phone Plugs and Jacks

TRS stands for Tip-Ring-Sleeve, the three connections on a “stereo” 1/4" phone jack or plug. TRS jacks and plugs are used in several different applications:

- Stereo headphones, and rarely, stereo microphones and stereo line connections. When wired for stereo, a 1/4" TRS jack or plug is connected tip to left, ring to right and sleeve to ground.
- Balanced mono circuits. When wired as a balanced connector, a 1/4" TRS jack or plug is connected tip to signal high, ring to signal low, and sleeve to ground.
- Unbalanced Send/Return circuits. When wired as send/return connector, a 1/4" TRS jack or plug is connected tip to signal send (output from mixer), ring to signal return (input back into mixer), and sleeve to ground.

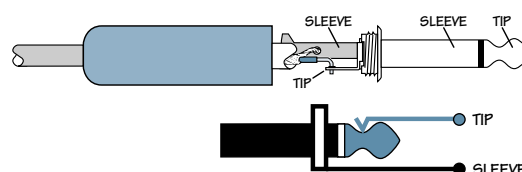
Mackie mixers do not directly accept single-plug type stereo microphones or line level sources. They must be separated into left and right cables that are plugged into their respective inputs.



1/4" TS Phone Plugs and Jacks

TS stands for Tip-Sleeve, the two connections on a “mono” 1/4" phone jack or plug. TS jacks and plugs are used in many different applications, and are always unbalanced. The tip is connected to the audio signal and the sleeve to ground.

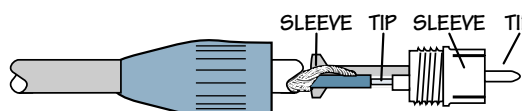
Some examples:

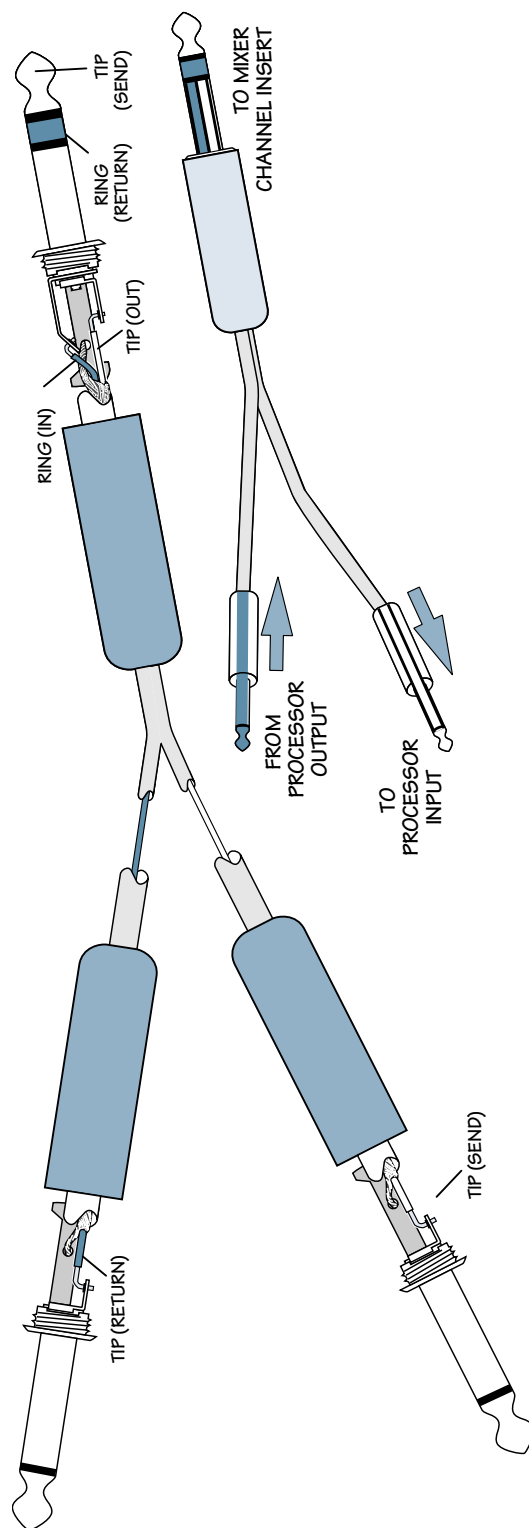


- Unbalanced microphones
- Electric guitars and electronic instruments
- Unbalanced line-level connections

RCA Plugs and Jacks

RCA or “phono” plugs and jacks are often used in home stereo and video equipment and in many other applications. They are unbalanced, and electrically identical to a 1/4" TS phone plug or jack. Connect the signal to the center pin and the ground or shield to the surrounding “basket”. Mackie uses RCA jacks for TAPE inputs and outputs on many mixers. Adapters to convert an RCA male plug to male 1/4" TS plug are available at any electronic shop, like Radio Shack.





some consoles wire the send and return opposite that of the Mackie, you usually won't find one off the rack that's marked "Send" and "Return". To save confusion, it's a good idea to identify and mark yours. A neatly printed paper label laid on the connector body and secured with clear tape won't smudge and will stay in place as long as you want it to.

Remember—a SEND goes to an INPUT, a RETURN goes to an OUTPUT. Hook them up backwards and your processor won't process.

Y-Cable

Sometimes called a "mult", a Y cable is used to split an output so you can send it to two inputs. We've shown Y-cables in multitrack recorder hookups where a bus output is split off to two recorder track inputs. You can also use a Y-cable to connect a track output to two inputs of the console where you may want to pan or EQ the track differently in two places in the mix.

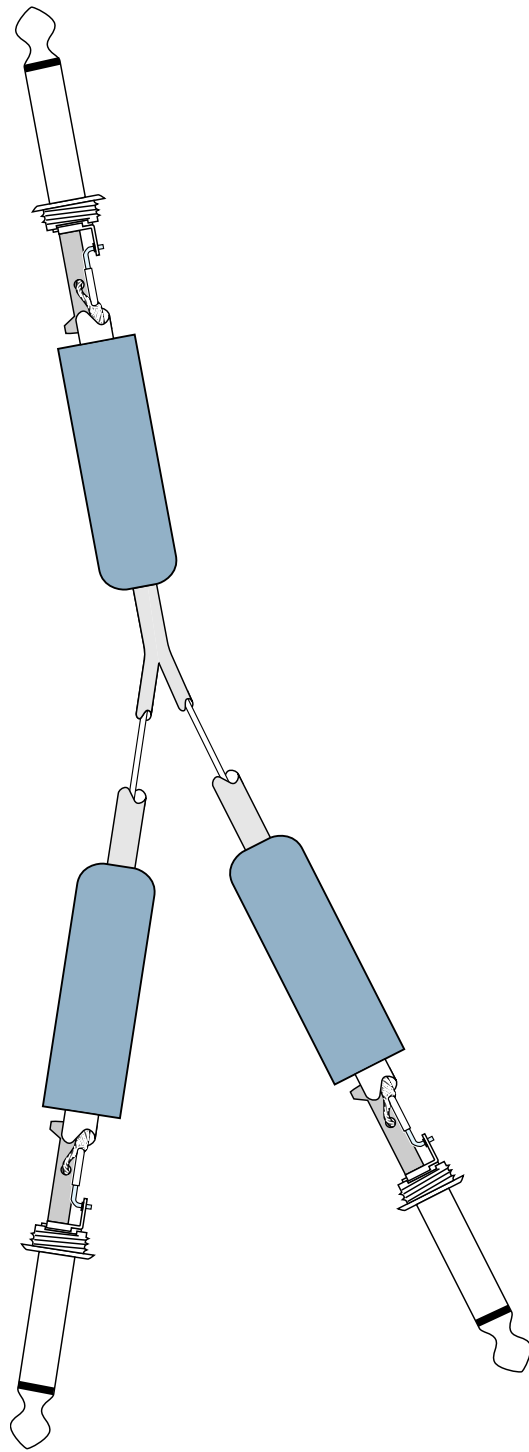
A Y-cable can have either TRS or TS plugs (or even RCA plugs) on both ends, but unlike the Insert cable, it has the same wiring on all branches, though sometimes it's convenient to have connectors of the opposite gender on opposite ends of the cable so you can use it "in line".

Sometimes a Y-cable can be used to combine two outputs and send them to a single input. One application is to get a mono signal from a device which has only a stereo output. How well this works is a function of the characteristics of the outputs you're combining. Most of the time you'll be OK, but occasionally you'll find an output that isn't happy driving the low impedance of another output. Experiment, and listen for distortion.

Insert Cable

A specially wired cable is used for connecting the input and output of a processing device to the INSERT jack on a Mackie mixer. This cable has a TRS plug to mate with the INSERT jack and two TS plugs for the processor input and output.

It's important to know which TS plug is the gozinta and which is the gozouta. Some commercially built Insert cables have them marked Tip and Ring, referring to the contacts on the other (the TRS) end. Since



Chapter 37: Patchbays—Let's Get Normal

“Do I need a patchbay?” is a common question. We've mentioned patchbays in passing, now it's time to explain them in some detail.

Basically, a patchbay is a bunch of conveniently mounted jacks which bring all your audio wiring together in one place. A patchbay makes it easy to reroute signals when your work calls for something out of the ordinary (which, any experienced studio engineer will tell you, is all the time).

As with many things in the studio, we can thank the telephone company for patchbays. In the days before automated switching, a telephone operator (be it in a city or an office building) used a patchbay to connect one of the few incoming phone lines to the individual home or office being called.

What To Patch

Certain connections are essentially permanent in a studio, but others often change with the task at hand. Like the telephone operator, with a patchbay, we can move our limited resources to wherever they're needed. That compressor you used on Channel 3 while tracking vocals last week might be needed on the bass on track 6 during today's mixing session. While you could dig through the spaghetti behind the console to shift a plug between mixer channel insert jacks, it's quicker and less error-prone if the channel insert jacks and signal processor inputs and outputs are in front so you can easily see where each unit is connected.

Ideally, every connection point in your studio should run through a patchbay. In large, professional studios, that's pretty much the way it's done. Small studios generally make compromises in the interest of economy, but often even a modest sound card based studio can benefit from a small patchbay. You can start out small, and once you get the hang of patching, you'll recognize when you need to expand.

Virtual Patchbays

Before delving into the nitty-gritty of patchbay hardware, we must acknowledge that there are studio setups which don't really benefit from a physical patchbay. The obvi-

ous example is the workstation-based studio where all the audio goes into a computer sound card on one pair of jacks, it leaves on another pair of jacks (or maybe as a recorded CD), and all signal processing is performed either with software plug-ins or built-in hardware.

Most routing that's required with such a system can be done within the computer, so a patchbay doesn't really help much until you start to expand beyond two inputs and two outputs. If you have several input sources such as mics or synthesizers, chances are you're connecting those to your Mackie mixer and using the mixer to route those to the sound card's input jacks.

Then there's the studio that has a sufficient number of mixer inputs so that every sound source in your toy box has its own dedicated input channel, and that there's a mixer output dedicated to each recorder track. A demo studio that does everything with one mic, a couple of synths, a sampler, and a drum machine is a good example. If this sounds like you, then the rest of this section is academic. But if you're a gear slut or are always trying new approaches, read on.

The Connector Menagerie

TS, TRS, XLR, RCA, terminal boards. Mackie, as well as other manufacturers of audio equipment, uses several types of connectors. One thing that a patchbay does is standardize those connectors so that every interconnection is always made using the same type of patch cable.

There are a few different kinds of connectors used on patchbays, too. On the front (“business”) side, you'll find standard 1/4" jacks of both the TRS and TS variety, long frame telephone-style jacks, a miniature phone jack called Bantam or Tiny-Tel, and occasionally RCA jacks. On the rear (equipment) side, you'll find solder or wire-wrap terminals, multi-pin connectors, punch-down terminal blocks, and individual 1/4" or even RCA phono jacks. With all those options, which one do you choose? The choice is usually dictated by your budget, your wiring skill, patience, and how much you value your time.

Quality is important here since all your signals will be going through these connectors. A poor connector contact *can* create distortion, and a plug can work loose from a jack that isn't solidly constructed. This is one of those places where you get pretty much what you pay for, and patchbays run from under \$100 to several hundred bucks.

A Balancing Act

Patchbays are available with two- or three-conductor jacks to accommodate either balanced or unbalanced wiring. Patchbays with balanced (TRS) jacks are a bit more expensive than those with unbalanced jacks, but they're a good investment. They'll always work with your unbalanced gear, and they'll be ready for your upgrade to balanced gear. Take this advice on faith and go for a balanced patchbay.

Patching Inserts

One head-scratcher is how to deal with one of the most commonly used patch points, the channel inserts. Nearly all inserts jacks are unbalanced, and like the Mackie mixers, use TRS jacks with the tip (or ring, depending on the manufacturer) being the send and the other contact being the return.

Since many signal processors (which frequently are patched in through inserts) have balanced connections, unbalancing them compromises common mode noise rejection. The good news is that patchbay wiring is usually fairly short and well shielded, so using unbalanced connections when necessary rarely causes harm.

Getting Normal

A basic patchbay is composed of two parallel horizontal rows of jacks. Convention is that the top row is used for outputs, with inputs below them. One of the things that makes a patchbay particularly useful is "normaling" between the top and bottom rows of jacks.

In a normalled patchbay, a signal that comes into a top row jack will go out the jack directly below it without the need for a patch cable between them. Normalled connections are used with things that are, . . . well, normally connected together. With normalled jack pairs and some good plan-

ning, you won't need a tangle of patch cables for your everyday setups.

Why bother wasting all those good jacks on normal connections? Well, because life in the studio doesn't stay normal very long. Let's say your 16-track recorder outputs are normalled through a patchbay to the line inputs or tape returns of console channels 1 through 16. As a result of some "can we do just one more guitar track?" unplanned production, you end up with the lead vocal on Track 2, background vocals on Tracks 5 and 11, the bass on Track 3, drums on Tracks 1, 7, 8, and 10, with guitars and keyboards scattered about on the remaining tracks.

Wouldn't it help your sanity during mixing if you could group the mixer channels in a more musically related manner, say the bass on Channel 1, the kick on Channel 2 (where it's convenient to balance with the bass), the other drums on the next three channels, the guitars all together in a group, then the keyboards, then the vocals? The solution? Un-do the normals and patch them where you want them using your patchbay.

Half- and Full-Normals

There are two flavors of normaling, half and full normal. With a full-normal connection, inserting a plug into either the top or bottom jack of a normalled pair breaks the connection between them.

With a half-normal arrangement, inserting a plug in the bottom jack of a pair breaks the connection, but inserting one into the top jack leaves the pair connected. This turns out to be very useful, and it's the arrangement most commonly used in studio patchbays.

With a half-normal connection you can split an output to two places conveniently—the one to which it's normalled and the one to which you patch it. This makes it easy to connect the guitar track to two console channels, setting up EQ and effects on one channel for lead and the other channel for rhythm. No Y-cable needed.

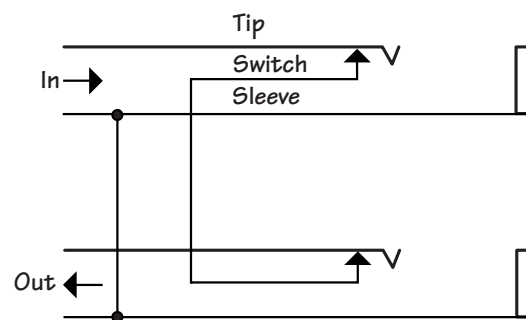
Since modern studio electronics have low impedance outputs and moderately high impedance inputs, putting a double load on a source is rarely a problem. You do need to keep your wits about you with half-normal patching, though. If you patch an output somewhere and haven't patched its corresponding input, you might wonder why

you're hearing that track when you know you have its (patched) fader all the way down. It could be coming through its normalised mixer channel that you forgot to mute.

A full-normal configuration eliminates that confusion, but it also precludes easy "multing" of a signal.

Although microphone patchbays aren't all that common, this is one place where the full normal configuration is almost always used. Since load impedance can substantially affect the sound of a microphone, and since most mic preamp inputs have relatively low impedance, you don't normally want to present a mic with a double load.

A normalised jack has a switch contact in addition to the signal contact that connects with the plug. Without a plug inserted, the switch is closed, electrically connecting jack terminal to its associated switch terminal. Inserting the plug breaks this connection.

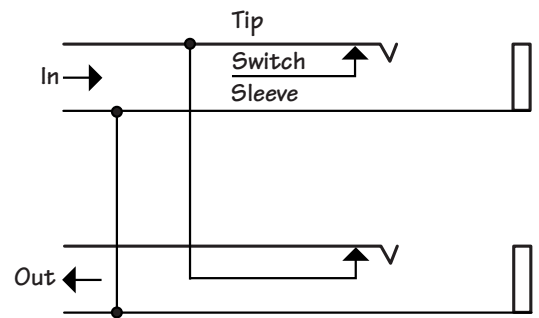


Jacks Wired Full Normal

A pair of jacks wired in the full normal configuration is illustrated above. Unbalanced jacks are shown for clarity, but balanced jacks work the same way. With no plugs inserted, the signal comes in to the top jack, flows through its switch contact to the switch contact on the bottom jack, and then out the bottom jack.

Inserting a plug in either jack opens its associated switch, breaking the connection between the two jacks.

In the half-normal configuration illustrated below, the switch on the bottom jack is connected directly to the input source, the tip contact of the top jack.



Jacks Wired Half Normal

Since in the half-normal arrangement, the switch contacts on the top jack aren't used, inserting a plug in the top jack doesn't break the connection between the input and output of the jack pair. Inserting a plug in the bottom jack, however, opens its switch, breaking the normal connection.

No Normals

You don't always want all of the jack pairs in your patchbay normalised. For example, you don't want a compressor's input and output connected together when it's not patched anywhere. It doesn't do any good, and it might cause the unit to go into oscillation because you're feeding its output back to the input.

Most patchbay jacks are configurable to be normalised either way, or not at all, as you choose.

E-Z Patchbays

When patchbay shopping, you'll find front panel jacks that accommodate common 1/4" phone plugs, either balanced (TRS) or unbalanced (TS). Most of these patchbays also have 1/4" jacks on the rear so they can be connected to your studio gear using easily obtainable pre-made multi-channel "snakes". Using a patchbay with common style front panel jack makes patching in "visiting" gear simple.

The style of patchbay with 1/4" jacks front and rear is usually called "prewired" because you don't have to solder wires to the jack terminals themselves, you just plug in mating cables.

There are a couple of variations on prewired patchbays that can save you some wiring time if they apply to your setup. Some have multi-pin connectors on the rear side wired per the Alesis ADAT or TASCAM DTRS convention (which Mackie also uses on dig-

ital multitrack recorders) so that a recorder can be connected to the patchbay using a special cable with multi-pin cables on each end, eliminating a lot of plugs.

Another patchbay variation is one built specifically for connecting to unbalanced insert jacks. A single TRS jack on the rear splits out to a half-normalled pair of TS jacks on the front. This allows you to connect the patchbay to a TRS insert jack with a single two-conductor cable (breaking the normalled connection in the console jack), and regain the insert send/return normaling at the patchbay jack pair.

Long Frame Patchbays— What the Pros Use

Long frame jacks (the traditional telephone company style) are found in pro studio patchbays. They're 1/4" in diameter but are designed for a special plug, one with a tip smaller in diameter than the ring or sleeve parts. This design prevents the small diameter tip from touching the ring contact of the jack on its way in, assuring that the ground (shield) is connected before either signal lead.

Long frame patchbays are expensive when purchased new, but they're much more robust than standard TRS jacks and will last a lifetime. You can put a standard 1/4" phone plug into a long frame jack, but leaving one in place for more than a short time can cause the jack contacts to be permanently bent out of shape. If you use long frame patchbays, be sure to have some adapters to standard 1/4" plugs handy to accommodate that occasional visiting signal processor or keyboard.

Bantam™ (also called Tiny-tel) jacks are miniature long frame jacks, popular because they're half the size of 1/4" jacks, allowing twice as many to fit in the same panel space. That's important in a large system. These patch bays are pretty much the standard in larger modern studios—it doesn't take very long before you need a few hundred patch points and Bantam jacks can save a lot of rack space.

What to Normal?

When planning a patchbay, you must decide which connections will be normal, which will be patched, and which will be

permanently connected and never see the patchbay. Two studios with identical equipment may set up their patchbays differently because of the work that they do, but there are certain things that always make good sense to bring to the patchbay. The following are by no means all the possibilities, but will get you thinking in the right direction.

Console channel inserts are good candidates for the patchbay. You usually have a many more mixer channels than things to patch in line with them, and you might need a particular processor on almost any channel at some time or other.

Since most 1/4" patchbays have about 24 jacks to a row, a sensible arrangement for a 16 channel console is to connect the insert jacks to 16 normalled pairs of jacks, and connect your compressors, equalizers, and serial type effects processors to the remaining 8 pairs of jacks without normaling. Use an Insert cable from the console to the patchbay, the same type of cable you'd use to connect a processor to the console insert jack.

Remember the golden rule of patchbays—sends (outputs) go to the top row of jacks, returns (inputs) go to the bottom row. Knowing that your patch cables always go from a top to a bottom row will keep you from making mistakes and patching something in backwards.

Most people adhere to that rule when wiring things that are purely sources (insert outputs, recorder or preamp outputs) or destinations (insert returns, console line inputs, recorder inputs), but break the rule for devices which are inserted in line with the signal flow such as equalizers, compressors or effect processors—devices which have their inputs and outputs associated with the same signal path.

The logic here is that the signal flow on the patchbay is always downward, so putting a processor input on the top row and its output on the bottom row makes a certain degree of sense. Use whatever logic will be most intuitive to you at 3 AM after a long overdub session and the band asks for a mix before they go home.

Multitrack recorder outputs are usually normalled to console line inputs or tape returns, but sometimes it's difficult to decide what console outputs, if any, to normal to multitrack recorder inputs.

Normalizing subgroup bus outputs (or main stereo outputs if it's not a multi-bus console) to the recorder inputs is fairly common. By using bus assign switches and pan pots, you can route any console input(s) to any recorder track while still having access to the recorder inputs for a direct patch from the console preamp (insert send or direct output) or the output of an outboard mic preamp.

It's very convenient to have all the recorder inputs, outboard mic preamp outputs, and at least a few console direct outputs available on a patchbay. Since you never know where you'll be using an effect processor, it's handy to have them wired to the patchbay too.

Having at least one auxiliary send and return normalled on the patchbay to a reverb or multi-effect processor is handy so that processor will be handy during tracking without any patching, and can be used either with the send and return, or patched in line (serial) with a track when recording or mixing.

Control room outputs usually can be wired directly to your monitor system. It's not likely you'll be sending them somewhere else other than perhaps to a set of alternate monitors, for which you'll probably use a switch. By connecting 2-track recorders (both inputs and outputs) through the patchbay, your usual mixdown recorder can be normalled to the mixer's main outputs, but you can easily patch your DAT to a CD-R or cassette deck for making dubs.

Wiring a Patchbay

Incorporating a patchbay into a studio setup requires quite a bit of additional wiring. You may have reservations about adding a patchbay because of all the extra cable length, but even -10 dBV unbalanced equipment has no problem driving upwards of 50 feet of good quality, well shielded cable. Don't sweat it, but don't go overboard on cable length either.

The easiest patchbays to install are the prewired ones with 1/4" jacks front and back. Configure each pair of jacks for the normalizing you want, plug in the cables, and you're ready to start patching. This style of patchbay is also the easiest to reconfigure when you decide, after living with it for a little while, that you'd like to relocate a few patch points.

Patchbay Maintenance

While there's nothing to burn out or align, patchbays do require a little TLC. If you don't have a spray can of Caig DeOxit™ Red contact cleaner in your studio, get one. A light spritz in the jacks every few months, and wiping the patch cable plugs with a cloth on which you've sprayed a little cleaner will keep your patchbay connections noise free. Don't forget the jacks in the rear, too, if you have prewired patchbays.

A Simple 8-Track Patchbay Setup

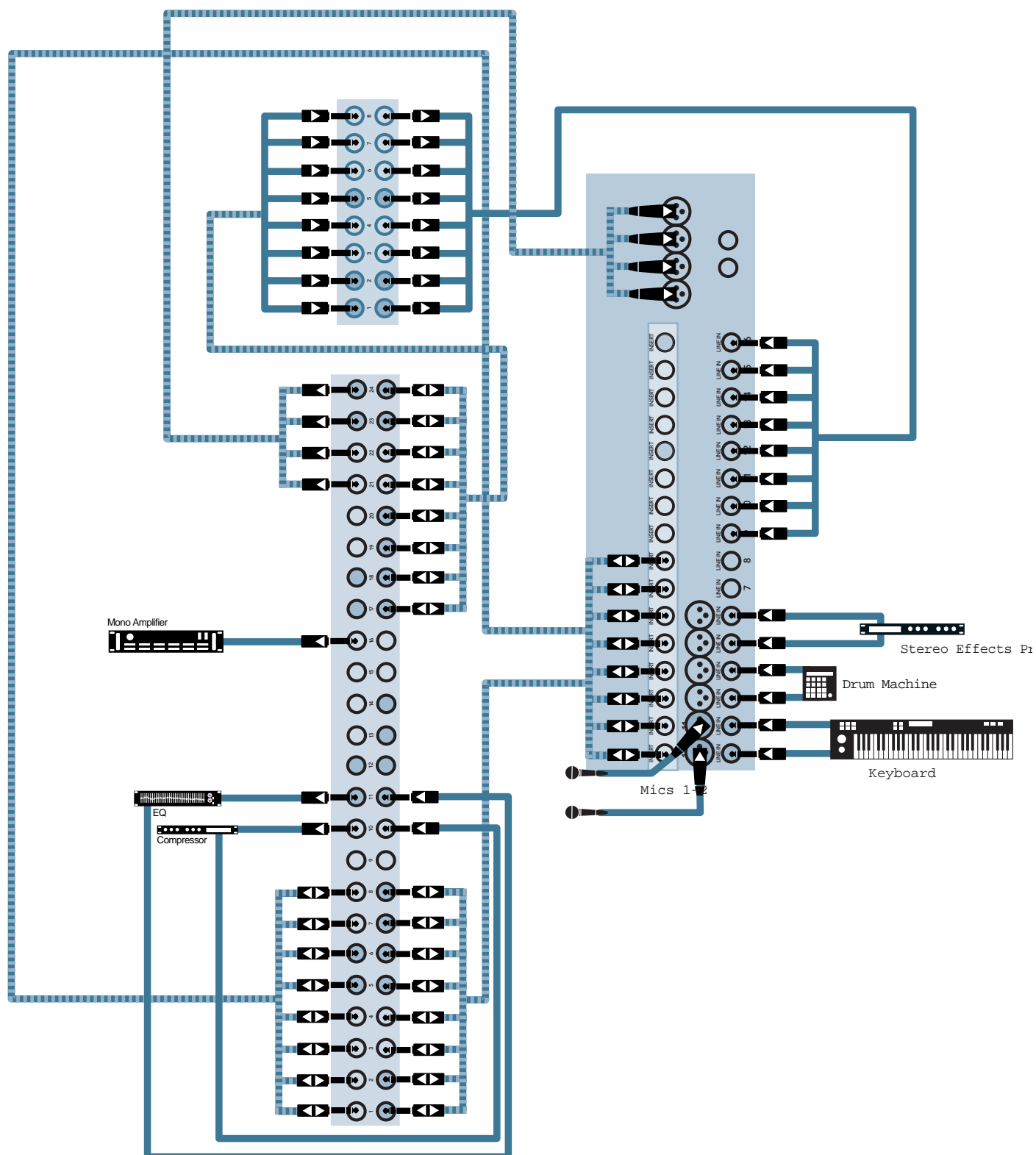
The illustration on the following page is just one suggestion of how a simple patchbay might be set up. For clarity, not everything is shown, and in reality you'd probably want a second patchbay, but we didn't want to scare you off immediately.

Eight channel inserts are shown connected to half-normalled jack pairs. The outboard processors are next on the patchbay with non-normalled jacks.

The four mixer subgroup outputs are brought to the patchbay and normalled to four recorder inputs. The other four recorder inputs are brought to the patchbay, but aren't normalled anywhere can be patched either to a subgroup, a channel insert output, or the output of the outboard mic preamp.

The recorder outputs are shown connected directly to mixer line inputs. That's because we ran out of space on this patchbay, and you will, too, sooner than you thought.

Adding a second patchbay to accommodate recorder outputs and console line inputs will allow you to rearrange tracks in a more sensible order at mixdown time. It will also give you some more jacks for the rest of the channel inserts and a few more processors. That should hold you for a while. You probably won't get confused with only eight tracks, but by the time you expand to 24, you'll be a patchin' fool.



Chapter 38: Maintenance, Troubleshooting Tips, and Repairs

Mackie mixers are notoriously bulletproof and reliable, but, hey . . . stuff happens. Any electronic product with this many components can occasionally suffer a minor casualty.

Even if we could build a mixer that never fails, there are those acts of nature or Congress that tend to fall upon consoles occasionally—spilled coffee, a toppling monitor, a thoughtless visitor eating peanut butter crackers over the fader slots.

This section offers some suggestions on how to get your mixer healthy again, or determine for sure that it's time for it to visit the hospital.

Troubleshooting



It'll be to your benefit to do a little basic troubleshooting first to determine whether your mixer is really malfunctioning or if there's pilot error or simply something unplugged. Searching for a loose power plug or signal connection before you dig into the mixer might save you some down time. It'll also save you money because if you send your mixer off for repair and the problem can't be duplicated at the factory or repair shop, you're out of business for a while, and you may be hit with a service charge as well as shipping costs.

We could write a whole book on troubleshooting, but we won't. Here are some things to look for that might get you right back on the air.

Power Considerations

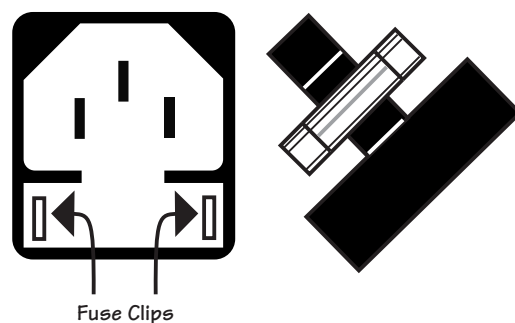
This sounds insultingly simple, but if the whole console is dead, is it plugged in? Really? Is the Power switch on? That's not as silly a question as it seems. On many Mackie mixers, the main power and phantom power switches are adjacent. If you switched the console on by feel, no matter how many times you've done it before, you may have flipped the wrong switch.

Is the power cable securely seated in the socket at the back of the mixer or power supply? Did someone unplug it from the ac outlet so they could plug in their boom box

to record the show? Did a fuse or circuit breaker in the building blow? Plug something else into the ac outlet to be sure there's juice coming out.

Check the Fuse

All Mackie mixers use a standard, fused IEC power connector. There's a fuse located inside the chassis-mounted power connector that you can check. Remove the power cable and slide out the fuse holder compartment. It comes straight out, perpendicular to the panel. Gentle prying with a tiny screwdriver or a thin blade will release it.



Once you've separated the fuse holder from the power connector, you can remove the fuse from the holder easily. Try it now, before you have a problem, note the size and rating of the fuse, and get a spare. Radio Shack might not be open in the middle of the gig.



A note about fuses: The fuse is there to protect the wiring, not to prevent disasters. A fuse can open up from a hard physical knock, a voltage surge on the power line, and sometimes it can just die of natural causes. If the fuse has blown, however, chances are pretty good that the replacement fuse will blow, too. If it does, this is a sign that your mixer is due for a trip to the hospital.

Intermittents

These are the most pesky problems. Sometimes it's there, sometimes it's not. Faulty cables, dirty plugs or jacks are often the culprit. A plug can sit in a jack doing its job perfectly for months and then suddenly decide (based on temperature, humidity, or the phase of the moon) to short out, stop

conducting, or partially conduct, causing crackling noises or distortion rather than simply interrupting the signal flow.

If you're having trouble with a single channel or a single signal path, for gosh sakes swap the cable with a known good one before calling the doctor. A quick shot of contact cleaner in a troublesome jack will often solve the problem. DeOxIt™ Red 5 from Caig Laboratories is effective and harmless to the inner workings when used as directed. It's stocked at most pro audio dealers and music stores that sell sound equipment.

Troubleshooting Checklist

Dumb as some of these suggestions may seem, troubleshooting is a systematic process. Some of the simplest problems are often overlooked in the heat of the moment. Keep cool and check everything.

Bad Channel

- Is the channel assigned to someplace where you can hear it? Is the MUTE/ALT 3-4 switch (1202/1402) in the correct position?
- Is the FADER or GAIN knob turned up?
- Is the problem with a mic channel? Is it a condenser mic that requires phantom power? Is the phantom power switch turned on?
- Try unplugging any INSERT devices to see if the channel path is restored. If it's OK, the problem is with the inserted device or the cable.
- Is the signal source working? If it's a tape deck or instrument, is its volume control turned up? If it has a headphone jack, listen with headphones to see if it's working. Solo the channel and see if the **Level-Setting Procedure** works. Try plugging the same source into another channel set up exactly like the problem channel. This will tell you if the trouble is with the channel or the source.
- Is the MUTE (ALT 3-4) switch pressed?
- On the 8-Bus, are the MIC/LINE and FLIP switches in the correct position?

Bad Output

- Is the associated level knob (Main output, AUX master, etc.) turned up?
- If the problem is with one of the MAIN OUTS, try unplugging all the others. For example, if it's the 1/4" Left Main out

that's dead, unplug the RCA and XLR Left outputs. If the problem goes away, it's not the mixer, it's most likely a short in one of those cables.

- If it's a stereo pair, try switching the channels. For example, if a left output is presumed dead, switch the left and right cables at the mixer end. If the problem switches sides, it's not the mixer.

Noise

- Turn the channel GAIN and AUX RETURN knobs down, one by one. If the noise disappears, it's either that channel or whatever is plugged into it, so unplug whatever that is. If the noise disappears, it's from your whatever.

Power Amplifier and Speaker Troubles

The usual channel swapping techniques work for diagnosing problems beyond the mixer. If one side of the PA system is louder than the other, after checking to be sure you don't have a source panned off center or have a submaster fader turned down, try swapping speaker cables at the amplifier end. If the problem moves to the other speaker, that indicates trouble with a speaker or speaker cable. If the problem remains on the same side, there's trouble with the amplifier output on that channel. Swapping input cables to the amplifier (if the amplifier is external) will tell you if you have a good input signal to the amplifier.

On the PPM series mixers, plugging something into the POWER AMP IN jacks disconnects the mixer outputs from the built-in power amplifiers. If you're not using an outboard amplifier, those jacks should be empty.



Prior to PPM Serial Number *****, plugging into the POWER AMP IN jacks does not disconnect the power amplifier from the mixer output. The power amplifier gets the combined signal from the internal mixer and the external device plugged into this jack.

Weak Bass

It's possible that the polarity of one of the speakers is reversed with respect to the other. This could be caused by a miswired cable or miswired speaker. Swap the two speaker leads at the power amplifier terminals on one channel only. If you have a powered mixer, try a different speaker cable.

PPM Mixer Shuts Down When The Music Gets Loud

This usually indicates that the thermal shut-down has activated.

- Check to be sure inputs are not clipping. Clipped audio drains more power than clean audio.
- Check for adequate air flow behind the mixer in the area of the heat sink. Move it into the shade if you're working outdoors. A sunbonnet looks cute. (On the mixer, not on *you*!)
- Make sure the speaker impedance is 2 ohms or greater. An almost-shortened speaker cable (one with a single strand of wire shorting to the opposite conductor) may still deliver some sound to the speaker but will look to the amplifier like you've connected a few extra speakers.

A Clean Fade(er)



Faders are not rocket science—they operate by dragging a metal contact (the wiper) across a carbon-based strip (the track). If you smoke or eat greasy hamburgers too near your mixer, it's possible for airborne crud to land on the fader track. Should that happen, you may hear scratchy noises or signal dropouts as the wiper stumbles over the crud. Do all you can to keep airborne crud out of your profession.

Work in air conditioned rooms whenever possible, avoid smoking near the mixer, keep food and drink away from the mixer, and for pity's sake, never put the mixer in your kitchen!

We also recommend “exercising” the faders—give them a few full-travel swipes once a week or so. That will help scare the crud away. As a rule, we do *not recommend the use of spray cleaners in pots and faders (though they're fine for jacks), but a blast of dry, purified air (computer and camera stores sell aerosol powered canned air in the “cleaning supplies” department) is good first aid.*

Repairs

Service for Mackie mixer in the U.S. is available from authorized warranty stations or from Mackie Designs, located in sunny Woodinville, Washington. Service for mixers living outside the United States can be

obtained through local dealers or distributors. If your mixer needs service from Mackie, follow these instructions:

1. Review the preceding troubleshooting suggestions. Please.
2. Call Tech Support at 1-800-258-6883, 9am to 5pm PST, to explain the problem and request a Return Authorization number. Have your mixer's serial number ready. You must have an RA number, or we may refuse the delivery.
3. Set aside the power cord, manuals, or anything else that you'll ever want to see again. We are responsible for the return of the mixer only.
4. Pack the mixer in its original package, including the foam end caps and box. This is **VERY IMPORTANT**. When you call for the RA number, please let Tech Support know if you need a new box and we can send you an empty one. *Mackie is not responsible for any damage that occurs due to non-factory packaging.* The shipping company probably won't be very sympathetic either, but using the factory packaging goes a long way toward settling damage claims.
5. Include a *legible* note stating your name, shipping address (no P.O. boxes), daytime phone number, RA number and a detailed description of the problem, including how we can duplicate it.
6. Write the RA number in **BIG PRINT** on top of the box.
7. Ship the mixer to us. We suggest insurance for all forms of cartage. Ship to this address:

Mackie Designs

SERVICE DEPARTMENT

16140 Wood-Red Rd. NE Ste 5

Woodinville, WA 98072

Ask Tech Support for the latest turn-around times when you call for your RA number. We normally send everything back prepaid using UPS BLUE (Second Day Air). However, if you rush your mixer to us by Next Day Air, we'll ship it back to you UPS RED (Next Day Air).

Section 4: Appendix

Section 4 Table of Contents

Appendix A: Glossary

This Glossary contains brief definitions of many of the audio and electronic terms and acronyms used in discussions of sound mixing and recording. Many of the terms have other meanings or nuances or very rigorous technical definitions which we have side-stepped here because we figure you already have a lot on your mind.

If you'd like to get more information, there are plenty of useful textbooks out there. We recommend the following titles: *The Audio Dictionary*, by Glenn White; *Tech Terms*, by Peterson & Oppenheimer; *Handbook for Sound Engineers*, by Glen Ballou, *Mackie Mixer Book* by Rudy Trubitt and *Sound Reinforcement Handbook*, by Gary Davis.

ACRONYM

An acronym for A Contrived Reduction Of Nomenclature Yielding Mnemonics

AFL

An acronym for After Fade Listen, which is another way of saying post-fader solo function.

assign

In sound mixers, assign means to switch or route a signal to a particular signal path or combination of signal paths.

attenuate

To reduce or make quieter.

aux

Short for Auxiliary.

auxiliary

In sound mixers, supplemental equipment or features that provide additional capabilities to the basic system. Examples of auxiliary equipment include: serial processors (equalizers, compressors, limiters, gates) and parallel devices (reverberation and delay).

aux send

A mixer bus output designed to send a signal to an auxiliary processor or monitor system.

aux return

A mixer input (sometimes a pair of inputs) with limited control capabilities, intended for bringing the output of an auxiliary processor or other line level source into the main mix bus. Aux returns can sometimes be assigned to other busses in the mixer.

balanced input

An input consists of two leads, neither of which is common to the circuit ground. This is a “differential pair”, where the signal consists of the *difference* in voltage between the two leads. Balanced input circuits can offer excellent rejection of common-mode noise induced into the line.

balanced output

In a classic balanced audio circuit, the output is carried on two leads (high or + and low or -) which are isolated from the circuit ground by exactly the same impedance.

A symmetrical balanced output carries the same signal at exactly the same level but of opposite polarity with respect to ground.

A special case of a balanced output carries the signal on only one lead, with the other lead being at zero voltage with respect to ground, but at the same impedance as the signal-carrying lead. This is sometimes called *impedance balanced*.

bandwidth

The band of frequencies that pass through a device with a loss of less than 3 dB, expressed in Hertz or in musical octaves. Also see Q.

bus

An electrical connection common to three or more circuits. In mixer design, a bus usually carries signals from a number of inputs

to a mixing amplifier, just like a city bus carries people from a number of neighborhoods to their jobs. It comes from the British “omnibus”.

Cannon

A manufacturer of electrical connectors who first popularized the three-pin connector now universally used for balanced microphone connections. In sound work, a Cannon connector is taken to mean a Cannon XLR-3 connector or any compatible connector. You can tell an audio geezer because he refers to this connector as “Cannon”. Today the term “XLR” is more common.

cardioid

Heart-shaped. In sound work, cardioid refers to the shape of the sensitivity vs. direction plot for a particular style of directional microphone. A cardioid mic rejects sound arriving from the rear.

channel

A functional path in an audio circuit: an input channel, an output channel, a recording channel, the left channel and so on.

channel strip

The physical realization of an audio channel on the front panel of a mixer; usually a long, vertical strip of controls.

chorusing

A time-based effect available in some digital delay effects units and reverbs. Chorusing involves a number of moving delays and pitch shifting, usually panned across a stereo field. Depending on how used, it can be lovely or grotesque.

clipping

A form of severe audio distortion that results from peaks of the audio signal attempting to rise above the capabilities of the amplifier circuit. Seen on an oscilloscope, the audio peaks appear clipped off. To avoid clipping, reduce the system gain in

or before the gain stage in which the clipping occurs. Also see headroom.

common mode

A signal which is referenced to the circuit common point, usually chassis ground.

condenser

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

console

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

cueing

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

dB

See decibel.

dBm

A unit of measurement of power in an electrical circuit, expressed in decibels referenced to 1 milliwatt. The “m” in dBm stands for “milliwatt.” In a circuit with an impedance of 600 ohms, this reference (0 dBm) corresponds to a signal voltage of

0.775 VRMS (because 0.775 V across 600 ohms equals 1 mW).

dBu

A unit of measurement of audio signal voltage in an electrical circuit, expressed in decibels referenced to 0.775 VRMS into any impedance. Commonly used to describe signal levels within a modern audio system. Nobody is really sure if “u” stands for anything.

dBv

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

dBV

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

decibel (dB)

The dB is a ratio of quantities measured in similar terms using a logarithmic scale. Many audio system parameters measure over such a large range of values that the dB is used to simplify the numbers. A ratio of 1000:1=60 dB. Since dB is a unitless quantity, it doesn't matter if it's volts or dollars. (just try asking the chief engineer for a 3 dB raise) When one of the terms in the ratio is an agreed upon standard value such as 1.23 V, 1 V or 1 mW, the ratio becomes an absolute value, i.e., +4 dBu, -10 dBV or 0 dBm.

delay

In sound work, delay usually refers to an electronic circuit or effects unit whose purpose it is to delay the audio signal for some short period of time. Delay can refer to one short repeat, a series of repeats or the complex interactions of delay used in chorusing or reverb. When delayed signals are mixed back with the original sound, a great number of audio effects can be generated,

including phasing and flanging, doubling, Haas precedence-effect panning, slap or slapback, echo, regenerative echo, chorusing and hall-like reverberation. Signal time delay is central to many audio effects units.

detent

A point of slight physical resistance (a click-stop) in the travel of a knob or slide control. Most knobs on Mackie mixers are detented to indicate their unity gain or centered position. It's handy in the dark.

dipping

The opposite of peaking, of course, used in audio to describe the shape of a frequency response curve. A dip in an EQ curve looks like a valley, or a dip. Dipping with an equalizer reduces a range of frequencies. (See guacamole.)

doubling

A delay effect, where the original signal is mixed with a medium (20 to 50 msec) delayed copy of itself. When used carefully, this effect can simulate double-tracking (recording a voice or instrument twice).

dry

Usually means without reverberation, or without some other applied effect like delay or chorusing. Dry is not wet, i.e. totally unaffected.

dynamic microphone

The class of microphones that generate electrical signals by the movement of a coil in a magnetic field. Dynamic microphones are rugged, relatively inexpensive, capable of very good performance and do not require external power.

dynamic range

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

echo

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

Echo and reverberation can be added in sound mixing by sending the original signal to an electronic (or electronic/acoustic) system that mimics natural echoes, and then some. The added echo is returned to the mix through additional mixer inputs.

effects device or effect processor

An external signal processor used to add reverb, delay, spatial or psychoacoustic effects to an audio signal. An effects processor may be used as an insert processor (serial) on a particular input or subgroup, or it may be used via the aux send/return system (parallel). See also echo, reverb.

EIN

Equivalent Input Noise. A specification that helps measure the “quietness” of a gain stage by deriving the equivalent input noise voltage necessary to obtain a given preamp’s output noise. Numerically, it’s the output noise at a given gain setting minus the gain. EIN is usually measured at maximum gain and typically ranges from -125 to -130 dBm.

EQ

Short for equalization.

EQ curve

A graph of the response of an equalizer, with frequency on the x (horizontal) axis and amplitude (level) on the y (vertical) axis. Equalizer types and effects are often named after the shape of the graphed response curve, such as peak, dip, bell, shelf, or notch.

equalization

Equalization (EQ) refers to purposefully changing the frequency response of a circuit, sometimes to correct for previous unequal response (hence the term, equalization), and more often to boost or cut the level at certain frequencies for sound enhancement, to remove extraneous sounds, or to create completely new and different sounds.

Bass and treble controls on your stereo are EQ; so are the units called parametrics and graphics and notch filters.

A lot of how we refer to equalization has to do with what a graph of the frequency response looks like. A flat response (no EQ) is a straight line; a peak looks like a hill, a dip is a valley, a notch is a really skinny valley, and a shelf looks like a plateau (or a shelf). The slope is the grade of the hill on the graph.

Aside from the level controls, EQs are probably the second most powerful controls on any mixer (no, the power switch doesn’t count!).

fader

Another name for an audio level control. Today, the term refers to a straight-line slide control rather than a rotary control.

family of curves

A composite graph showing on one chart several examples of possible EQ curves for a given equalizer or equalizer section.

filter

A simple equalizer designed to remove certain ranges of frequencies. A low-cut filter (also called a high-pass filter) attenuates frequencies below its cutoff frequency. There are also high-cut (low-pass) filters, bandpass filters, which cut both high and low frequencies but leave a band of frequencies in the middle untouched, and notch filters, which remove a narrow band but leave the high and low frequencies alone.

flanging

A term for an effect similar in sound to phasing. Before we had electronic delay

units, flanging was accomplished by playing two tape machines in synchronization, then delaying one slightly by rubbing a finger on the reel flange. Get it?

FOH

An acronym for Front Of House. See house and main house speakers. Nobody involved with audio ever goes to the Back of House because they never have time to drink enough beer.

frequency

The number of times an event repeats itself in a given period of time. Generally the time period for audio frequencies is one second, and frequency is measured in cycles per second, abbreviated Hz, honoring the physicist Dr. Heinrich Hertz (who did not invent the rental car). One Hz is one cycle per second. One kHz (kilohertz) is 1000 cycles per second.

The audio frequency range is generally considered to be 20 Hz to 20,000 Hz. This covers the fundamental pitch and most overtones of musical instruments.

gain

The measure of how much a circuit amplifies a signal. Gain may be stated as a ratio of input to output voltage, current or power, such as a voltage gain of 4, or a power gain of 1.5, or it can be expressed in decibels, such as a line amplifier with a gain of 10 dB.

gain stage

An amplification point in a signal path, either within a system or a single device. Overall system gain is distributed between the various gain stages.

graphic EQ

A graphic equalizer uses slide pots for its boost/cut controls, with its operating frequencies evenly spaced through the audio spectrum. In a perfect world, a line drawn through the centers of the control shafts would form a graph of the frequency response curve. Or, the positions of the slide pots give a graphic representation of

boost or cut levels across the frequency spectrum. Get it?

ground

Also called earth. Ground is defined as the point of zero voltage in a circuit or system, the reference point from which all other voltages are measured.

In electrical power systems, ground connections are used for safety purposes, to keep equipment chassis and controls at zero voltage and to provide a safe path for errant currents. This is called a *safety ground*. Maintaining a good safety ground is essential to prevent electrical shock. Follow manufacturer's suggestions and good electrical practices to ensure a safely grounded system. Never remove or disable the grounding pin on the power cord.

In sensitive electronic equipment, tiny currents and voltages riding on the ground (so it's not truly zero volts) can cause noise in the circuits and hamper operation. Often a ground separate from the power ground is used as the reference point for the electronics, isolating the sensitive electronics from the dirty power ground. This is called a *technical ground*.

Quality audio equipment is designed to maintain a good technical ground and also operate safely with a good safety ground. See the detailed section on grounding and shielding in this book for a full discourse.

ground loop

A ground loop occurs when the technical ground within an audio system is connected to the safety ground at more than one place. This forms a loop around which unwanted current can, and does flow, causing noise in the audio system. Never disable the safety ground in an attempt to solve hum problems.

Haas precedence effect

A psychoacoustic effect in which the time of arrival of a sound to the left and right ears affects our perception of direction. If a signal is presented to both ears at the same time and at the same volume, it appears to be directly in front of us. But if the signal to one ear, still at the same volume, is delayed

slightly, the sound appears to be coming from the earlier (non-delayed) side.

headroom

The difference between nominal operating level and peak clipping in an audio system. A mixer with a nominal operating level of +4 dBu and a maximum output level of +22 dBu has 18 dB of headroom. Plenty of room for surprise peaks.

Hertz

The unit of frequency, equal to 1 cycle per second. Abbreviated Hz. kHz 1000 Hz, and is usually pronounced “kay” (with “Hertz” implied) by sound professions who ask for “a little more two and a half K” when they want you to boost 2.5 kHz.

house

In Sound Reinforcement parlance, “house” refers to the systems (and even persons) responsible for the primary sound reinforcement in a given hall, building, arena or “house.” Hence we have the house mixer or house engineer, the house mix, the house mix amps, the main house speakers and so on.

Hz

Short for Hertz.

impedance

The ac resistance, capacitance, and inductance in an electrical circuit, measured in ohms. In audio circuits (and other ac circuits) the impedance in ohms can often be much different from the circuit resistance as measured by a dc ohmmeter.

Maintaining proper circuit impedance relationships is important to avoid distortion and minimize added noise. Mackie input and output impedances are set to work well with the vast majority of audio equipment.

input module

A holdover from the days when the only way that real consoles were built was in modular fashion, one channel per module. See channel strip.

insert

Noun—a place where a signal path can be broken and a processing device placed in line with the signal. It’s usually a TRS jack with one conductor being an output (send) and the other being an input (return). The jack is wired with a *normalised connection* so that with nothing plugged in, the send and return are connected together, as if it wasn’t even there. In Mackie mixers, the *insert* jacks are wired with tip as send, ring as return, and sleeve as ground.

Verb—we don’t want to go there.

knee

A knee is a sharp bend in a curve (an EQ frequency response or compressor gain curve) not unlike the sharp bend in your leg.

level

Another word for signal voltage, power, strength or volume. Audio signals are sometimes classified according to their level. Commonly used levels are: microphone level (-40 dBu or lower), instrument level (-20 to -10 dBu), and line level (-10 to +30 dBu).

line level

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

main (house) speakers

The main loudspeakers for a sound reinforcement system. These are usually the largest and loudest loudspeakers, and are usually positioned so that their sound seems to come from the area of the main stage.

mains

Short for *main* or *house speakers* in a sound reinforcement system.

master

A control affecting the final output of a bus on which one or more signals are mixed. A mixer may have several master controls,

which may be slide faders or rotary controls.

mic amp

See *mic preamp*.

mic level

The typical level of a signal from a microphone. A mic level signal (usually but not always coming from a microphone) is generally lower than -30 dBu. With a very quiet source (a pin dropping?) the signal can be -70 dBu or lower.

Some microphones, notably vintage or vintage-style condenser mics, deliver a higher signal level than this for the same sound pressure level. A “hot” mic output level isn’t necessarily a measure of the microphone’s quality, it’s just an option that the designer chose.

mic pre

Short for *mic preamp*.

mic preamp

Short for microphone preamplifier. An amplifier whose job is to bring the very low microphone level signal up to line level, or in the case of a mic preamp built into a mixer, the mixer’s internal operating level (approximately 0 dBu).

Mic preamps often have their own volume control, called a trim control, to properly set the gain for a particular source. Setting the mic preamp gain correctly with the trim control is an essential step in establishing good signal-to-noise ratio and sufficient headroom for your mix.

mixer

An electronic device used to combine various audio signals into a common output. Different from a blender, which combines various fruits into a common libation.

monaural

Long for *mono*. Literally, pertaining to or having the use of only one ear.

In the audio field, monaural describes a signal or system which carries audio information on a single channel with the intent of reproducing it from a single source. One microphone is a mono source; many microphones mixed to one channel is a mono mix; a stereo (or, to be picky, a two-channel) mix of many microphones panned left and right is a stereo mix of mono sources.

Monaural listening, and therefore mono compatibility of a stereo mix, is more important than you may realize. Most people hear television audio in mono. Most clock radios are mono.

monitor

In sound reinforcement, monitor speakers (or monitor headphones or in-the-ear monitors) are those speakers used by the performers to hear themselves. In the video and broadcast world, monitor speakers are often called foldback speakers. In recording, the monitor speakers are those used by the engineer and production staff to listen to the recording as it progresses. In zoology, the monitor lizard is the lizard that observes the production staff as the recording progresses. Keep the lizard out of the mixer.

mono

Short for *monaural*.

mult

Short for *multiple*. In audio work, a mult is a parallel connection (in a patch bay or with specially built cables or wiring) used to feed an output to more than one input. A “Y” cable is a type of mult connection. Also used a verb, as in “Why did you mult the flanger into every input in the board?”

noise

Whatever you don’t want to hear. Could be hum, buzz or hiss; could be crosstalk or digital hash or your neighbor’s stereo; could be white noise or pink noise or brown noise; or it could be your mother-in-law reliving the day she had her gallstone removed.

noise floor

The residual level of noise in any system. In a well designed mixer, the noise floor will be a quiet hiss, which is the thermal noise generated by electrons bouncing around in resistors and semiconductor junctions. The lower the noise floor and the higher the headroom, the more usable dynamic range a system has.

normal

A wiring method which electrically ties together two jacks or two poles of one jack so that in *normal* operation, there is signal flow between them. Inserting a plug breaks this connection, allowing the signal path to be modified. *Normal* wiring is common in patchbays and *insert jacks*.

PA

Acronym for Public Address. Today, people who work with PA systems like to say they're working in "sound reinforcement". See SR.

pan, pan pot

Short for panoramic potentiometer. A pan pot is used to position (or even dynamically move) a monaural sound source in a stereo mixing field by adjusting the source's volume between the left and right channels. Our brains sense stereo position by hearing this difference in loudness when the sound strikes each ear, taking into account time delay, spectrum, ambient reverberation and other cues.

patchbay

A collection of usually a large number of jacks allowing convenient access to various points in a system's interconnect wiring. A patchbay can make re-routing signals very convenient without having to fish around with cables in the back of racks or consoles. See *spaghetti*.

parametric EQ

A "fully" parametric EQ is an extremely powerful equalizer that allows smooth, continuous, and independent control of each

of the three primary EQ parameters: frequency, gain, and bandwidth. "Semi" parametric EQs allow control of fewer parameters, usually frequency and gain (i.e., they have a fixed bandwidth, but variable center frequency and gain).

peaking

The opposite of dipping, of course. A peak is an EQ curve that looks like a hill, or a peak. Peaking with an equalizer amplifies a band of frequencies.

PFL

An acronym for Pre Fade Listen. Broadcasters would call it cueing. Sound folks call it being able to solo a channel with the fader down.

phantom power

A system of providing electrical power for condenser microphones (and some electronic pickup devices) from the microphone input jack. The system is called phantom because the power is carried on standard microphone audio wiring in a way that is "invisible" to ordinary dynamic microphones. Mackie mixers use standard +48 volt dc power, switchable on or off. Most quality condenser microphones are designed to use +48 Vdc phantom power. Check the manufacturer's recommendations.

Generally, phantom power is safe to use with non-condenser microphones as well, especially dynamic microphones. However, unbalanced microphones, some electronic equipment (such as some wireless microphone receivers) and some ribbon microphones can short out the phantom power and be severely damaged. Check the manufacturer's recommendations and be careful!

phase

The time relationship between two signals, expressed in degrees around a circle. 0 and 360 degrees represents an in-phase relationship—both signals change in the same way at the same time. Anything else is out of phase.

180 degrees out of phase is a special case which, for a continuous waveform, means

that at any given time the two signals have the same amplitude but are opposite in polarity. The two legs of a differential output are 180 degrees out of phase. The *phase reverse* switch found on some mixers or mic preamps actually reverses the signal polarity.

When out-of-phase signals are mixed, there will be some cancellation at certain frequencies, the frequencies and the degree of cancellation being a function of the amount of phase shift and the relative amplitude of the signals. Attention to mic placement and careful listening will allow you to use this effect creatively.

phasing

An dynamic effect in which the phase relationship between the fundamental and overtone components of a sound is continually changing. This is done by passing the signal through an automatically sweeping filter. The effect is often simulated by mixing original signal with a delayed (1 to 10 msec) version of itself. The time of the delay is slowly varied, and the combination of the two signals results in a dramatic moving comb-filter effect. A comb filter can be found in your back pocket.

phone jack

Ever see those old telephone switchboards with hundreds of jacks and patch cords and plugs? Or the plug on the end of a headphone cable? Those are phone jacks and plugs, now used widely with musical instruments and audio equipment. A phone jack is the female connector, and we use them in 1/4" two-conductor (TS) and three-conductor (TRS) versions.

phone plug

The male counterpart to the phone jack, right above.

phono jack

Short for *RCA phono jack*.

phono plug

Short for *RCA phono plug*.

post-fader

A term used to describe an aux send (or other output) that is connected so that it is affected by the setting of the associated channel fader. Sends connected this way are typically (but not always) used for effects. A post-fader output from a mixer channel usually is also post-EQ. If pain persists, see your mixer's block diagram. Also see pre-fader.

pot, potentiometer

In electronics, a variable resistor that varies the potential, or voltage. In audio, any rotary or slide control.

pre-fader

A term used to describe an aux send (or other output) that is connected so that it is not affected by the setting of the associated channel fader. Sends connected this way are typically (but not always) used for monitors (foldback). See post-fader.

proximity effect

The property of many directional microphones to accentuate their bass response when the source-to-mic distance is small, typically three inches or less. Singers generally like this effect even more than singing in the shower.

Q

A way of stating the bandwidth of a filter or equalizer section. An EQ with a Q of .75 is broad and smooth, while a Q of 10 gives a narrow, pointed response curve. To calculate the value of Q, you must know the center frequency of the EQ section and the frequencies at which the upper and lower skirts fall 3 dB below the level of the center frequency. Q equals the center frequency divided by the difference between the upper and lower 3 dB-down frequencies. A peaking EQ centered at 10 kHz whose -3 dB points are 7.5 kHz and 12.5 kHz has a Q of 2.

RCA phono jack

Long for *RCA jack* or *phono jack*. An RCA phono jack is an inexpensive connector (female) introduced by RCA and originally used to connect phonographs to radio receivers. The phono jack was (and still is) widely used on consumer stereo equipment and video equipment but was quietly fading into obscurity in the professional and semi-professional sound world. Then phono jacks began cropping up in early project-studio multitrack recorders, which (unfortunately) gave them a new lease on life. Since so many stereo recorders are fitted with them, we decided we'd have to put a couple on our mixers for your convenience. But make no mistake: the only thing that the phono jack (or plug) has going for it is low cost.

RCA phono plug

The male counterpart to an RCA phono jack.

regeneration

Also called recirculation. A delay effect created by feeding the output of a delay back into itself to cause a delay of the delay of the delay. You can do it right on the front panel of many effects units, or you can route the delay return back into itself on your mixer. Can be a great deal of fun at parties.

Regeneration is also a fancy name for *feedback*. Feedback makes oscillators work and reduces distortion in amplifiers. Feedback in sound reinforcement systems, a form of oscillation itself, makes you popular with dogs and unpopular with musicians and audience alike.

return

A return is a mixer line input dedicated to the task of returning processed or added sound from reverb, echo and other effects devices. Depending on the internal routing of your mixer and your own inclination, you could use returns as additional line inputs, or you could route your reverb outputs to ordinary line inputs rather than the returns.

reverberation, reverb

The sound remaining in a room after the source of sound is stopped. It's what you hear in a large tiled room immediately after you've clapped your hands.

Reverberation and echo are terms that are often used interchangeably, but in audio parlance a distinction is usually made: reverberation is considered to be a diffuse, continuously smooth decay of sound, whereas echo is one or more distinct, recognizable repetitions of a word, note, phrase or sound which decreases in amplitude with every repeat.

Reverberation and echo can be added in sound mixing by sending the original sound to an electronic (or electronic/acoustic) system that mimics natural reverberation, or worse. The added reverb is returned to the blend through additional mixer inputs.

Highly reverberant rooms are called live; rooms with very little reverberation are called dead. A sound source without added reverb is dry; one with reverb or echo added is wet.

RMS

An acronym for root mean square, a conventional way to measure the effective average value of an audio signal or other ac voltage. Most ac voltmeters are calibrated to read RMS volts though on many meters that calibration is accurate only if the waveform is sinusoidal.

send

A term used to describe the output of a secondary mix of the input signals, typically used for foldback monitors, headphone monitors or effects devices. Mackie mixers call it an Aux Send.

shelving

A term used to describe the shape of an equalizer's frequency response. A shelving equalizer's response begins to rise (or fall) at some frequency and continues to fall (or rise) until it reaches the shelf frequency, at which point the response curve flattens out and remains flat to the limits of audibility. If you were to graph the response, it would

look like a shelf. Or more like a shelf than a hiking boot. See also peaking and dipping.

slap, slapback

A single-delay echo without any repeats. Also see echo.

Solo

Italian for alone. In audio mixers, a solo circuit allows the engineer to listen to individual channels, buses or other circuits singly or in combination with other soloed signals.

spaghetti

That mess of wires and cables in the back of your rack and/or console. You really *can* tame this.

SR

An acronym for Sound Reinforcement, which refers to the process (or a system for) amplifying acoustic and electronic sounds from a performance or speech so that a large audience can hear clearly. Or, in popular music, so that a large audience can be excited, stunned or even partially deafened by the tremendous amplification. The term “SR” is to “PA” (Public Address) as the term “environmental cleanup technologist” is to “garbage collector”.

stereo

Believe it or not, stereo comes from a Greek word that means solid. We use stereo or stereophony to describe the illusion of a continuous, spacious sound field that is seemingly spread around the listener by two or more related audio signals. In practice, stereo often is taken to simply mean two channels.

sweep EQ

An equalizer that allows you to “sweep” or continuously vary the frequency of one or more sections.

symmetrically balanced

See *balanced*.

tinnitus

The ringing in the ears that often results from prolonged exposure to very loud sound levels. A sound in the ears, such as buzzing, ringing, or whistling, caused by volume knob abuse!

trim

In audio mixers, the gain adjustment for the first amplification stage of the mixer. The trim control allows the mixer to accommodate the wide range of input signal levels that come from real-world sources. It is important to set the trim control correctly; its setting determines the overall noise performance in that channel of the mixer. See *mic preamp*.

TRS

Acronym for Tip-Ring-Sleeve, the three parts of a two-conductor (plus shield) phone plug. Since the plug or jack can carry two signals and a common ground, TRS connectors are often referred to as stereo or balanced plugs or jacks. Another common TRS application is for *insert* jacks, used for inserting an external processor into the signal path.

TS

Acronym for Tip-Sleeve, the two parts of a single conductor (plus shield) phone plug. TS connectors are sometimes called mono or unbalanced plugs or jacks. A 1/4” TS phone plug or jack is also called a standard phone plug or jack.

unbalanced

An electrical circuit in which the two legs of the circuit do not have the identical impedance to ground. Often one leg is also at ground potential. Unbalanced circuit connections require only two conductors (signal “hot” and ground). Unbalanced audio circuitry is less expensive to build, but under certain circumstances is more susceptible to noise pickup.

unity gain

A circuit or system that has its voltage gain adjusted to be one, or unity. A signal will leave a unity gain circuit at the same level at which it entered—no amplification, but no loss either. In Mackie mixers, unity gain is achieved by setting all variable controls to the marked and usually detented “U” setting. Mackie mixers are optimized for best headroom and noise figures with all gain stages beyond the preamp set at unity gain.

VLZ

Acronym for very low impedance, a Mackie design principle. VLZ is one of the most important reasons why inherent noise levels on Mackie mixing boards are so minuscule. Thermal noise is generated in all real world electronic components that have impedance. By keeping the impedances within the mixer circuitry low, Mackie keeps internal noise to a minimum.

volume

The sound level in an audio system. Perhaps the only thing that some bands have too much of.

VRMS

Acronym for Volts Root Mean Square. See *RMS*.

wet

A signal with added reverberation or other effect like echo, delay or chorusing.

XDR

Acronym for eXtended Dynamic Range. XDR is collection of Mackie circuit design elements which contribute to the low noise, high headroom mic preamp circuit employed on many Mackie mixers.

XLR connector

See *Cannon*.

Y-Cable

A cable with one input and two outputs, used to *mult* a source to two inputs.

Z

The electrical symbol for *impedance*.

zymurgy

The science of brewing, an important part of Mackie technology since the factory is located less than a mile from the Red Hook brewery. Besides, we needed something other than just plain “Z” to put in this A-to-Z glossary.

Appendix B: Block Diagrams and Specifications

I202-VLZ PRO Specifications

Main Mix Noise

20 Hz–20 kHz bandwidth, 1/4" Main out, channels 1–4
Trim @ unity gain, channel EQs flat, all channels assigned
to Main Mix, channels 1 and 3 Pan left, 2 and 4 Pan right.

Main Mix knob down,
channel Gain knobs down: –100 dBu
Main Mix knob unity, channel Gain knobs down: 86.5 dBu
(90 dB Signal to Noise Ratio, ref +4 dBu)
Main Mix knob @ unity, channel Gain knobs @ unity:
–84.5 dBu

Total Harmonic Distortion (THD)

1 kHz @ +14 dBu, 20 Hz–20 kHz.

Mic pre @ insert : 0.0007%

Attenuation (Crosstalk)

1 kHz relative to 0 dBu, 20 Hz–20 kHz bandwidth,
Line in, 1/4" Main Out, Trim @ unity

Main Mix knob down: –85 dBu
Channel Alt / Mute switch engaged: –84 dBu
Channel Gain knob down: –83 dBu

Frequency Response

Any input to any output

20 Hz to 60 kHz: +0 dB/–1 dB
20 Hz to 100 kHz: +0 dB/–3 dB

Equivalent Input Noise (EIN)

Mic in to Insert Send out, max gain

150 ohm termination: –129.5 dBm unweighted

Common Mode Rejection (CMR)

Mic in to Insert Send out, max gain

1 kHz: better than –90 dB

Maximum Levels

Mic in: +22 dBu
Tape in: +16 dBu
All other inputs: +22 dBu
Main Mix XLR out: +28 dBu
All other outputs: +22 dBu

Impedances

Mic in: 1.3 kilohms
Channel Insert return: 2.5 kilohms
All other inputs: 10 kilohms or greater
Tape out: 1.1 kilohms
All other outputs: 120 ohms

EQ

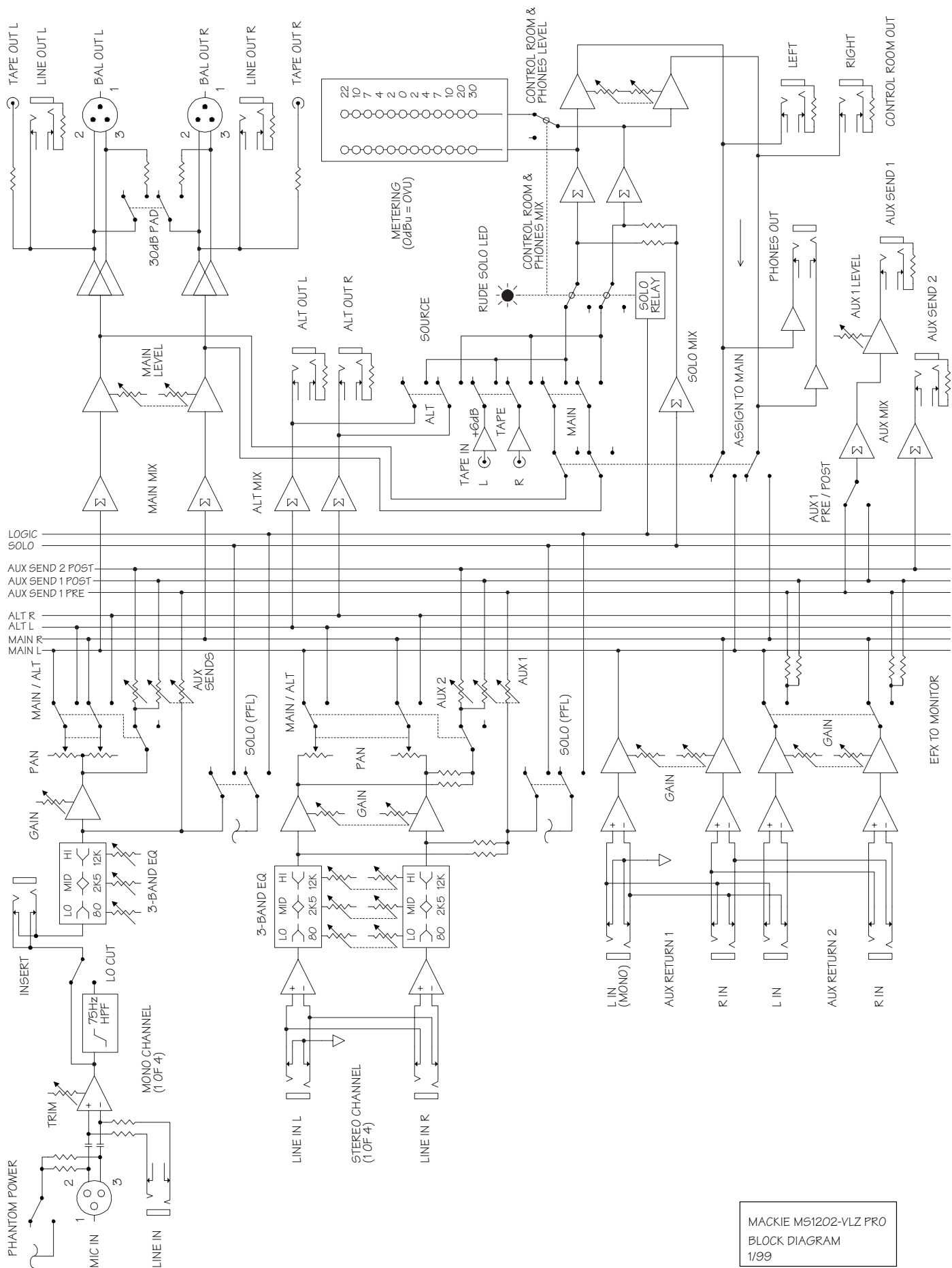
High Shelving ±15 dB @ 12 kHz
Mid Peaking ±12 dB @ 2.5 kHz
Low Shelving ±15 dB @ 80 Hz

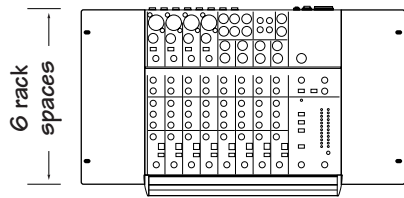
Power Consumption

120 Vac, 50/60 Hz, 25 watts

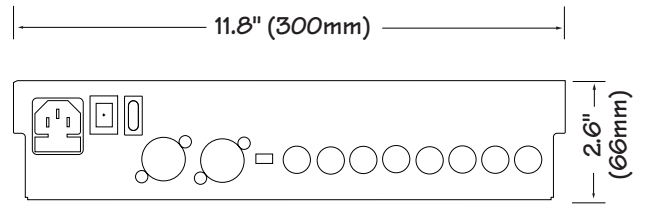
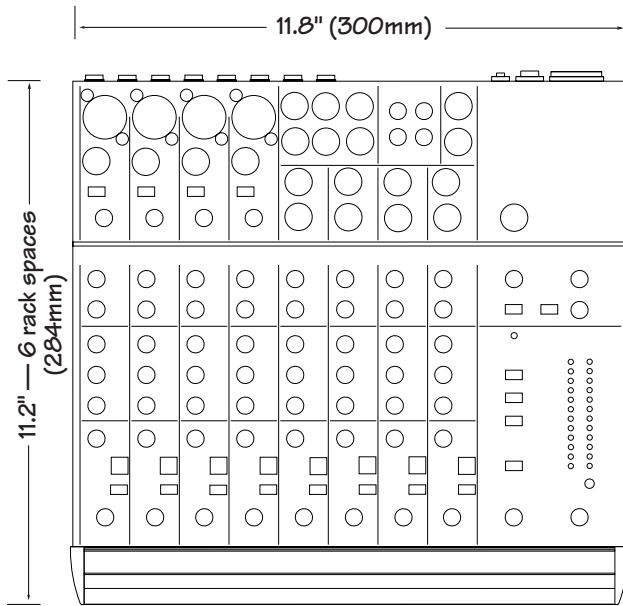
Physical Dimensions

Height 11.2" (284 mm)
Width 11.8" (300 mm)
Depth 2.6" (66 mm)
Weight 6 lbs 8 oz. (3 kg)

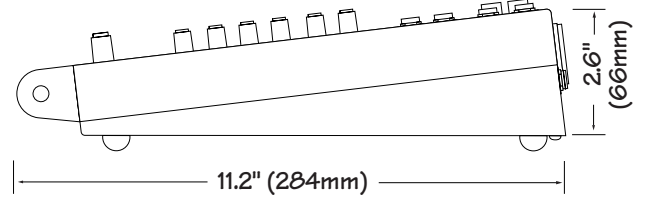




with optional rack ears (RM1202-VLZ)



1202-VLZ PRO
WEIGHT
6.5 lbs.
(3 kg)



I402-VLZ PRO

Specifications

Main Mix Noise

20 Hz–20 kHz bandwidth, 1/4" Main out, channels 1–6
Trim @ unity gain, channel EQs flat, all channels assigned
to Main Mix, channels 1 and 3 Pan left, 2 and 4 Pan right.

Main Mix fader down, channel faders down: –100.0 dBu
Main Mix fader unity, channel faders down: –86.5 dBu
(90 dB Signal-to-Noise Ratio, ref +4 dBu)
Main Mix fader @ unity,
channel faders @ unity: –84.5 dBu

Total Harmonic Distortion (THD)

1 kHz @ +14 dBu, 20 Hz–20 kHz

Mic pre @ insert : 0.0007%

Attenuation (Crosstalk)

1 kHz relative to 0 dBu, 20 Hz–20 kHz
bandwidth, Line in, 1/4" Main Out, Trim @ unity

Main fader down: –85 dBu
Channel Alt / Mute switch engaged: –84 dBu
Channel fader down: –83 dBu

Frequency Response

Mic input to any output

20 Hz to 60 kHz: +0 dB/–1 dB
20 Hz to 100 kHz: +0 dB/–3 dB

Equivalent Input Noise (EIN)

Mic in to Insert Send out, max gain

150 ohm termination: –129.5 dBm unweighted

Common Mode Rejection (CMR)

Mic in to Insert Send out, max gain

1 kHz: better than –90 dB

Maximum Levels

Mic in: +22 dBu
Tape in: +16 dBu
All other inputs: +22 dBu
Main Mix XLR out: +28 dBu
All other outputs: +22 dBu

Impedances

Mic in: 1.3 kilohms
Channel Insert return: 2.5 kilohms
All other inputs: 10 kilohms or greater
Tape out: 1.1 kilohms
All other outputs: 120 ohms

EQ

High Shelving: +/–15 dB @ 12 kHz
Mid Peaking: +/–12 dB @ 2.5 kHz
Low Shelving: +/–15 dB @ 80 Hz

Power Consumption

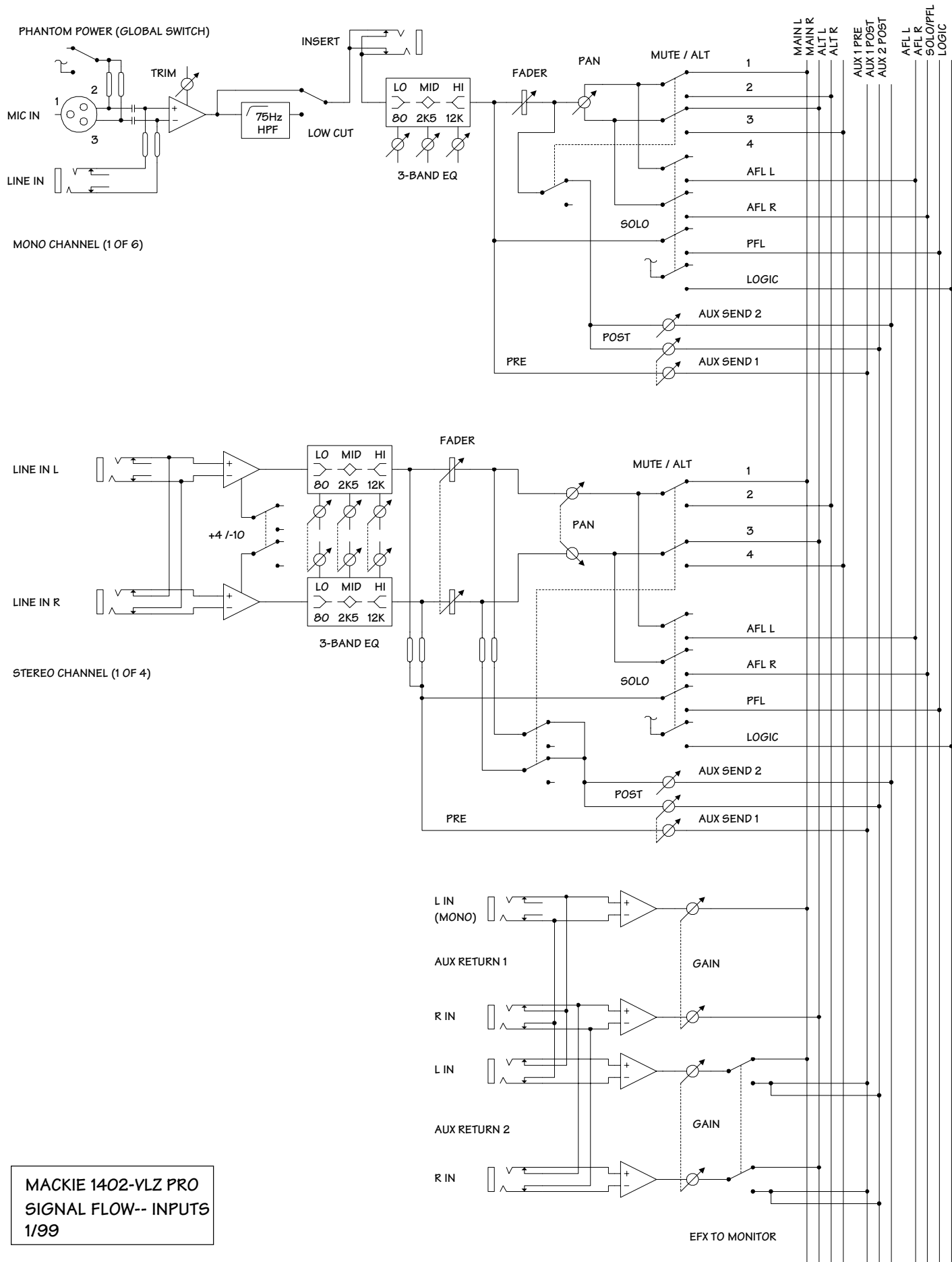
120 Vac, 50/60 Hz, 25 watts

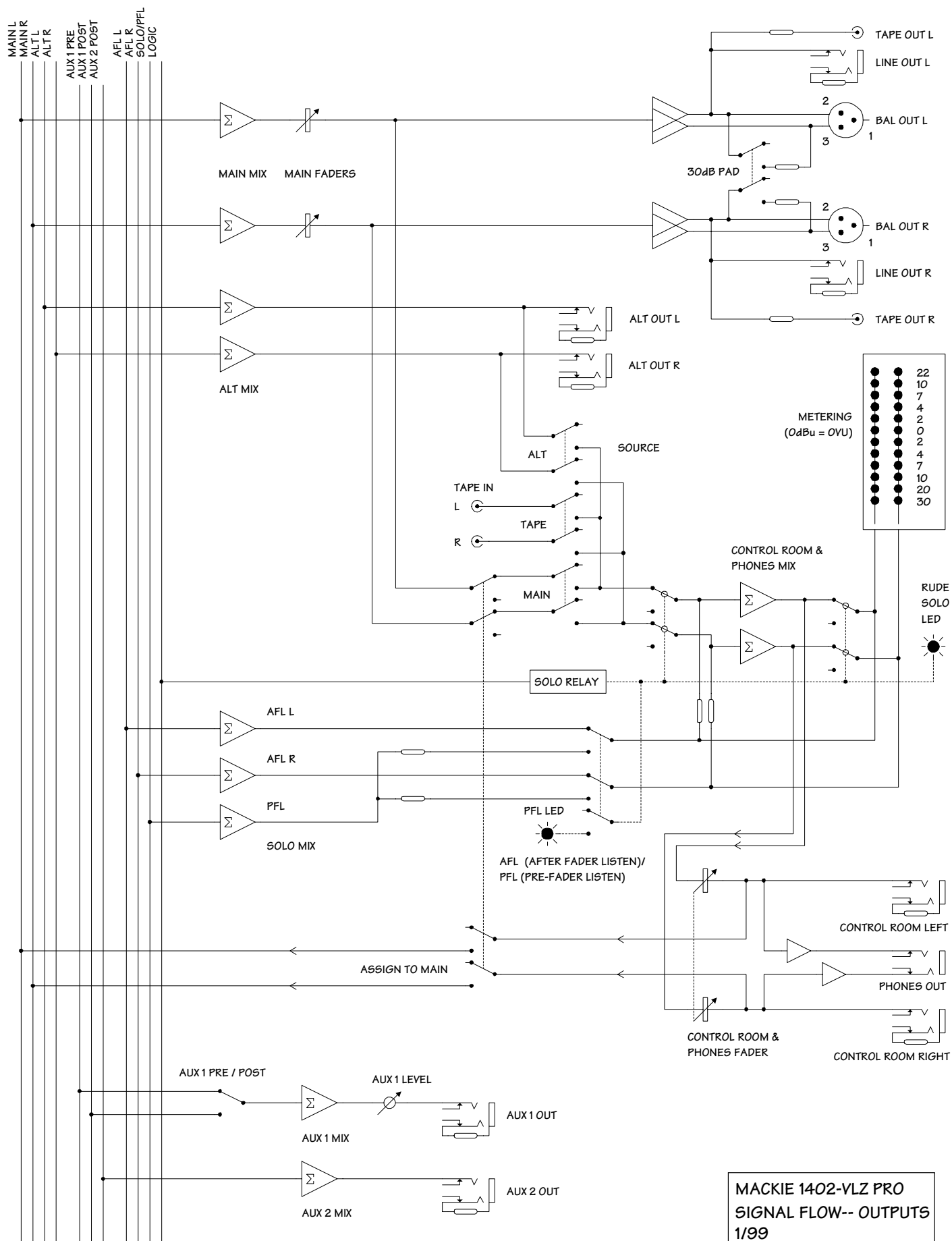
Fuse Rating

120 V: 500mA slo blo, 5 x 20 mm
220–240 V: 250mA slo blo, 5 x 20 mm

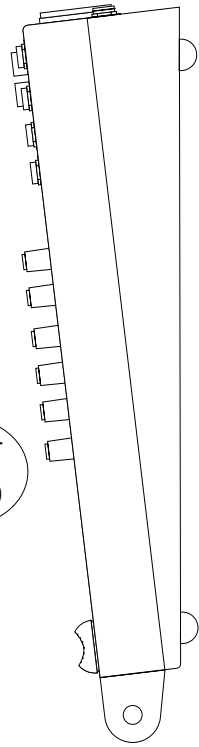
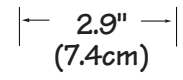
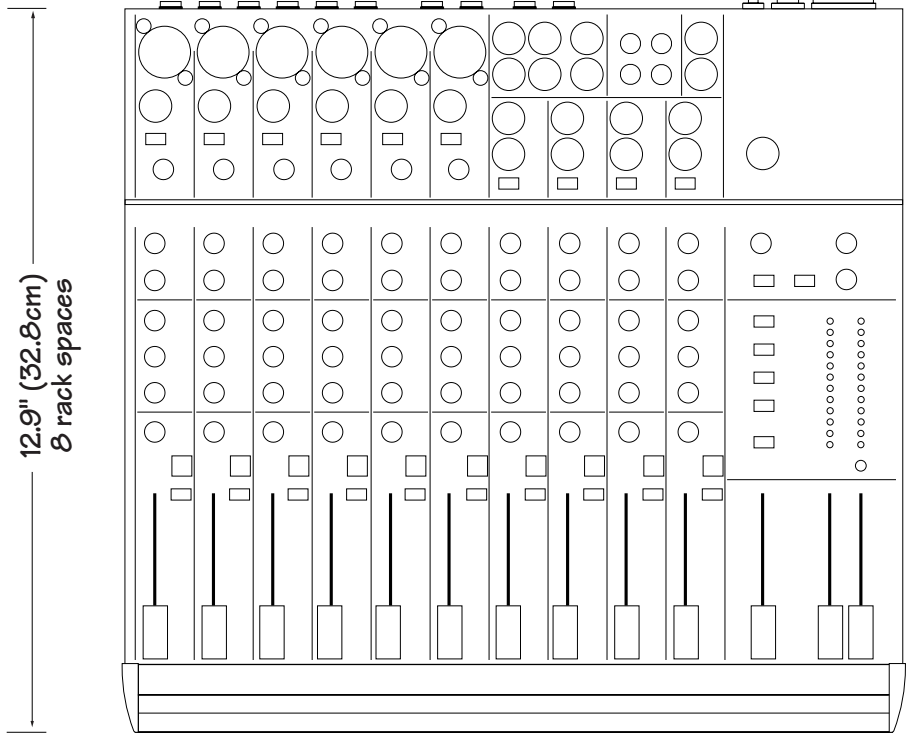
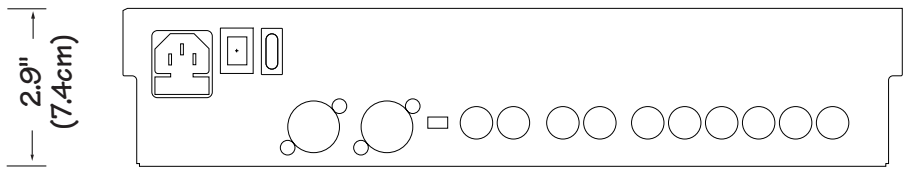
Physical Dimensions

Height: 2.9" (327 mm)
Width: 14" (356 mm)
Depth: 2.6" (74 mm)
Weight: 9 lbs 8 oz. (4.5 kg)





**MACKIE 1402-VLZ PRO
SIGNAL FLOW-- OUTPUTS
1/99**



I604-VLZ PRO

Specifications

120 V:
220–240 V:

1 A Slo Blo, 5 x 20 mm
0.5 A Slo Blo, 5 x 20 mm

Physical Dimensions

Height	17.6" (447 mm)
Width	17.3" (442 mm)
Depth	5" (127 mm)
Weight	20 lbs (9.1kg)

Main Mix Noise

20 Hz–20 kHz bandwidth, ¼" Main Out, channel Trims @ unity gain, channel EQs flat, all channels assigned to Main Mix, odd channels panned left, even channels panned right

Main Mix fader unity, channel faders down: –86.5 dBu
(90 dB Signal to Noise Ratio, ref +4 dBu)
Main Mix fader @ unity,
channel faders @ unity: –84.0 dBu

Total Harmonic Distortion (THD)

1 kHz @ +14 dBu: 20 Hz–20 kHz
Mic in to Main out: below 0.0007%

Attenuation (Crosstalk)

1 kHz relative to 0 dBu, 20 Hz–20 kHz bandwidth,
Line in, ¼" Main Out, Trim @ unity

Channel Mute switch engaged: –84 dBu
Channel Gain knob down: –84 dBu

Frequency Response

Mic input to any output

20 Hz to 60 kHz: +0 dB/–1 dB
20 Hz to 100 kHz: +0 dB/–3 dB

Equivalent Input Noise (EIN)

Mic in to Insert Send out, max gain

150 ohm termination: –129.5 dBm unweighted

Common Mode Rejection (CMR)

Mic in to Insert Send out, max gain

1 kHz: better than 90 dB

Maximum Levels

Mic in: +22 dBu
All other inputs: +22 dBu
Main Mix ¼" TRS outputs: +28 dBu
All other outputs: +22 dBu

Impedances

Mic in: 1.3 kilohms
Channel Insert return: 2.5 kilohms
All other inputs: 10 kilohms or greater
Tape out: 1.1 kilohms
All other outputs: 120 ohms

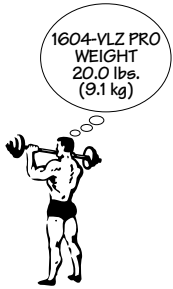
EQ

High Shelving: ±15 dB @ 12 kHz
Mid Peaking: ±15 dB, sweep 100 Hz–8 kHz
Low Shelving: ±15 dB @ 80 Hz
Low Cut Filter: 18 dB/octave, –3 dB @ 75 Hz

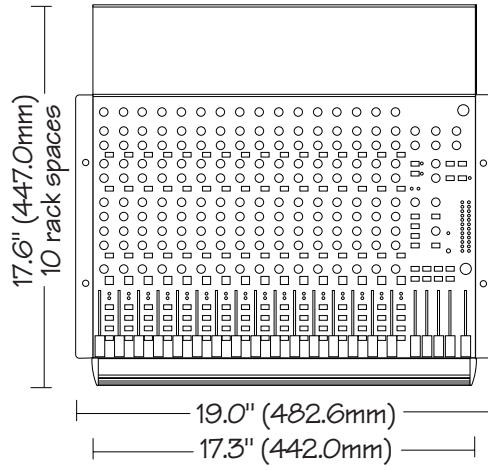
Power Consumption

120 Vac, 50/60 Hz, 50 watts

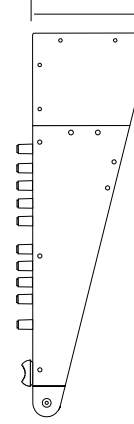
Fuse Ratings



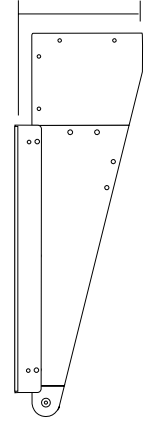
1604-VLZ-PRO with pod normal



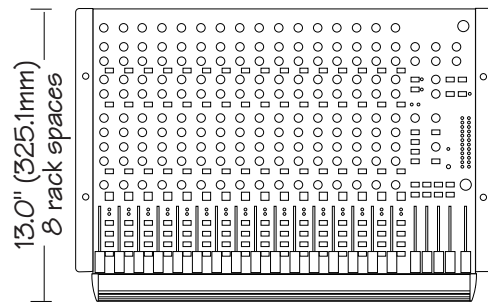
5.0" (127mm)



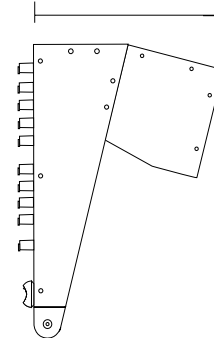
5.6" (142.2mm)



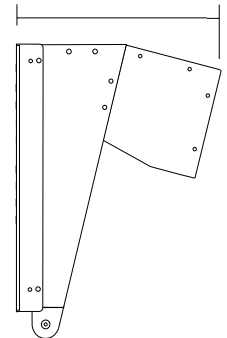
1604-VLZ PRO with pod to back



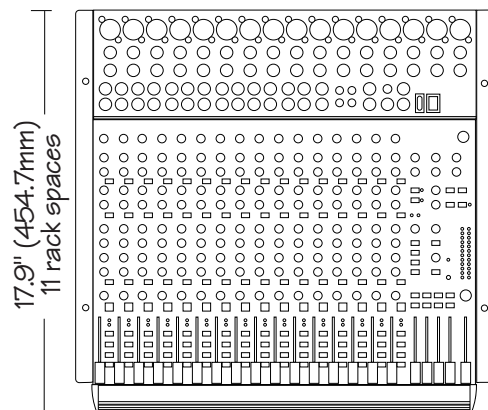
8.7" (221.0mm)



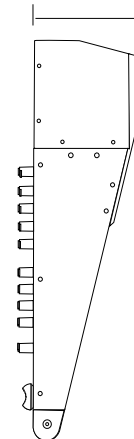
9.3" (236.2mm)



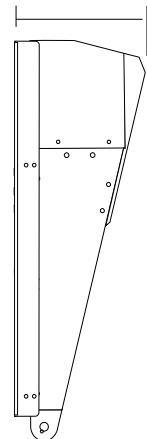
1604-VLZ PRO
with optional rotopod



6.0" (152.4mm)



6.6" (167.6mm)



1642-VLZ PRO

Specifications

Main Mix Noise

20 Hz–20 kHz bandwidth, ¼" Main Out, channel Trims @ unity gain, channel EQs flat, all channels assigned to Main Mix, odd channels panned left, even channels panned right

Main Mix fader unity, channel faders down: –86.5 dBu
(90 dB Signal to Noise Ratio, ref +4 dBu)

Main Mix fader @ unity,
channel faders @ unity: –84.0 dBu

Total Harmonic Distortion (THD)

1 kHz @ +14 dBu: 20 Hz–20 kHz
Mic in to Insert out: below 0.0007%

Attenuation (Crosstalk)

1 kHz relative to 0 dBu, 20 Hz–20 kHz bandwidth,
Line in, ¼" Main Out, Trim @ unity

Channel Mute switch engaged: –84 dBu
Channel Gain knob down: –84 dBu

Frequency Response

Mic input to any output

20 Hz to 60 kHz: +0 dB/–1 dB
20 Hz to 100 kHz: +0 dB/–3 dB

Equivalent Input Noise (EIN)

Mic in to Insert Send out, max gain

150 ohm termination: –129.5 dBm unweighted

Common Mode Rejection (CMR)

Mic in to Insert Send out, max gain

1 kHz: better than 90 dB

Maximum Levels

Mic in: +22 dBu
All other inputs: +22 dBu
Main Mix ¼" TRS and XLR outputs: +28 dBu
All other outputs: +22 dBu

Impedances

Mic in: 1.3 kilohms
Channel Insert return: 2.5 kilohms
All other inputs: 10 kilohms or greater
Tape out: 1.1 kilohms
All other outputs: 120 ohms

3 band EQ (Channels 1 - 8)

High Shelving: ±15 dB @ 12 kHz
Mid Peaking: ±15 dB, sweep 100 Hz–8 kHz
Low Shelving: ±15 dB @ 80 Hz
Low Cut Filter: 18 dB/octave, –3 dB @ 75 Hz

4-band EQ (Channels 9 - 16)

High Shelving: ±15 dB @ 12 kHz
Hi Mid Shelving: ±15 dB, 3 kHz
Lo Mid Shelving: ±15 dB, 800 Hz
Low Shelving: ±15 dB @ 80 Hz

Low Cut Filter (Ch9-12): 18 dB/octave, –3 dB @ 75 Hz

Power Consumption

120 Vac, 50/60 Hz, 40 watts

Fuse Ratings

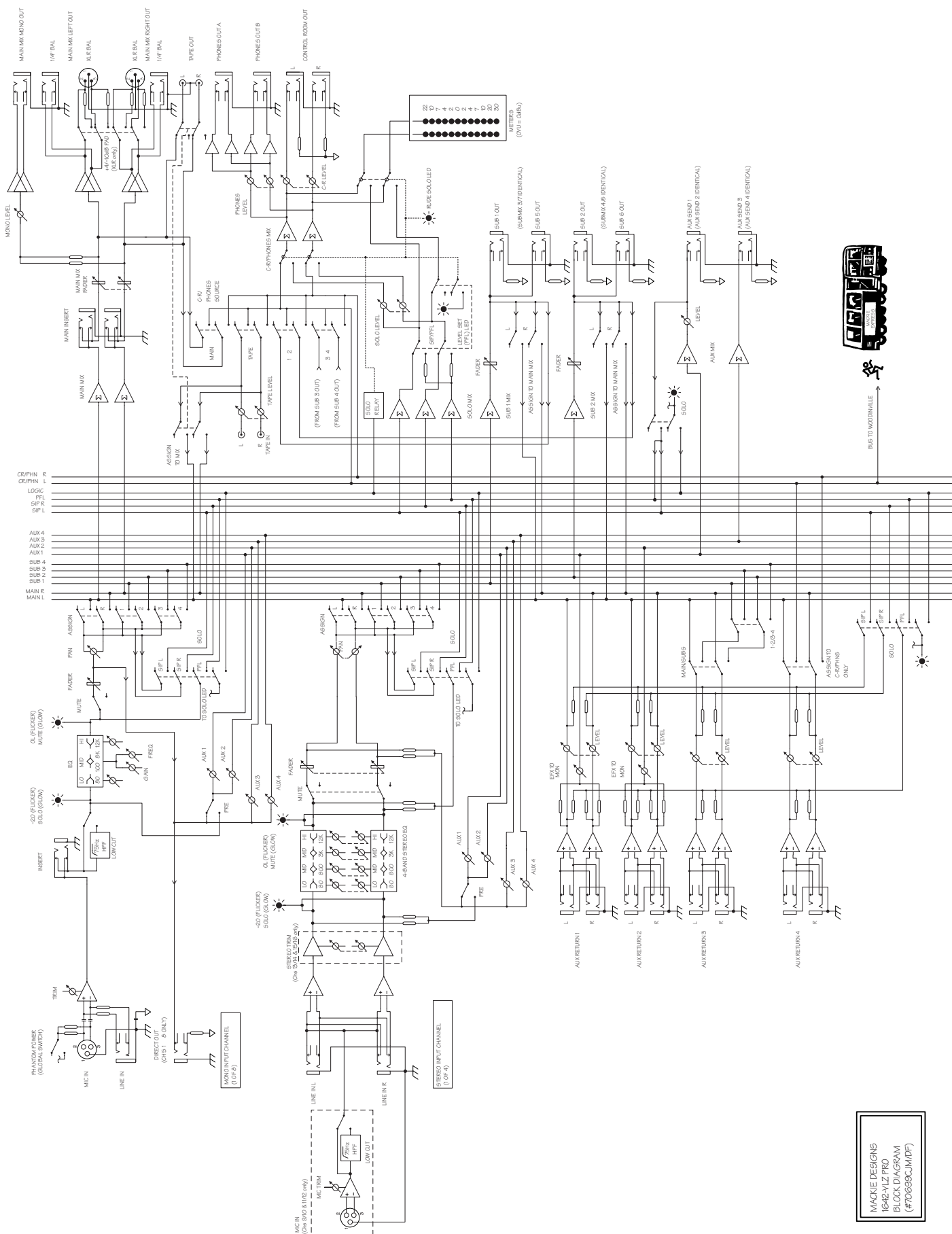
120 V: 1 A Slo Blo, 5 x 20 mm
220–240 V: 0.5 A Slo Blo, 5 x 20 mm

Physical Dimensions

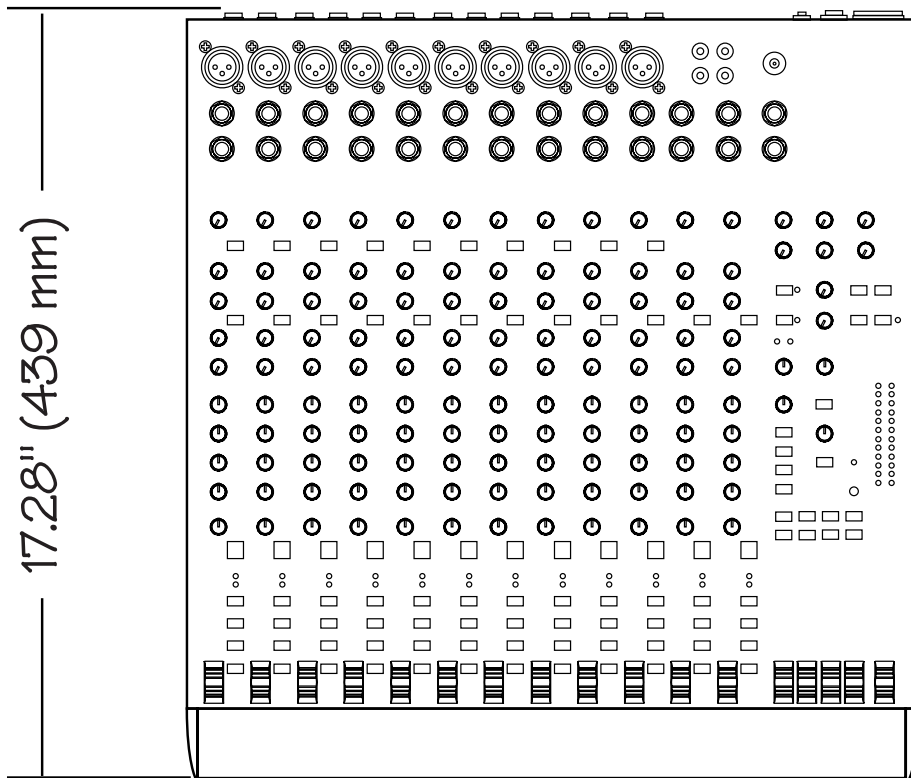
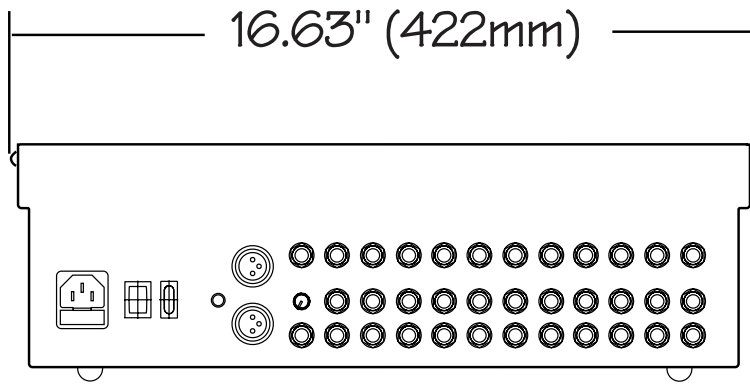
Height: 17.28" (439 mm)
Width: 16.63" (422 mm)
Depth: 5.45" (138 mm)
Weight: 18.2 lbs (8.3 kg)

Rack mounting

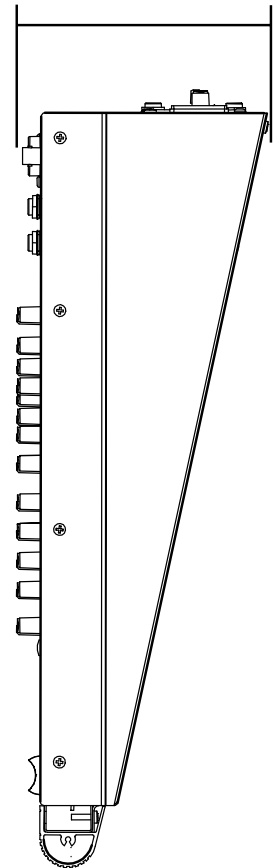
Optional rack ears are available which will allow the 1642-VLZ PRO to be mounted in a standard 19" rack. Allow ten rack spaces for the unit, and allow two empty rack spaces on top (for room to plug anything in).



MACKIE DESIGNS
 1642-V12 PRO
 BLOCK DIAGRAM
 (#70699CJ.M/D/F)



5.45" (138mm)



CFX•I2, CFX•I6, and CFX•20 Specifications

Mixer Section

Frequency Response

Mic Input to any Output (Trim at 0 dB):
+0, -1 dB, 32 Hz to 20 kHz

Distortion

THD and SMPTE IMD; 20 Hz to 20 kHz

Mic Input to Main Output:
< 0.05% @ +4 dBu output

Noise

20 Hz to 20 kHz BW (150 Ω source impedance)

Equivalent Input Noise (EIN): -127 dBu
Residual Output Noise: Main, Monitor, & Effects outputs
Channel & Master levels off
-95 dBu

Common Mode Rejection Ratio (CMRR)

60 dB @ 1 kHz, Trim @ 0 dB

Crosstalk

Adjacent Inputs or Input to Output: -90 dB @ 1 kHz
Fader Off -90 dB @ 1 kHz
Mute Switch and Break Switch Mute -80 dB @ 1 kHz

Input Level Trim Control Range

+6 to -50 dB

Phantom Power

+48 Vdc

Equalization

Low Cut: 100 Hz, -18 dB/octave

Mono Channel EQ:

High ± 15 dB @ 12 kHz
Mid ± 15 dB @ 100 Hz to 8 kHz
Low ± 15 dB @ 80 Hz

Stereo Channel EQ:

High ± 15 dB @ 12 kHz
High Mid ± 15 dB @ 3 kHz
Low Mid ± 15 dB @ 400 Hz
Low ± 15 dB @ 80 Hz

Graphic EQ (9 bands):

Q = 1.414, ISO octave centers
 ± 15 dB @ 63, 125, 250, 500, 1k, 2k, 4k, 8k, 16 kHz

Mixer Rated Output

Main, Sub, Aux, & Efx: +4 dBu
Maximum Rated Output: +20 dBu

Maximum Input Levels

Mic Input: -28 dBu, Trim @ +50 dB
+18 dBu, Trim @ +6 dB
Line Input: -8 dBu, Trim @ +30 dB
+38 dBu, Trim @ -15 dB

Insert Input, Stereo Line Input, Tape Input, and
Effects Return: +20 dBu

Maximum Voltage Gain

Mic Input to
Insert Output: 50 dB
Tape Output: 66 dB
Sub Output: 66 dB
Main Output: 76 dB
Aux Send: 71 dB
Line Input to
Insert Output: 30 dB
Tape Output: 46 dB
Sub Output: 46 dB
Main Output: 56 dB
Aux Send: 51 dB
Stereo Line Input to
Tape Output: 40 dB
Sub Output: 40 dB
Main Output: 50 dB
Aux Send: 45 dB
Tape Input to
Main Output: 30 dB
Effects Return to
Main Output: 30 dB

Input Impedance

Mic Input: 3 k Ω , balanced
Line Input: 40 k Ω , balanced
Insert Input, Stereo Line Input,
Tape Input, and Effects Returns: 10 k Ω , unbalanced

Output Impedance

Main Output, Insert Output, Tape Output,
Sub Output, and Effects Sends: 150 Ω

Digital Effects

Resolution: 16-bit, 2-channel
Number of Presets: 16

Channel Level Set LED (Sensitivity)

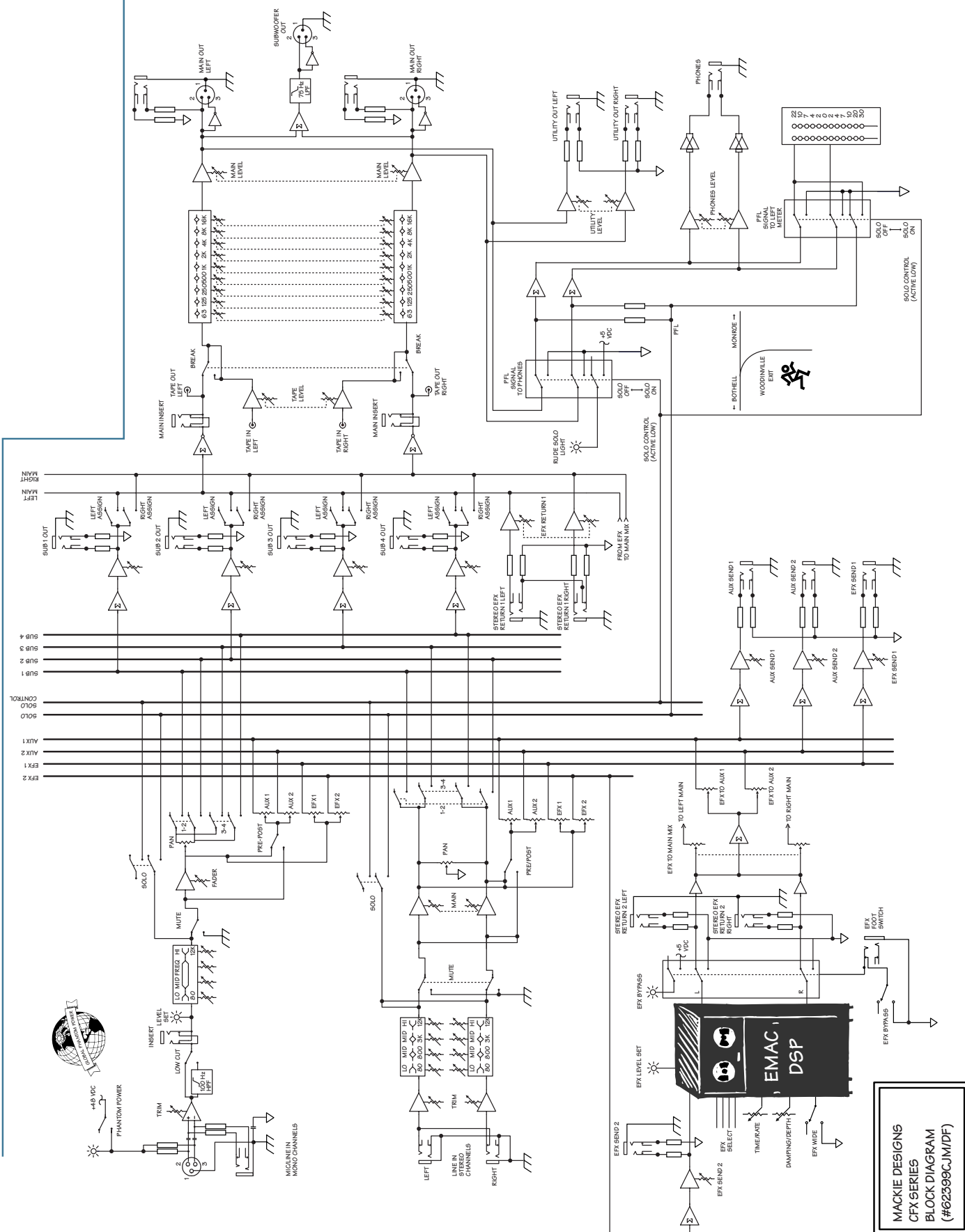
0 dBu (normal operating level)

VU Meters

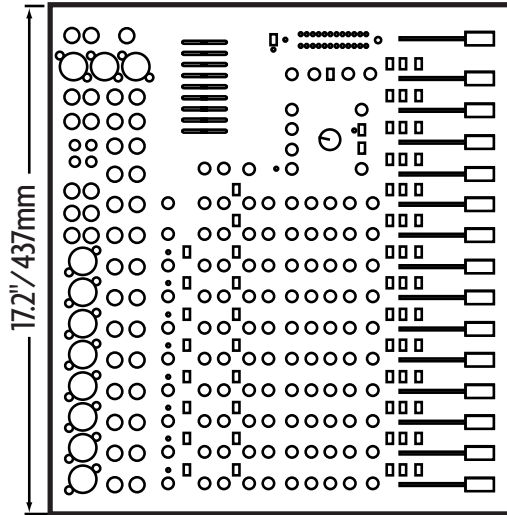
Main L/R
12 segments:
Clip, +10, +7, +4, +2, 0, -2, -4, -7, -10, -20, -30

Disclaimer

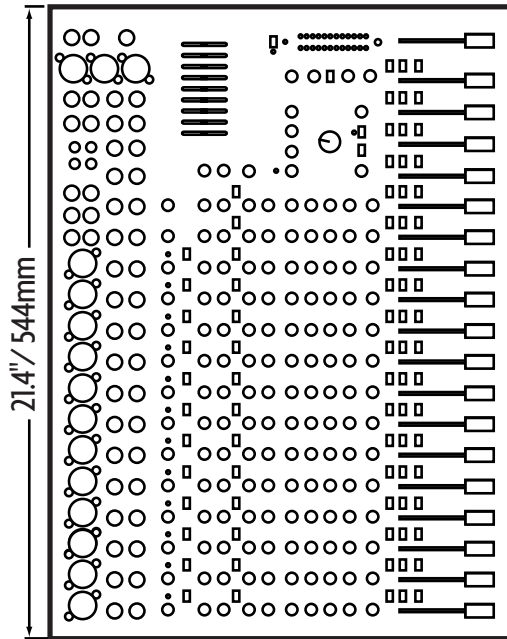
Since we are always striving to make our products better by incorporating new and improved materials, components, and manufacturing methods, we reserve the right to change these specifications at any time without notice.



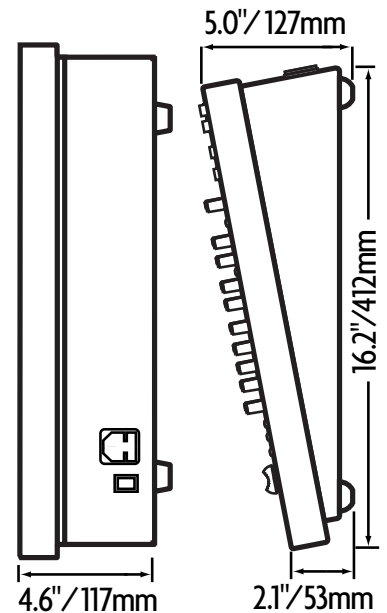
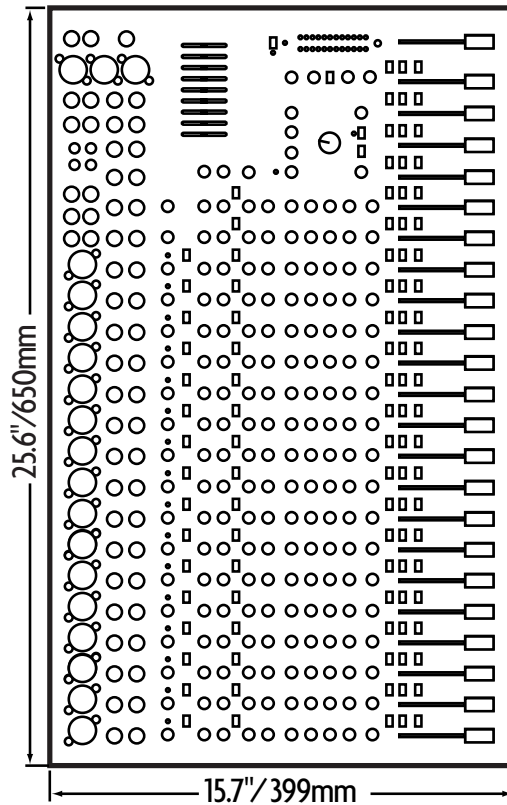
CFX•12
WEIGHT
17.8 lbs.
(8.1 kg)



CFX•16
WEIGHT
21.0 lbs.
(9.5 kg)



CFX•20
WEIGHT
24.3 lbs.
(11.0 kg)



DFX•6 and DFX•12

Specifications

Mixer Section

Frequency Response

Mic Input to any Main output (GAIN at 0 dB):
+0, -1 dB, 16 Hz to 30 kHz

Distortion

THD and SMPTE IMD; 20 Hz to 20 kHz

Mic Input to Main Output: < 0.05% @ +4 dBu output

Noise

20 Hz to 20 kHz BW (150 Ω source impedance)

Equivalent Input Noise (EIN): -127 dBu

Residual Output Noise: Main, Monitor, & Effects outputs
Channel & Master levels off
-95 dBu

Common Mode Rejection Ratio (CMRR)

60 dB @ 1 kHz, Gain @ unity

Crosstalk

Adjacent Inputs or Input to Output: -90 dB @ 1 kHz

Fader Off -90 dB @ 1 kHz

Mute Switch and Break Switch Mute -80 dB @ 1 kHz

Input Level Trim Control Range

0 to +50 dB

Phantom Power

+48 Vdc

Equalization

Low Cut: 75 Hz, -18 dB/octave

Channel EQ:

High ± 15 dB @ 12 kHz

Low ± 15 dB @ 80 Hz

Stereo Graphic EQ (5 bands):

Q = 1.414, ISO octave centers

± 12 dB @ 60, 250, 1k, 3.5 k, 12 kHz

Mixer Rated Output

Main, Aux, & EFX: +4 dBu

Maximum Rated Output: +20 dBu

Maximum Input Levels

Mic Input: -28 dBu, Gain @ +50 dB
+18 dBu, Gain @ +6 dB

Line Input: -8 dBu, Gain @ +30 dB
+38 dBu, Gain @ -15 dB

Insert Input, Stereo Line Input,

Tape Input, and Effects Return: +20 dBu

Maximum Voltage Gain

Mic Input to

Insert Output: 50 dB

Tape Output: 60 dB

Main Output: 76 dB

Aux Send: 85 dB

Line Input to

Insert Output: 30 dB

Tape Output: 40 dB

Main Output: 56 dB

Aux Send: 66 dB

Stereo Line Input to

Tape Output: 20 dB

Main Output: 40 dB

Aux Send: 50 dB

Tape Input to

Main Output: 22 dB

Effects Return to

Main Output: 22 dB

Input Impedance

Mic Input: 3 k Ω , balanced

Line Input: 40 k Ω , balanced

Insert Input, Stereo Line Input,

Tape Input, and Effects Returns: 10 k Ω , unbalanced

Output Impedance

Main Output, Insert Output, Tape Output, and

Effects Sends: 150 Ω

Digital Effects

Resolution: 16-bit, 2-channel

Number of Presets: 16

Channel Level Set LED (Sensitivity)

0 dBu (normal operating level)

VU Meters

Main L/R

12 segments:

Clip (+16), +12, +8, +4, 0, -4, -8, -12, -16, -20, -24,
and -30

Disclaimer

Since we are always striving to make our products better by incorporating new and improved materials, components, and manufacturing methods, we reserve the right to change these specifications at any time without notice.

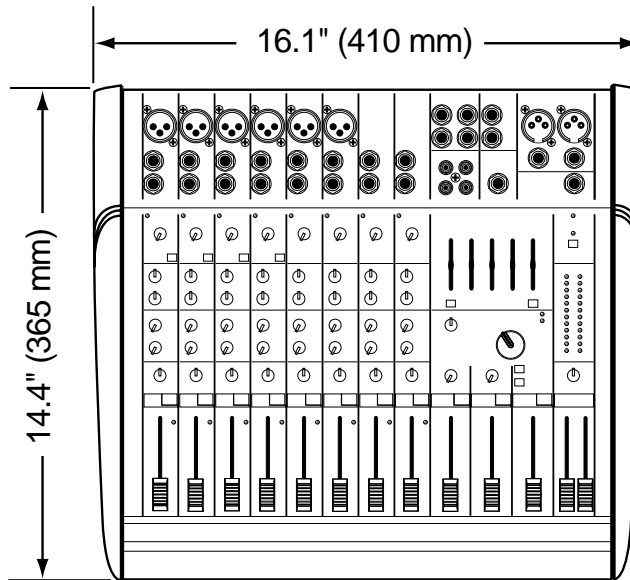
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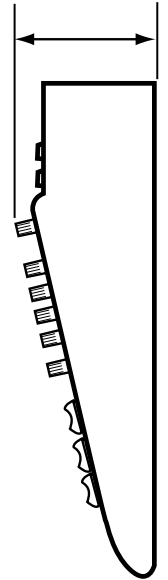
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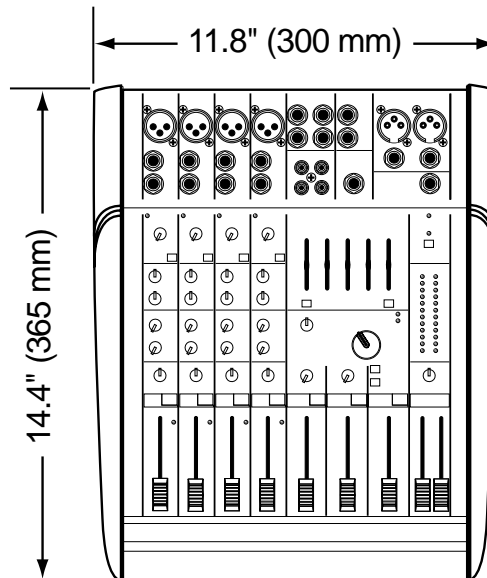
DFX•12
WEIGHT
10.0 lbs.
(4.5 kg)



4.1" (105 mm)



DFX•6
WEIGHT
8 lbs.
(3.6 kg)



PPM Series Powered Mixers (406M, 408M, 408S, 808M, 808S) Specifications

Mixer Section

Frequency Response

Mic Input to Main Mixer Output (Trim at 0 dB):	+0, -1 dB, 32 Hz to 20 kHz
	+0, -3 dB, 16 Hz to 80 kHz
Mic Input to Power Amp Output @ rated power output:	+0, -1 dB, 32 Hz to 20 kHz
	+0, -3 dB, 16 Hz to 55 kHz

Distortion

THD and SMPTE IMD; 20 Hz to 20 kHz

Mic Input to Main Mixer Output:	< 0.005% @ +4 dBu output
Mic Input to Power Amp Output:	< 0.15%, 250 mW to rated power

Common Mode Rejection Ration (CMRR)

60 dB @ 1 kHz, Trim @ 0 dB

Noise

20 Hz to 20 kHz BW (150 Ω source impedance)

Equivalent Input Noise (EIN):	-127 dBu
Residual Output Noise: Main, Monitor, & Effects outputs	
Channel & Master levels off	-95 dBu
Main Mixer Output Noise: Master @ nominal (-10 dB),	
all channels off	-92 dBu
Master & 1 input channel @ nominal	
(-10 dB & -20 dB), Trim @ 0 dB	-85 dB

Crosstalk

Adjacent Inputs or Input to Output:	-90 dB @ 1 kHz
Fader Off	-90 dB @ 1 kHz
Break Switch Mute	-80 dB @ 1 kHz

Input Level Trim Control Range

0 to -40 dB

Phantom Power

+15 Vdc

Equalization

Rumble Reduction:	75 Hz, -18 dB/octave
Channel EQ:	
High	± 15 dB @ 12 kHz
Mid	± 12 dB @ 2.5 kHz
Low	± 15 dB @ 80 Hz
Graphic EQ (9 bands):	
Q = 1.414, ISO octave centers	
± 15 dB @ 63, 125, 250, 500 1k, 2k, 4k, 8k, 16 kHz	

Main Mixer Section Rated Output

Main Mixer, Monitor, & Effects:	+4 dBu
Maximum Main Mixer Section Output:	+20 dBu

Maximum Input Levels

Mic Input:	-20 dBu, Trim @ 0 dB (HI)
	+20 dBu, Trim @ -40 dB (LOW)
Line Input:	0 dBu, Trim @ 0 dB (HI)
	+40 dBu, Trim @ 40 dB (LOW)
Insert Input:	+20 dBu
Stereo Line Input:	+20 dBu
Tape Input:	+20 dBu
Effects Return:	+20 dBu
Power Amp In:	+22 dBu

Input Sensitivity

Minimum Input Level to produce +4 dBu @	
Main Mixer Output	
Mic Input:	-68 dBu
Insert Input:	-28 dBu
Line Input:	-48 dBu
Stereo Line Input:	-28 dBu
Tape Input:	-18 dBu
Effects Return:	-18 dBu

Maximum Voltage Gain

Mic Input to	
Insert Output:	40 dB
Tape Output:	60 dB
Main Mixer Output:	72 dB
Line Input to	
Insert Output:	20 dB
Tape Output:	20 dB
Main Mixer Output:	52 dB
Stereo Line Input to	
Tape Output:	20 dB
Main Mixer Output:	32 dB
Tape Input to	
Tape Output:	10 dB
Main Mixer Output:	22 dB
Effects Return to	
Main Mixer Output:	22 dB
Monitor Output:	22 dB

Input Impedance

Mic Input:	3 k Ω , balanced
Insert Input:	10 k Ω , unbalanced
Line Input:	40 k Ω , balanced
Stereo Line Input:	10 k Ω , unbalanced
Tape Input:	10 k Ω , unbalanced
Effect Return:	10 k Ω , unbalanced
Power Amp In:	10 k Ω , unbalanced

Output Impedance

Main Mixer Output:	150 Ω
Insert Output:	150 Ω
Tape Output:	150 Ω
Monitor Output:	150 Ω
Effects Send:	150 Ω
Power Amp Out:	0.032 Ω @ 1 kHz

Digital Effects

Resolution:	16-bit, 2-channel
Sample Rate:	31.25 kHz
Bandwidth:	15.6 kHz

VU Meters

Main and Monitor

8 segments: Clip, +5, 0, -5, -10, -15, -20, -30

Power Amplifier Section

Maximum Power at 1% THD, Midband, Both Channels Driven

406M, 408M, 408S

250 watts per channel into 2 Ω

200 watts per channel into 4 Ω

125 watts per channel into 8 Ω

808M, 808S

600 watts per channel into 2 Ω

450 watts per channel into 4 Ω

300 watts per channel into 8 Ω

Continuous Sine Wave Average Output Power, Both Channels Driven (Rated Power)

406M, 408M, 408S

180 watts per channel into 4 Ω from 40 Hz to 20 kHz,
with no more than 0.15% THD

110 watts per channel into 8 Ω from 40 Hz to 20 kHz,
with no more than 0.10% THD

808M, 808S

340 watts per channel into 4 Ω from 40 Hz to 20 kHz,
with no more than 0.15% THD

240 watts per channel into 8 Ω from 40 Hz to 20 kHz,
with no more than 0.10% THD

Power Bandwidth

< 10 Hz to 30 kHz (+0, -1 dB) @ rated power into 4 Ω

Frequency Response

< 10 Hz to 30 kHz (+0, -1 dB)

< 10 Hz to 55 kHz (+0, -3 dB)

Distortion

THD, SMPTE IMD: < 0.10% @ 8 Ω

< 0.15% @ 4 Ω

Signal-to-Noise Ratio

> 105 dB below rated power into 8 Ω

Channel Separation

> 75 dB @ 1 kHz

Damping Factor

> 250 @ 1 kHz

Amp Input Impedance

10 k Ω unbalanced, 20 k Ω balanced

Input Sensitivity

406M, 408M, 408S

1.35 volts (+4.8 dBu) for rated power into 4 ohms

808M, 808S

1.76 volts (+7.1 dBu) for rated power into 4 ohms

Gain (Amp In to Speaker Out)

26.4 dB (21 V/V)

Maximum Input Level

9.75 volts (+22 dBu)

Rise Time

406M, 408M, 408S

< 5 μ s

808M, 808S

< 6.2 μ s

Slew Rate

406M, 408M, 408S

> 40 V/ μ s

808M, 808S

> 50 V/ μ s

Load Angle

8(\pm jx) time independent at 8 Ω

4(1 \pm jx) time dependent at 4 Ω

High Frequency Overload and Latching

No latch up at any frequency or level.

High Frequency Stability

Unconditionally stable, driving any reactive or capacitive load

Turn On Delay

3 seconds

AC Power Requirements

United States: 120 Vac, 60 Hz

Europe: 240 Vac, 50 Hz

Japan: 100 Vac, 50/60 Hz

Korea: 220 Vac, 60 Hz

(Capable of operation from 75% to 110% of rated line voltage)

Physical

Height: 11.7 inches (297 mm)

Width: 20.5 inches (521 mm)

Overall Depth: 13 inches (330 mm)

Weight:

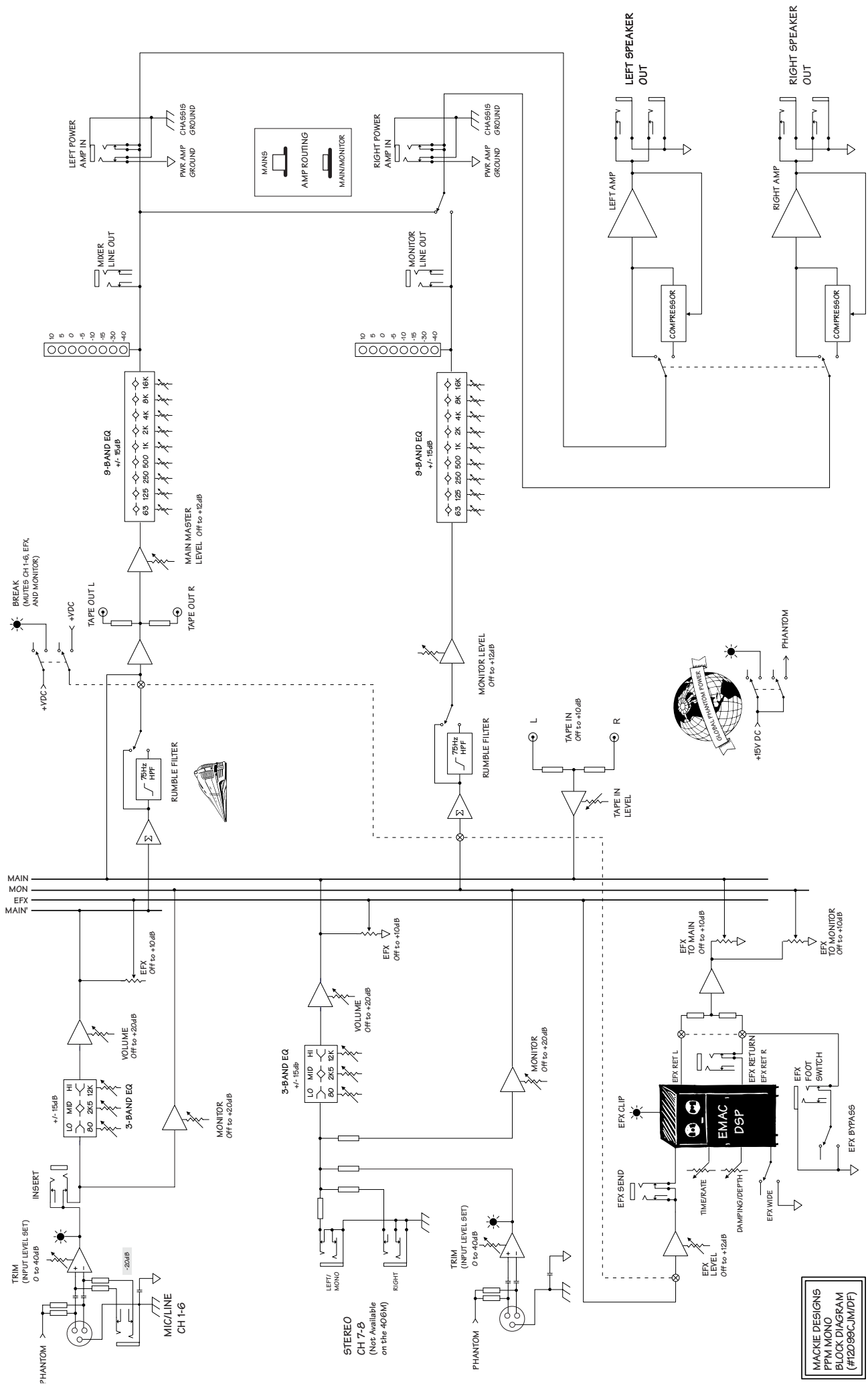
406M, 408M, 408S 32 pounds (14.5 kg)

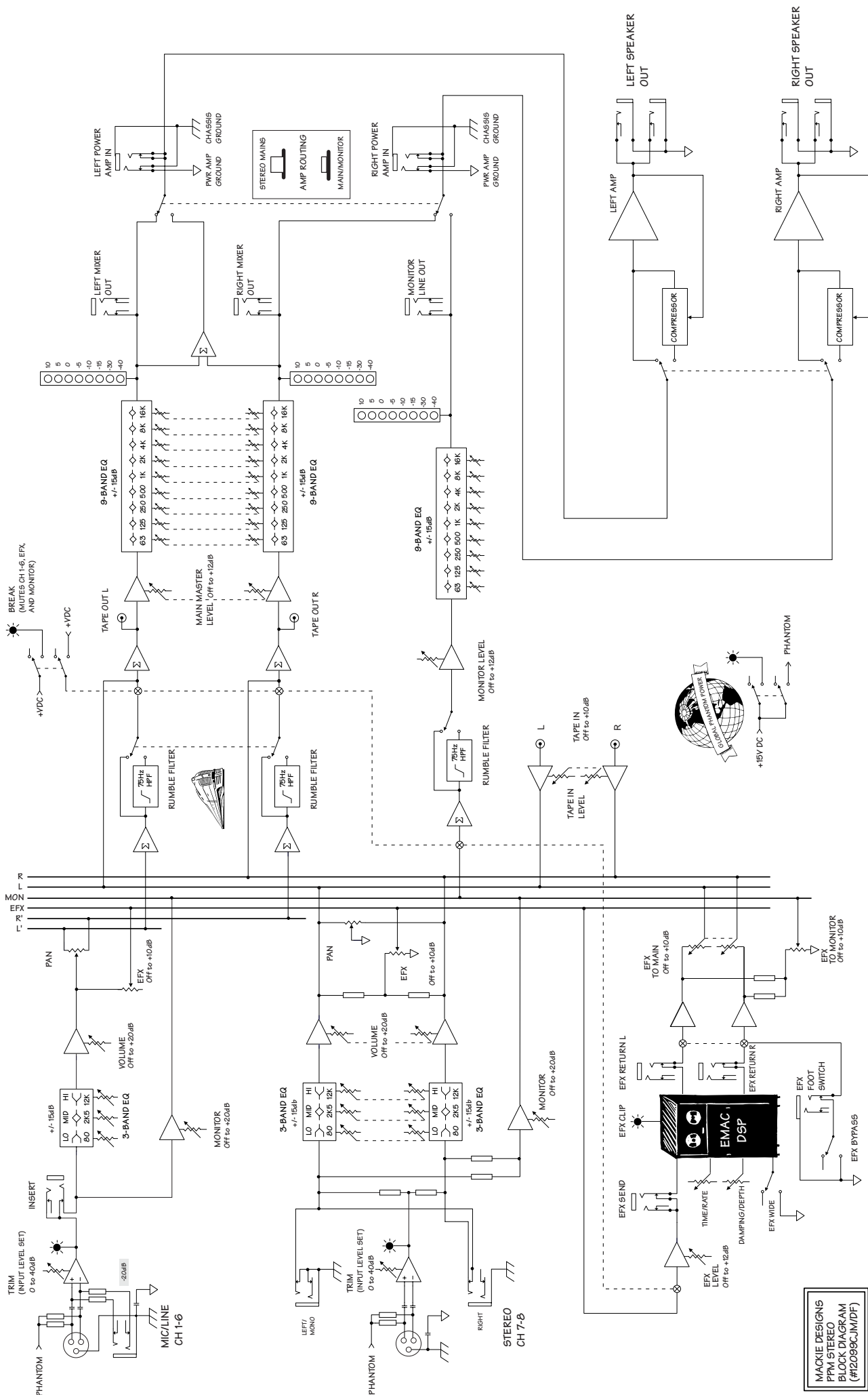
808M, 808S 36 pounds (16.3 kg)

Disclaimer

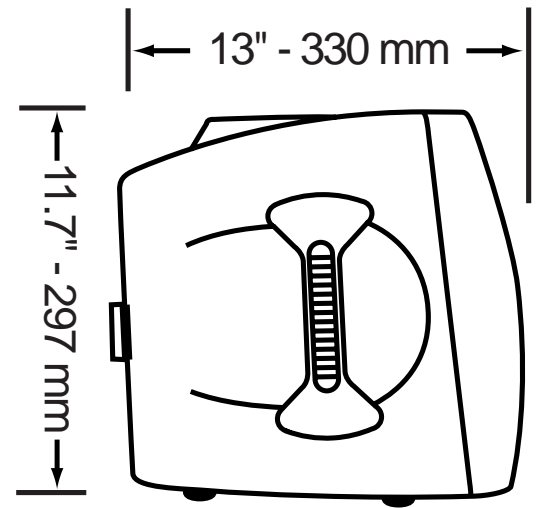
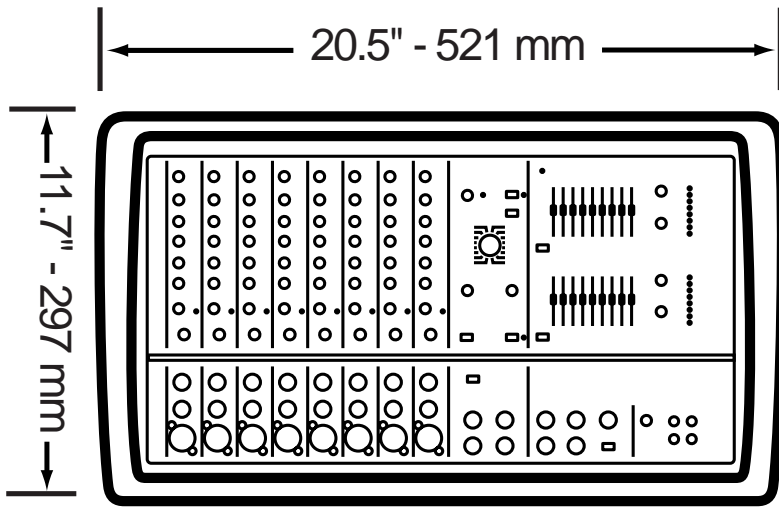
Since we are always striving to make our products better by incorporating new and improved materials, components, and manufacturing methods, we reserve the right to change these specifications at any time without notice.

This product is sold by weight, not by volume.





MAKIE DESIGNS
PPM STEREO
BLOCK DIAGRAM
(#12098CJM/DF)



24•4-VLZ PRO and 32•4-VLZ PRO Specifications

Since we are always striving to make our products better by incorporating new and improved materials, components, and manufacturing methods, we reserve the right to change these and other specifications at any time without notice.

Noise (20 Hz to 20 kHz bandwidth, Line inputs to Main L/R outputs, all channels assigned, panned L/R):

Master fader down, Ch. gains down	−94.7 dBu
Master fader @ unity, Ch. gains down	−87.4 dBu
Master fader @ unity, Ch. gains @ unity	−83.5 dBu

Total Harmonic Distortion

(1KHz @ +14 dBu 20 Hz–20 kHz):

Mic input to insert output	<0.0007%
Other outputs	< 0.004%

Crosstalk (1 kHz @ 0 dBu, 20 Hz to 20Khz bandwidth, channel in to Main Left outputs):

Channel fader down, channels at Unity	−89.5 dB
Channel muted, channels 2–16 at Unity	−88.7 dB

Frequency Response (any input to any output):

20 Hz to 50KHz	+0/−1 dB
20 Hz to 100KHz	+0/−3 dB

Maximum Levels

Mic preamp input	+22 dBu
All other inputs	+22 dBu
Balanced XLR outputs	+28 dBu
All other outputs	+22 dBu

Impedances

Mic preamp input	1.5 k Ω
All other inputs	>10 k Ω
All outputs	120 Ω

Equalization

Lo EQ Shelving 80 Hz	±15 dB
Mid EQ _(mono ch) Peak 100–8 kHz	±15 dB
Hi EQ Shelving 12 kHz	±15 dB

Microphone Preamp

E.I.N. (150 Ω terminated, max gain):	−129.5 dBm
---	------------

Power Requirements

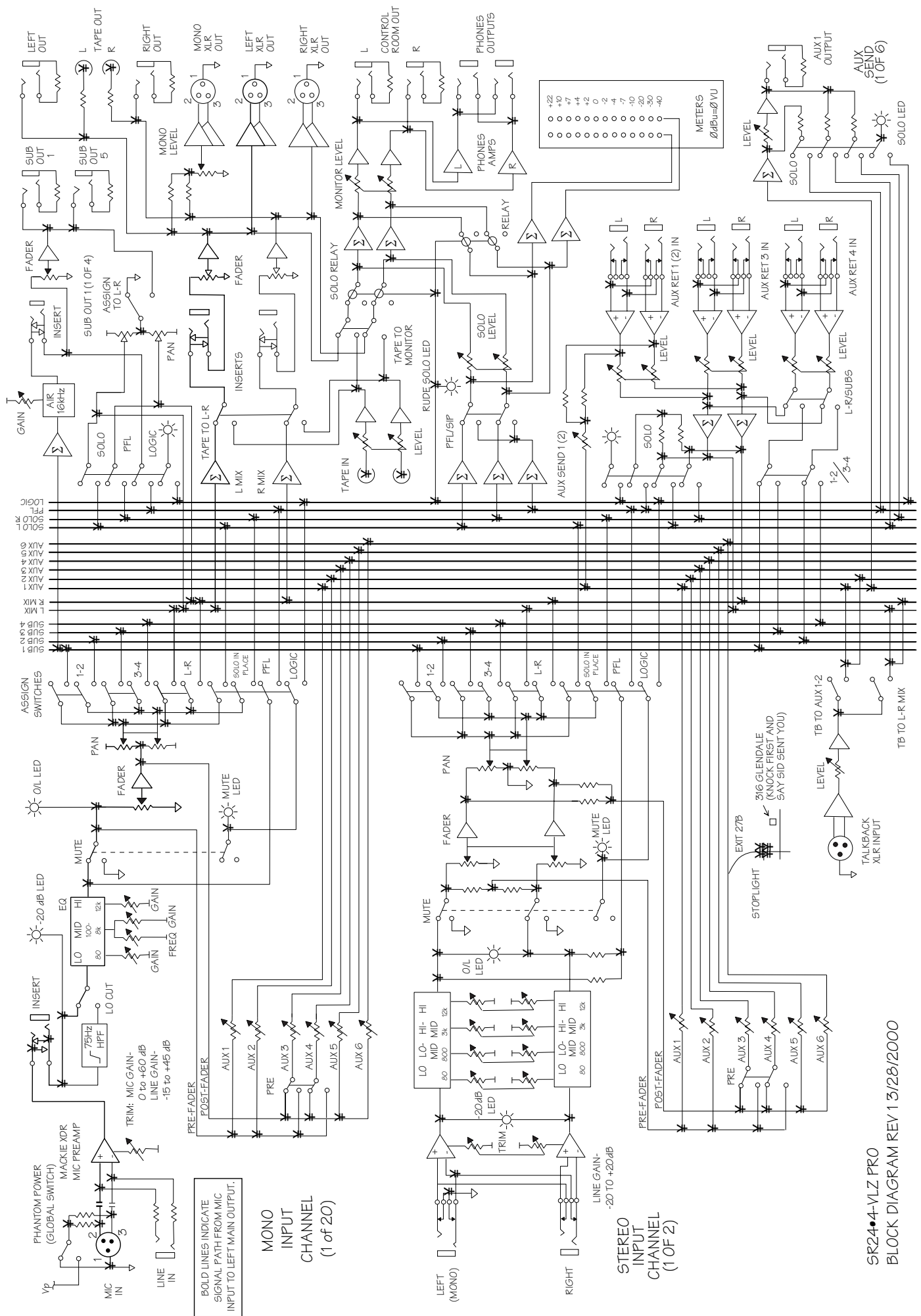
60 watts

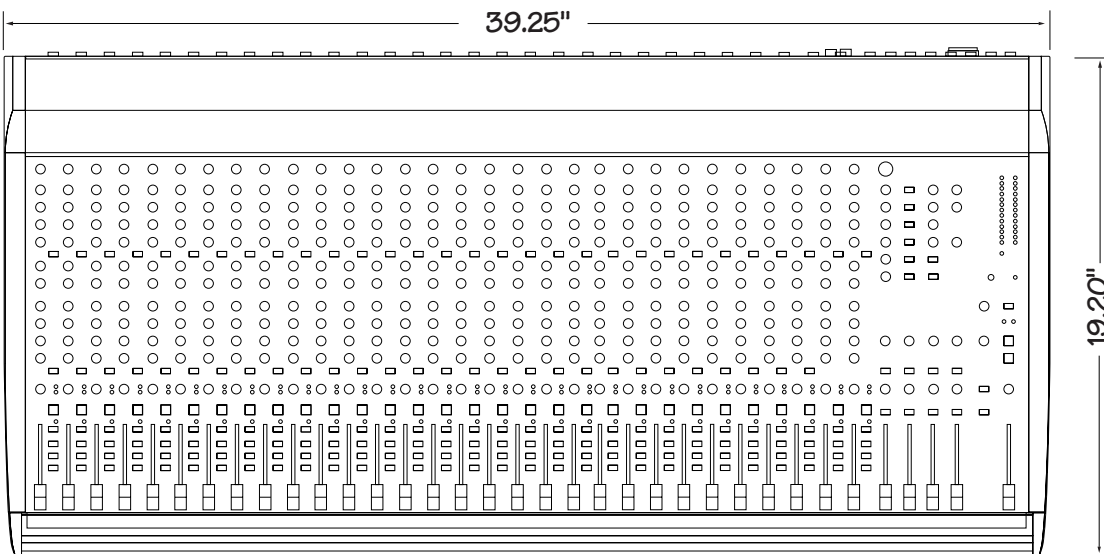
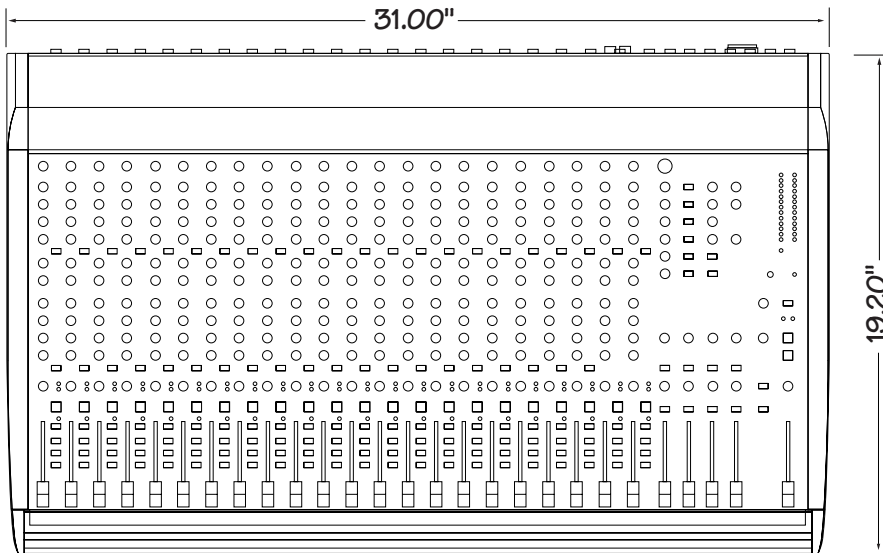
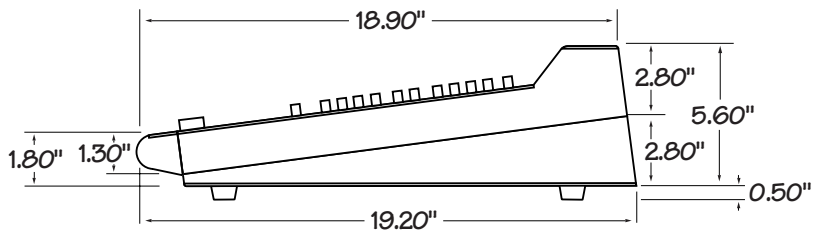
Physical Dimensions (24•4-VLZ PRO)

Height	19.2" (488 mm)
Width	31" (787 mm)
Depth	5.6" (142 mm)
Weight	31 lbs (14 kg)

Physical Dimensions (32•4-VLZ PRO)

Height	19.2" (488 mm)
Width	39.25" (997 mm)
Depth	5.6" (142 mm)
Weight	41 lbs (19 kg)





8•Bus16•8, 24•8, and 32•8 Specifications

Noise (Measured 20 Hz to 20 kHz bandwidth, Tape Returns selected, no EQ, Channel Pans alternating L/R, L/R. “Faders up” refers to Unity gain, 0 dBu position.)

Main L/R Output Noise

Master fader down	−101 dBu
Master fader up/no ch.’s assigned	−95 dBu
Master fader up/24 chs. assigned	−90 dBu
Master fader up/24 chs. assigned, ch. faders up	−86 dBu
+4 dB operating level S/N ratio	−90 dBu

Submaster Output Noise

Master fader down	−99 dBu
Master fader up/no ch.’s assigned	−96 dBu
Master fader up/24 chs. assigned	−90 dBu
Master fader up/24 chs. assigned, ch. faders up	−86 dBu
+4 dB operating level S/N ratio	90 dB

Total Harmonic Distortion (1 kHz @14 dBu measured 20 Hz-20 kHz, mic input, 1 channel assigned)

Direct output	0.0013% typical
L/R Mix output	0.0014% typical
Submaster output	0.0015% typical

Crosstalk. (1 kHz measured relative to 0 dBu, measured 20 Hz to 20 kHz.)

Line In to Adjacent Channel	−91 dBu
L/R mix output w/channel down	−95 dBu
Submaster output w/channel down	−96 dBu
L/R mix output w/channel muted	−95 dBu
Submaster output w/channel muted	−96 dBu
L/R mix output w/channel unassigned	−91 dB
Submaster output w/channel unassigned	−95 dBu
L/R mix output w/adjacent channel assigned	−92 dBu
Submaster output w/adjacent channel assigned	−94 dBu
L/R mix output pan pot attenuation	−87 dBu
Submaster output pan pot attenuation	−87 dBu

Frequency Response

+0 dB/−1 dB, any input to any output 20 Hz to 60 kHz
+0 dB/−3 dB, any input to any output, 10 Hz to 120 kHz

E.I.N.

Mic input (150 Ω termination, 20 Hz-20 kHz) −129.5 dBm

Common Mode Rejection Ration (CMRR)

Mic input, max gain @1 kHz	−83 dBu
Line input, minimum gain @1 kHz	−45 dBu
Tape input, no gain @1 kHz	−45 dBu

Maximum Levels

Mic input	+14 dBu
all other inputs	+22 dBu
L/R Mix balanced output	+28 dBu
all other outputs	+22 dBu

Impedances

Microphone input	1.5 k Ω
Channel insert return	2.5 k Ω
All other inputs	10 k Ω or greater
All outputs	120 Ω

Equalization

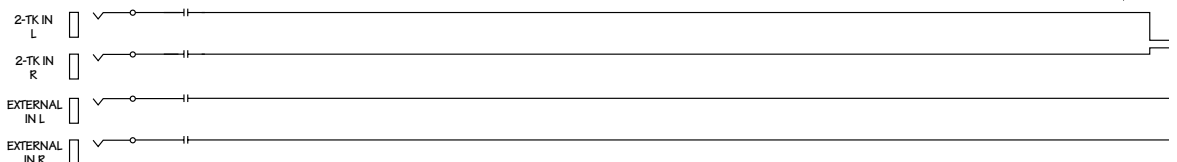
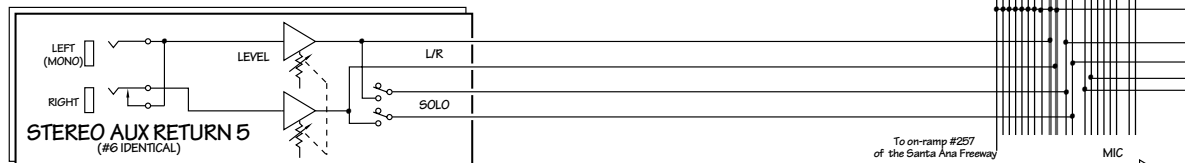
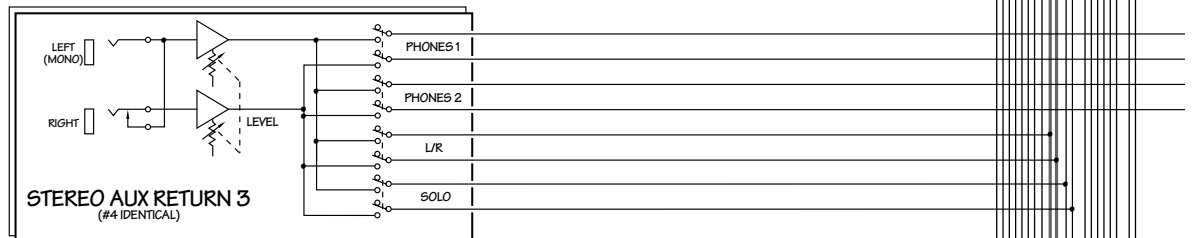
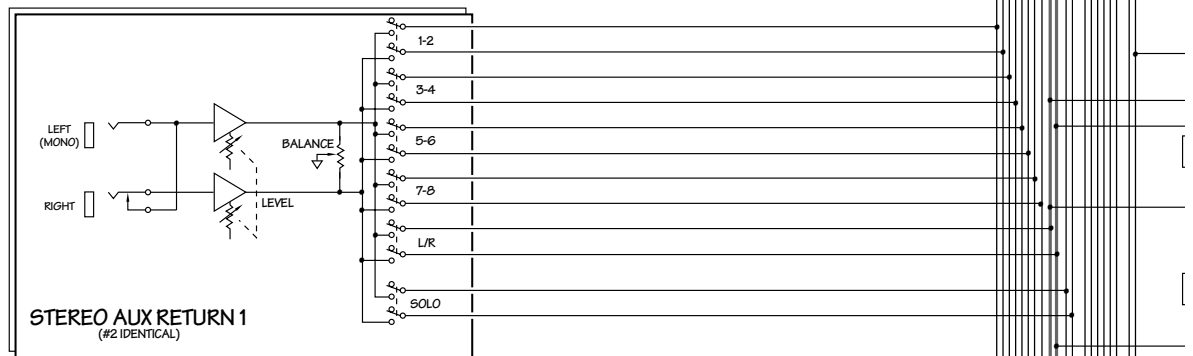
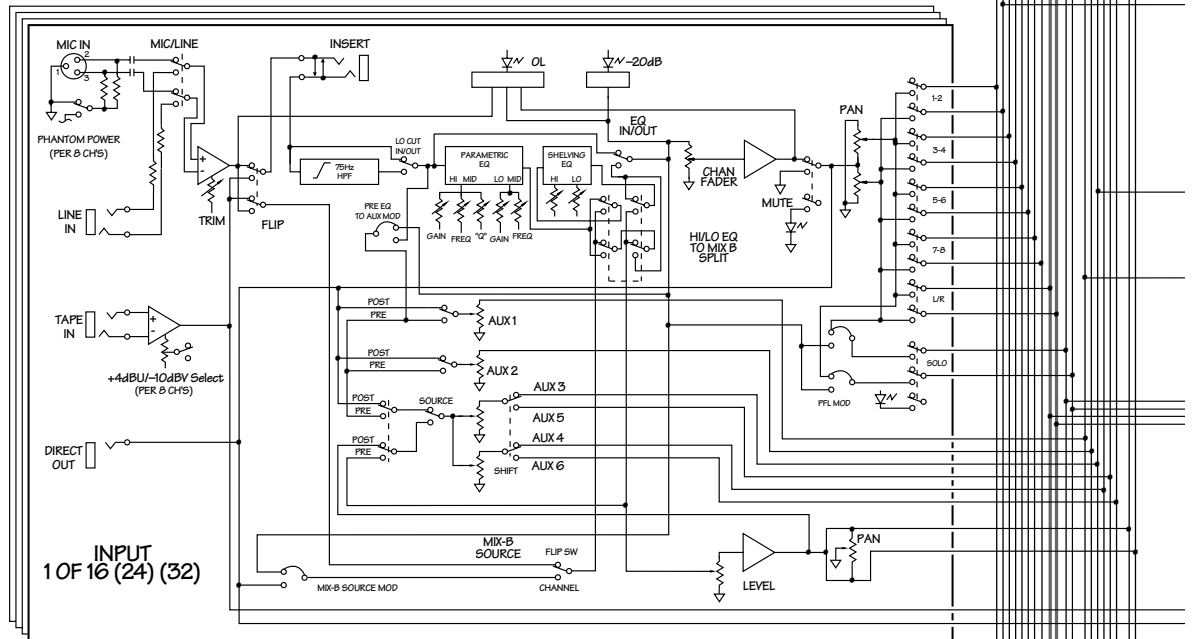
Hi Mid, full parametric	+/-15 dB freq. sweep from 500 Hz-18 kHz
Bandwidth (Q) variable from 1/12 octave to 3 octaves	
Lo Mid, swept	45 Hz-3 kHz +/-15 dB
Hi, shelving	12 kHz +/-15 dB
Lo, shelving	80 Hz +/-15 dB
Lo Cut (HPF)	75 Hz,18 dB/octave (Tchebechev)

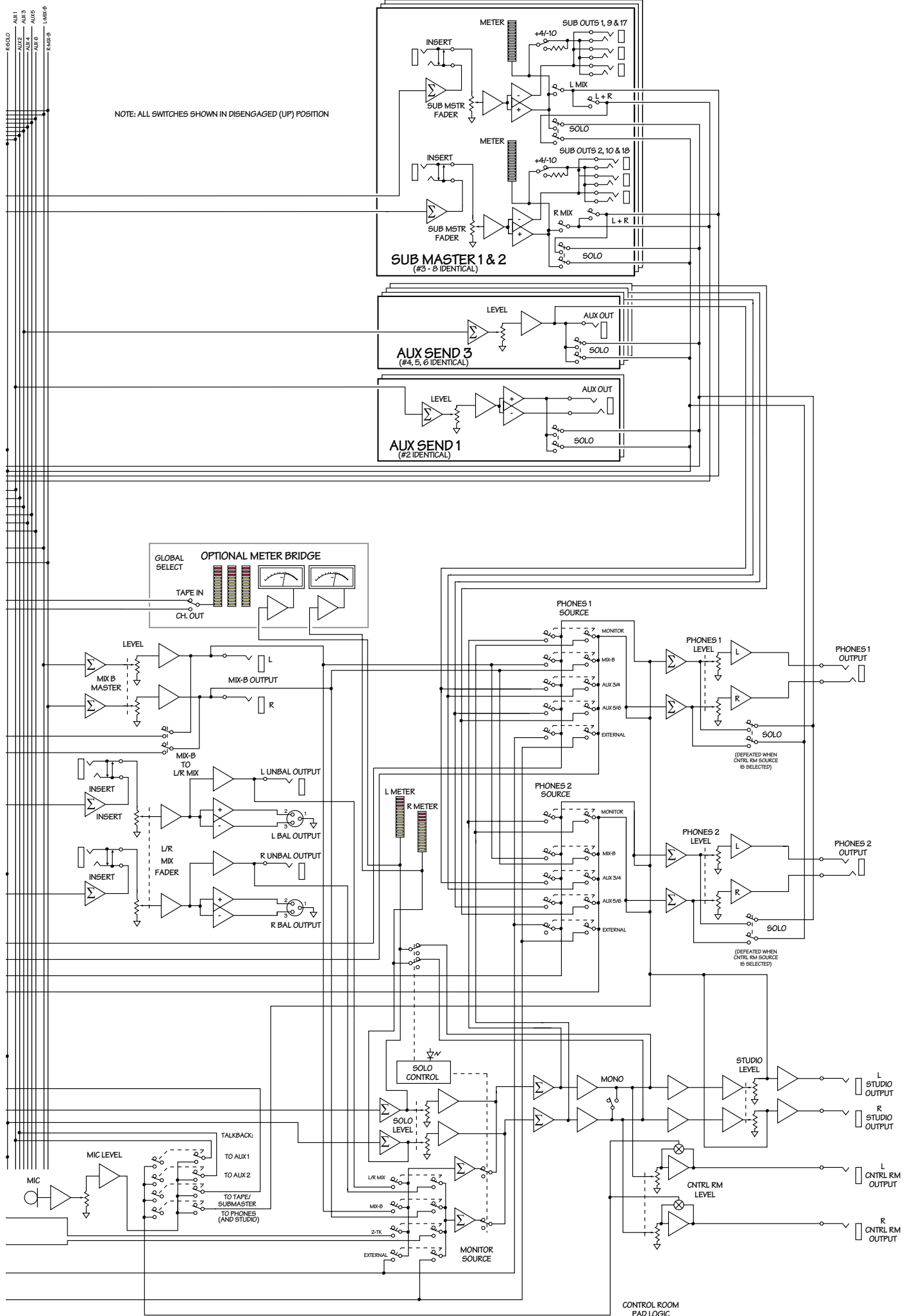
8-BUS CONSOLE BLOCK DIAGRAM

Rev. 5.1- 6/6/95

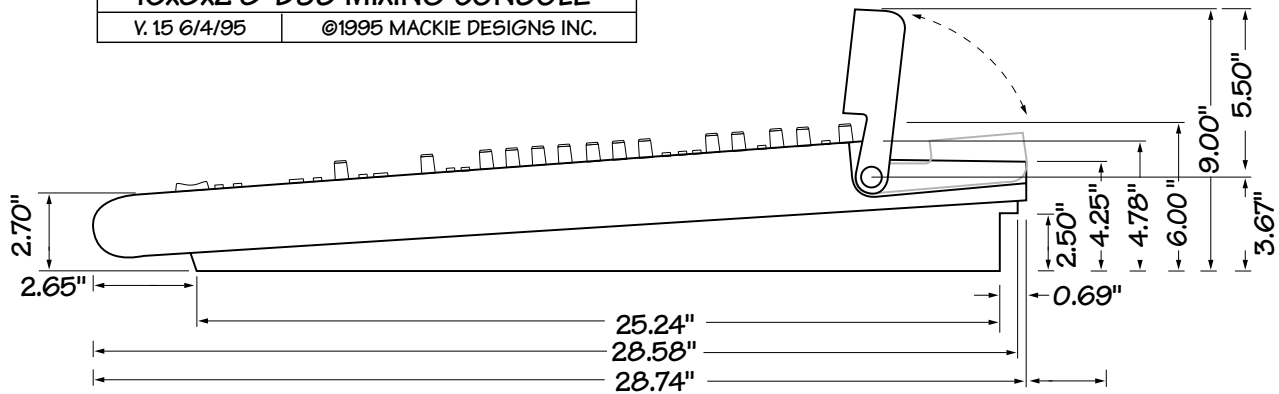
©1995 Mackie Designs

OPTIONAL 24-E
24-CHANNEL MACKIE 8-BUS
EXPANDER CONSOLE

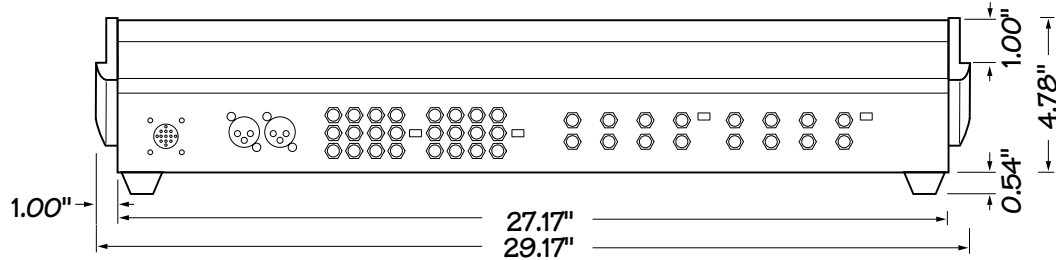




MACKIE. 16•8
16x8x2 8-BUS MIXING CONSOLE
 V. 15 6/4/95 ©1995 MACKIE DESIGNS INC.



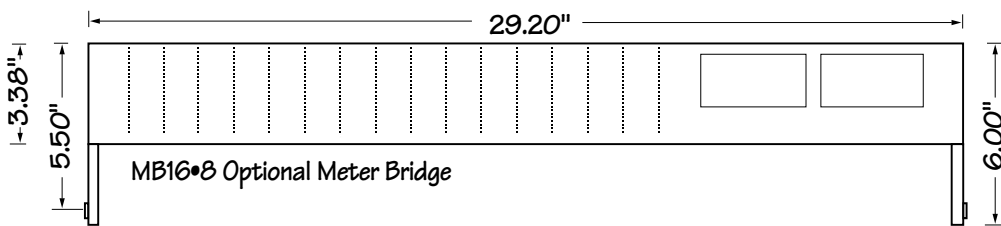
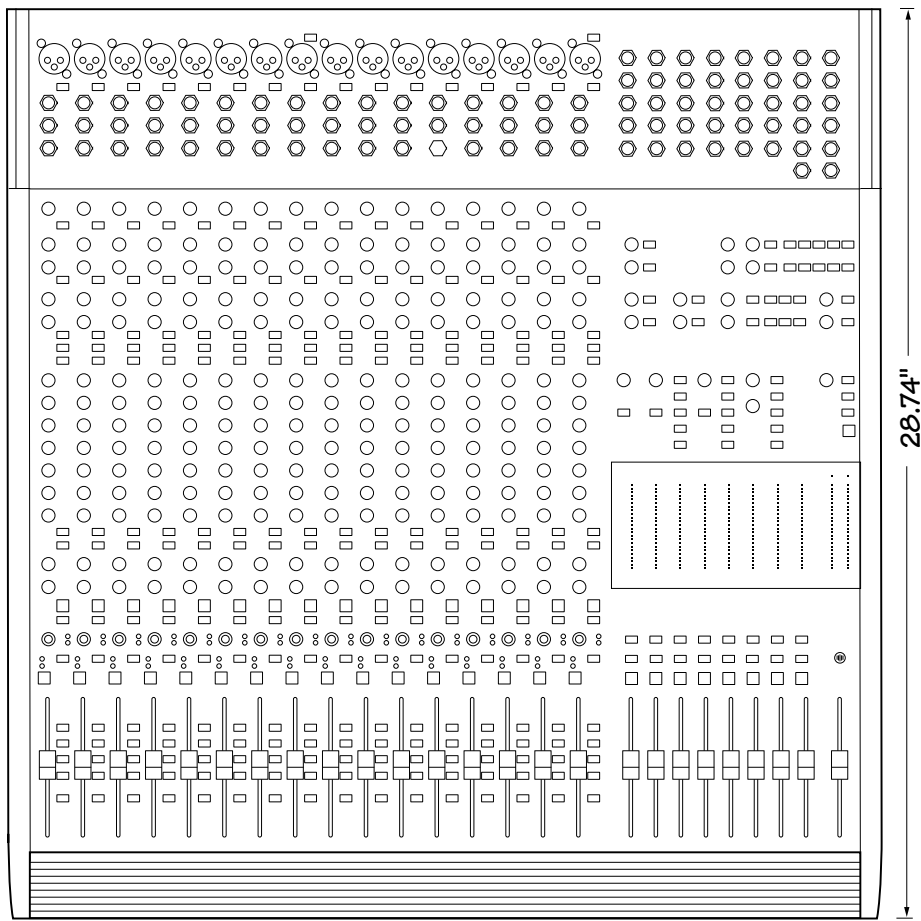
Note: add 3.75" to depth for power supply cable clearance



16•8 console
 WEIGHT*
 50 lbs.

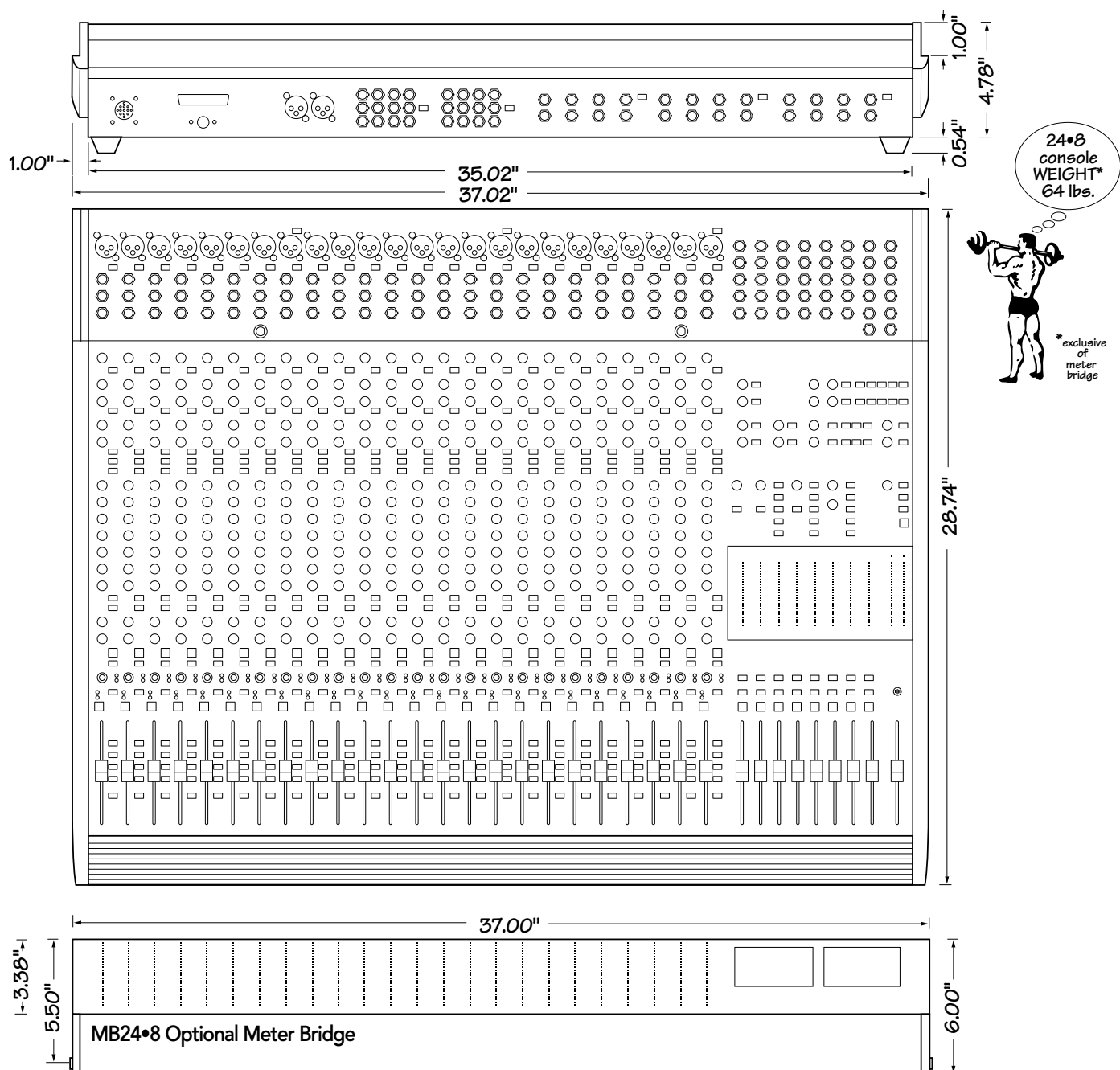
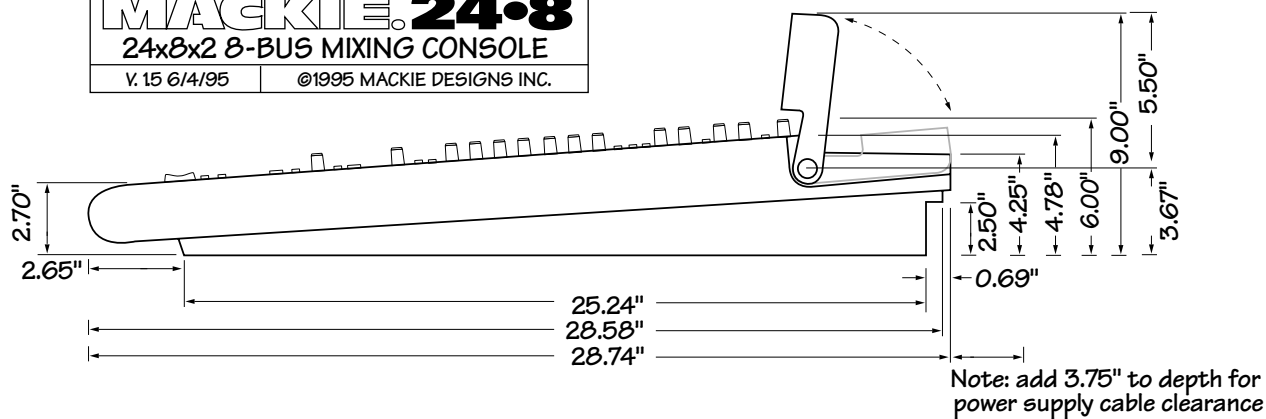


*exclusive of meter bridge

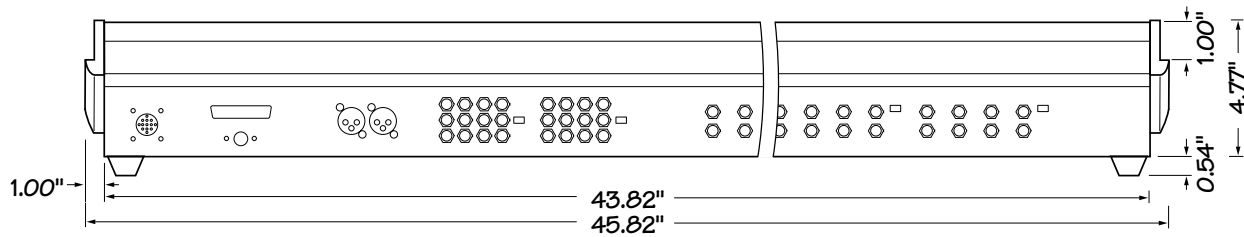
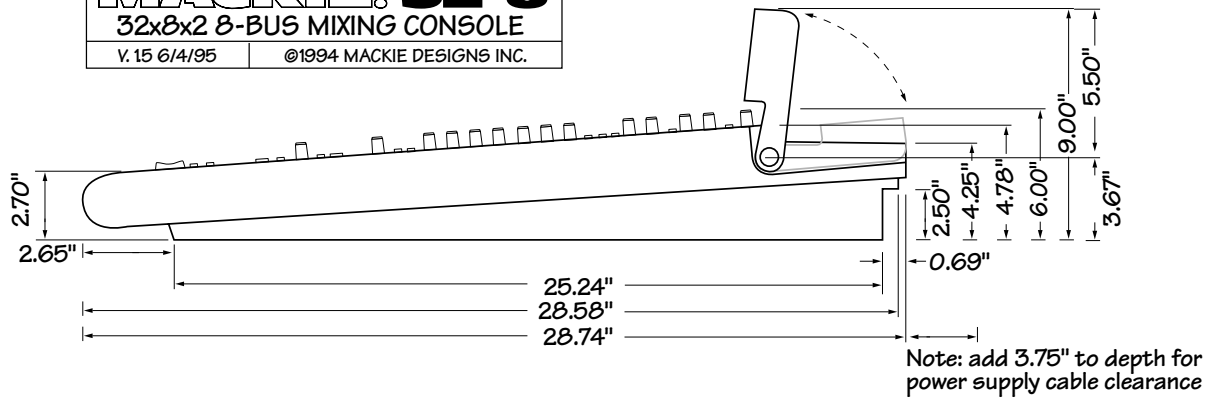


MB16•8 Optional Meter Bridge

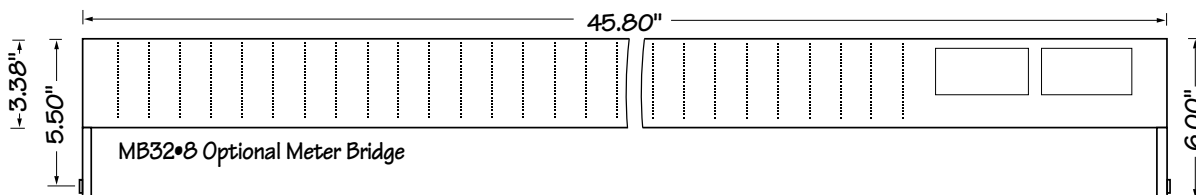
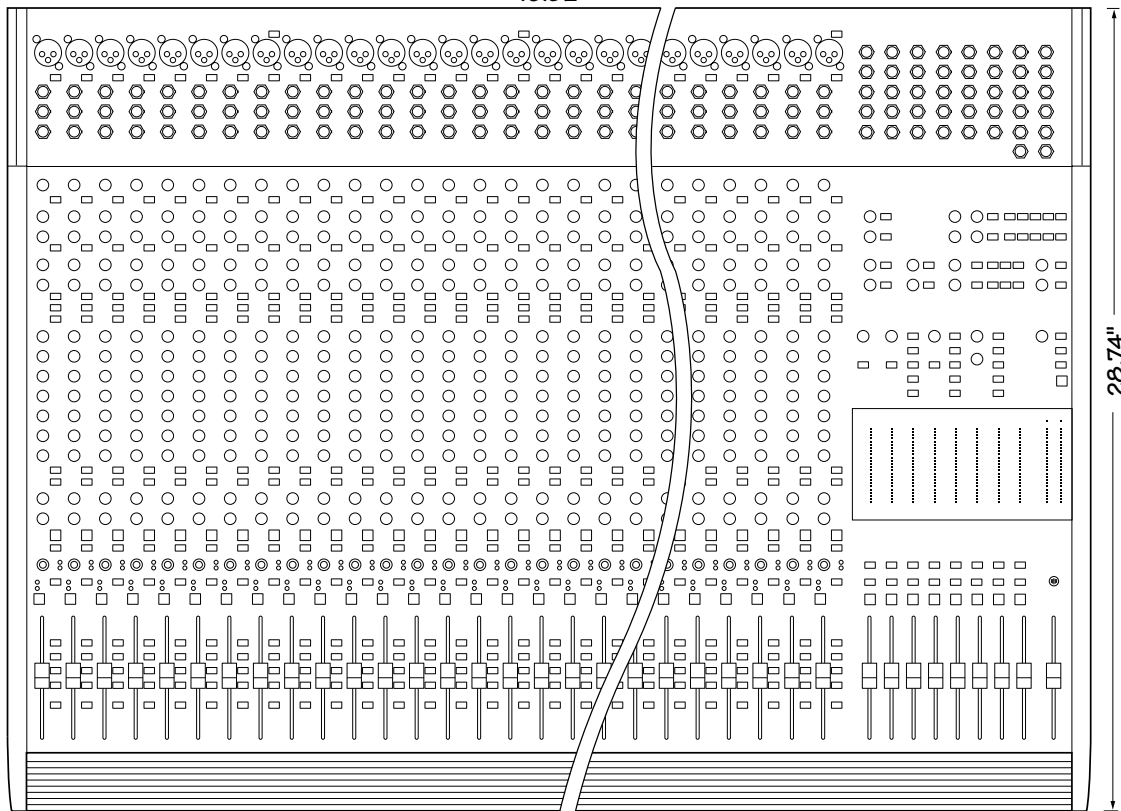
MACKIE.24•8
24x8x2 8-BUS MIXING CONSOLE
 V. 15 6/4/95 ©1995 MACKIE DESIGNS INC.



MACKIE. 32•8
32x8x2 8-BUS MIXING CONSOLE
 V. 15 6/4/95 ©1994 MACKIE DESIGNS INC.



32•8 console
 WEIGHT*
 78 lbs.



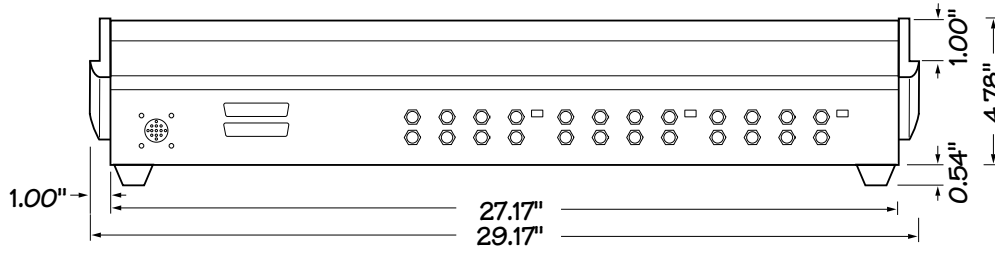
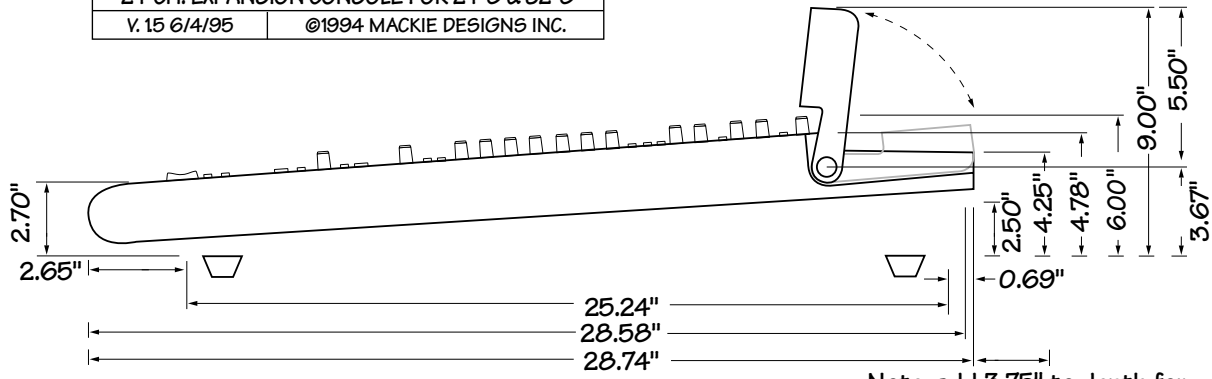
MB32•8 Optional Meter Bridge

MACKIE. 24•E

24-CH. EXPANSION CONSOLE FOR 24•B & 32•B

V. 15 6/4/95

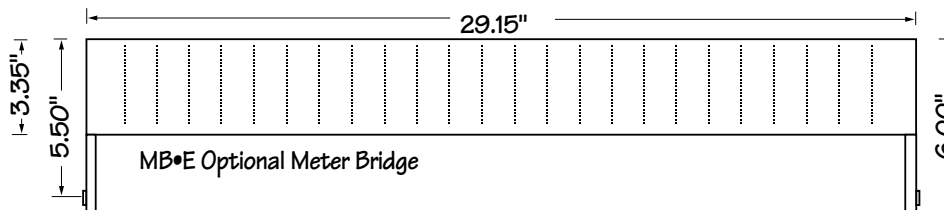
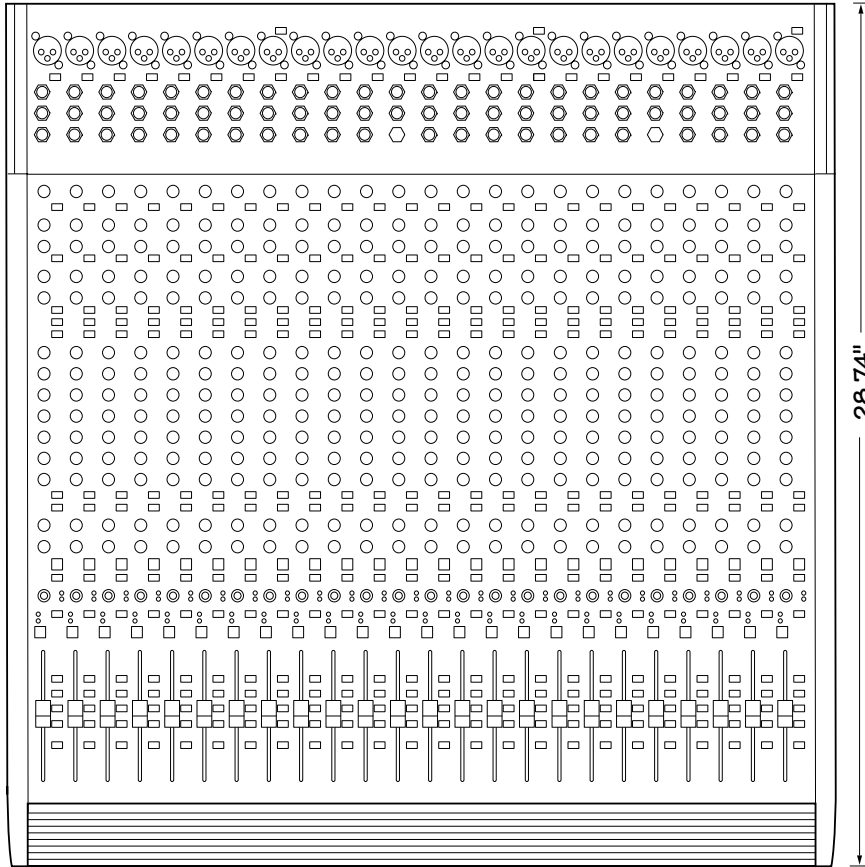
©1994 MACKIE DESIGNS INC.



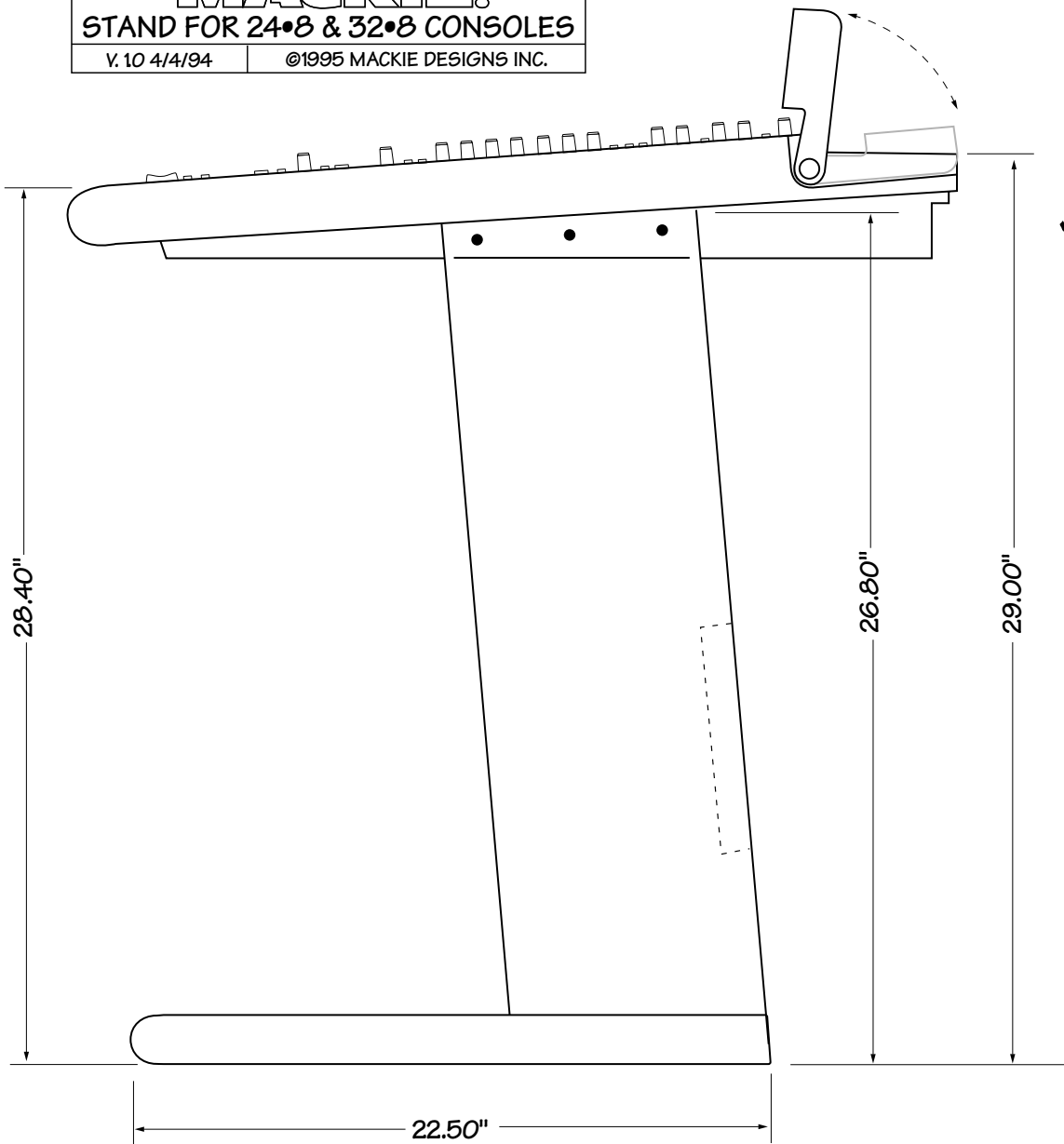
24•E
console
WEIGHT*
50 lbs.



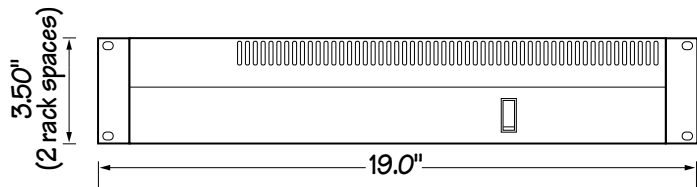
*exclusive
of meter
bridge



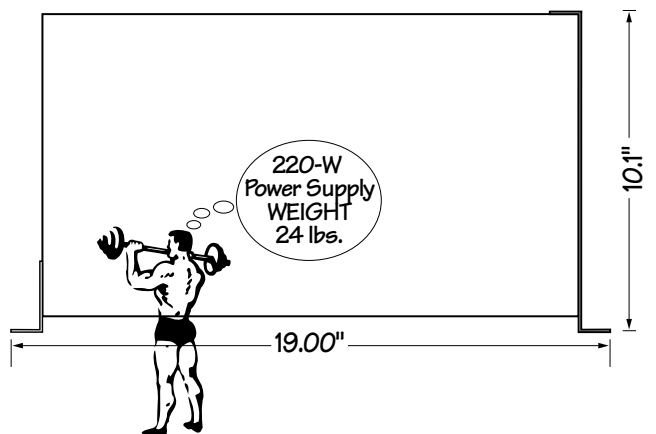
MACKIE.
STAND FOR 24•8 & 32•8 CONSOLES
 V. 10 4/4/94 ©1995 MACKIE DESIGNS INC.



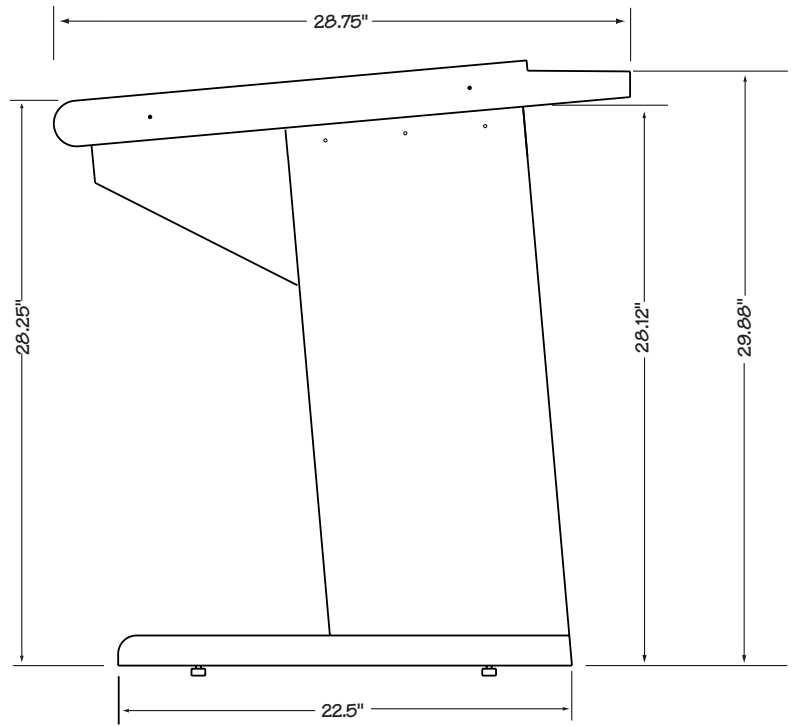
Stand
 WEIGHT
 44 lbs.



MACKIE.
220-WATT POWER SUPPLY FOR
8•BUS CONSOLES & EXPANDER
 V. 12 4/4/94 ©1994 MACKIE DESIGNS INC.

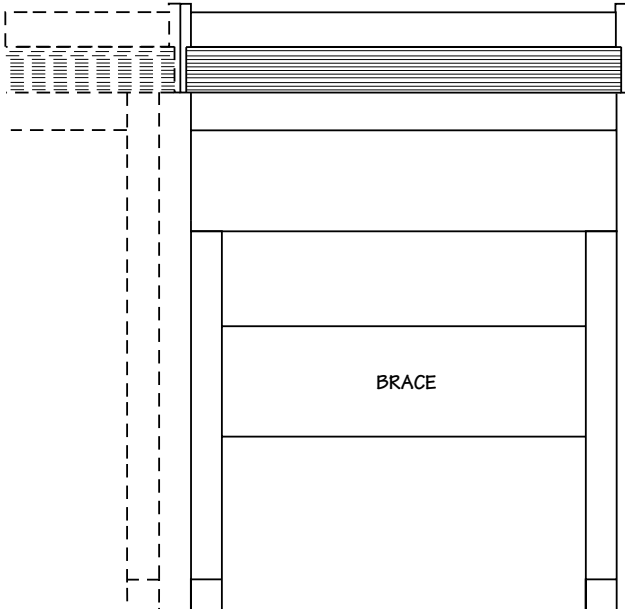


220-W
 Power Supply
 WEIGHT
 24 lbs.

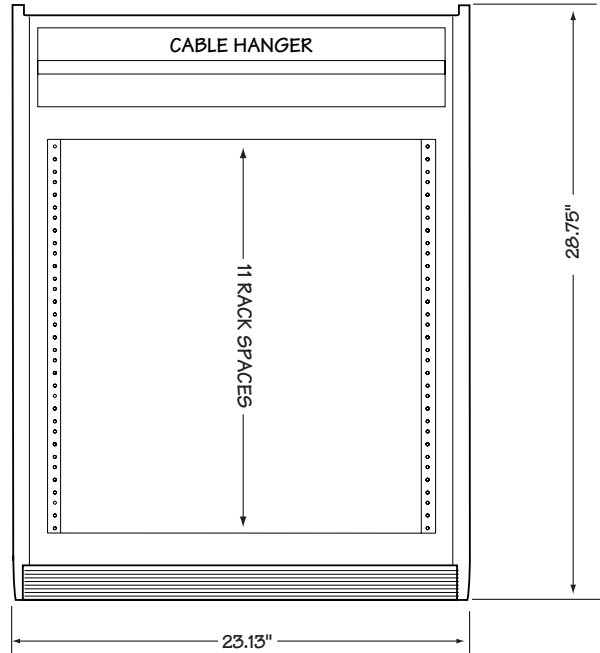


SIDE VIEW

[existing console]



FRONT VIEW



TOP VIEW

*ALL SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE

Appendix C: Modifications

General Notes about Modifications

We've tried to make these mixers as universal as possible, but you can't please everyone all the time. For certain applications, you may find these signal routing changes beneficial.

They're easy to make if you have experience soldering and working on electronic equipment, but if you don't have the experience, this is NOT a good place to learn. Take your mixer and this book to a qualified technician.

Here are the obligatory warnings and disclaimers that our lawyers made us include. We think it's a good idea to heed them (the warnings, not the lawyers), too.

Caution! These modification instructions are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than changing the fuse unless you are qualified to do so. Refer all servicing and modifying to qualified personnel.

Mackie Disclaimer

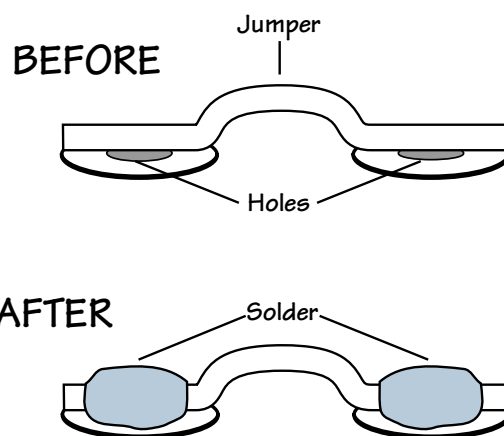
Any modification of any Mackie Designs product must be performed by a competent electronic technician. Mackie Designs accepts no responsibility for any damages or injuries caused by any modification, regardless of the source of the modification instructions or the qualifications of the technician performing them. In the case of such damages, Mackie Designs may declare warranty privileges void. BE CAREFUL!

About Jumpers

These modifications involve cutting traces on the circuit board and installing jumpers between pads. We recommend solid (non-stranded) wire, 26-28 gauge. The type of wire used for wire-wrapping is easy to form, lays neatly against the circuit board, and has insulation that doesn't melt easily from the heat of a soldering iron.

When installing jumpers, do not run their ends through holes in the circuit board.

Rather, solder them flat against the desired pad (the flat silver area, possibly with a hole in the middle). Make sure the ends of these flat wires do not extend beyond the pad.



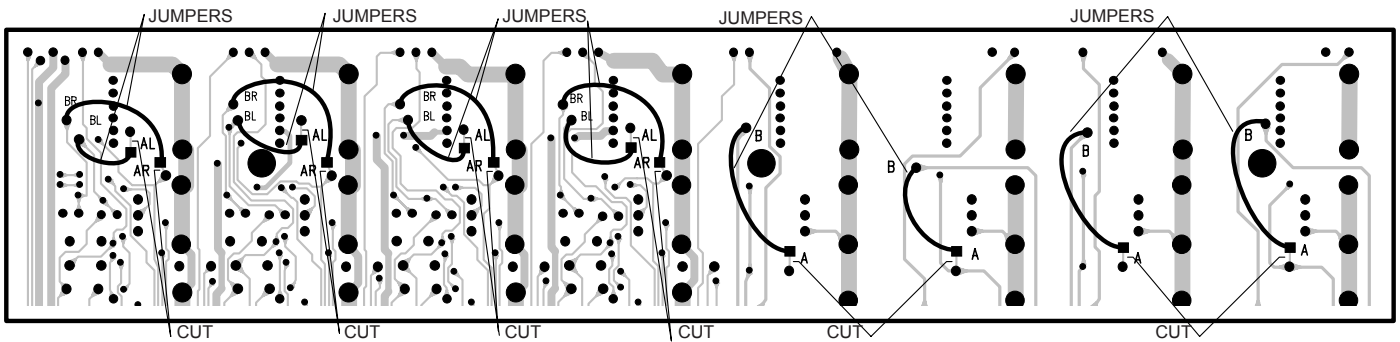
1202-VLZ PRO Modifications

Pre-Fader Mod (Aux To Monitor)

This modification changes AUX SEND 2 to be *pre-fader, pre-mute* instead of *post-fader, post-mute*. ("Fader" refers to the channel GAIN knob, and "Mute" refers to the channel's MUTE/ALT 3-4 switch.) In order to convert the entire mixer, it must be done on each channel. Is slightly more involved for the stereo channels 5-12. The work area is on the underside of the circuit board, near the channel AUX SEND knobs.

1. Remove all cords, including the power cable, from the 1202-VLZ PRO.
2. Place the mixer upside-down on a dry, non-marring surface.
3. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.
4. Using a sharp "X-acto" type knife, cut the conductor at point 'A' (channels 1-4) or the conductors at points 'AL' and 'AR' (channels 5-12). Be careful to cut all the way through the conductor, and do not cut any nearby traces.
5. Add a jumper from point 'B' to the square pad at point 'A' (channels 1-4) or from points 'BL' to 'AL' and 'BR' to 'AR' (channels 5-12).

6. Repeat for all channels.
7. Check your work very carefully, then put the bottom cover back the way you found it. You're done!



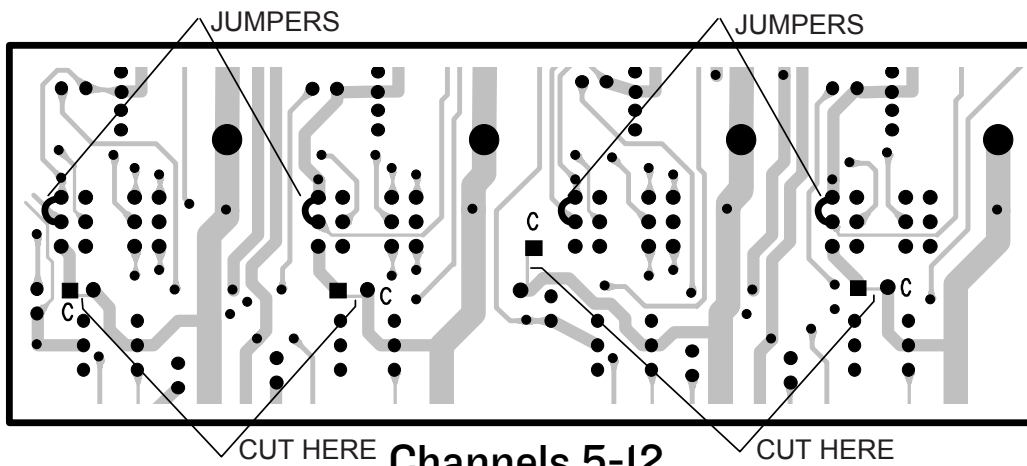
Pre-Fader Mod

Pre-Mute Mod

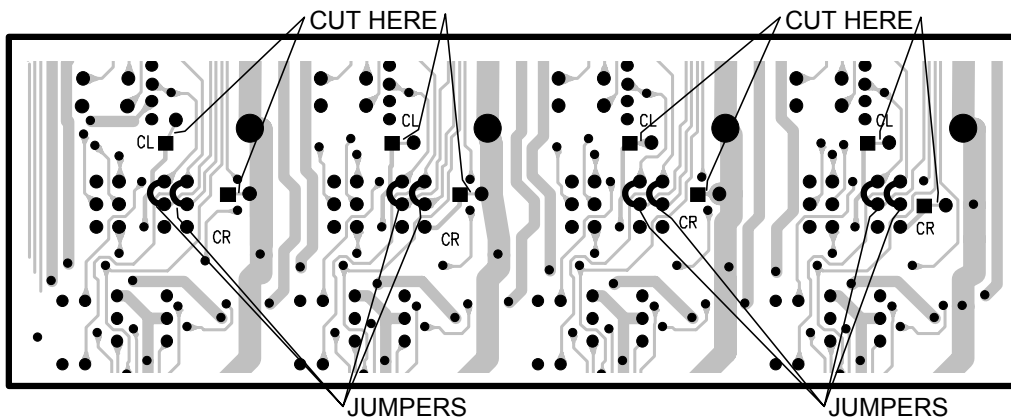
This modification changes AUX SEND 1 (in *post* mode) and AUX SEND 2 to receive signal regardless of the channel's MUTE/ALT 3-4 switch position, but still be *post-fader* (GAIN knob). In order to convert the entire mixer, it must be done on each channel. It is slightly more involved for the stereo channels 5 through 12. The work area is on the underside of the circuit board, near the channel MUTE/ALT 3-4 switches.

1. Remove all cords, including the power cable, from the 1202-VLZ PRO.
2. Place the mixer upside-down on a dry, non-marring surface.
3. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.
4. Using a sharp "X-acto" type knife, cut the conductor at point 'C' (channels 1-4) or the conductors at points 'CL' and 'CR' (channels 5-12). Be careful to cut all the way through the conductor, and do not cut any nearby traces.
5. Locate the 12 pins that comprise the underside of each MUTE/ALT 3-4 switch.
6. Add jumpers as shown on the illustration below—they're not marked on the circuit board itself, so be careful.
7. Repeat for all channels.
8. Check your work very carefully, then put the bottom cover back the way you found it. You're done!

Channels 1-4



Channels 5-12



Pre-Mute Mod

Main Mix Source Mod

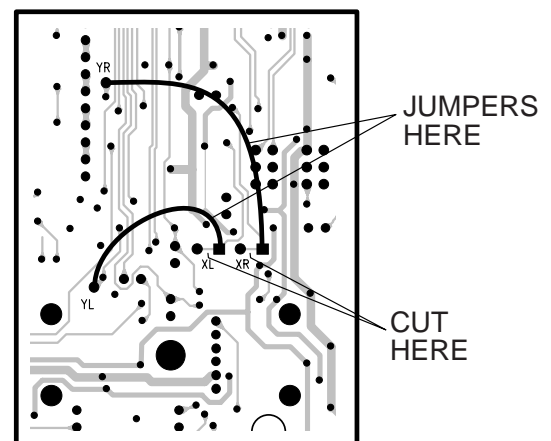
This modification changes the SOURCE matrix's MAIN MIX selection to tap the stereo signal before the MAIN MIX level control (*pre*) instead of after (*post*). This could be especially handy for live work

where the engineer wants to be able to control the MAIN MIX level (sent to the house system) without changing the level in his headphones. The work area is on the underside of the circuit board, near the MAIN MIX level control.

Caution: This modification also causes the meters to indicate pre MAIN MIX levels. They will no longer indicate the signal level at the MAIN OUTS, but rather the signal level at the PHONES and CONTROL ROOM outputs (when MAIN MIX SOURCE is selected).

1. Remove all cords, including the power cable, from the 1202-VLZ PRO.
2. Place the mixer upside-down on a dry, non-marring surface.

3. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.
4. Using a sharp "X-acto" type knife, cut the conductor at points 'XL' and 'XR'. Be careful to cut all the way through the conductor, and do not cut any nearby traces.
5. Add a jumper from point 'YL' to the square pad at point 'XL' and from point 'YR' to the square pad at point 'XR'.
6. Check your work very carefully, then put the bottom cover back the way you found it. You're done!



Main Mix Source Mod

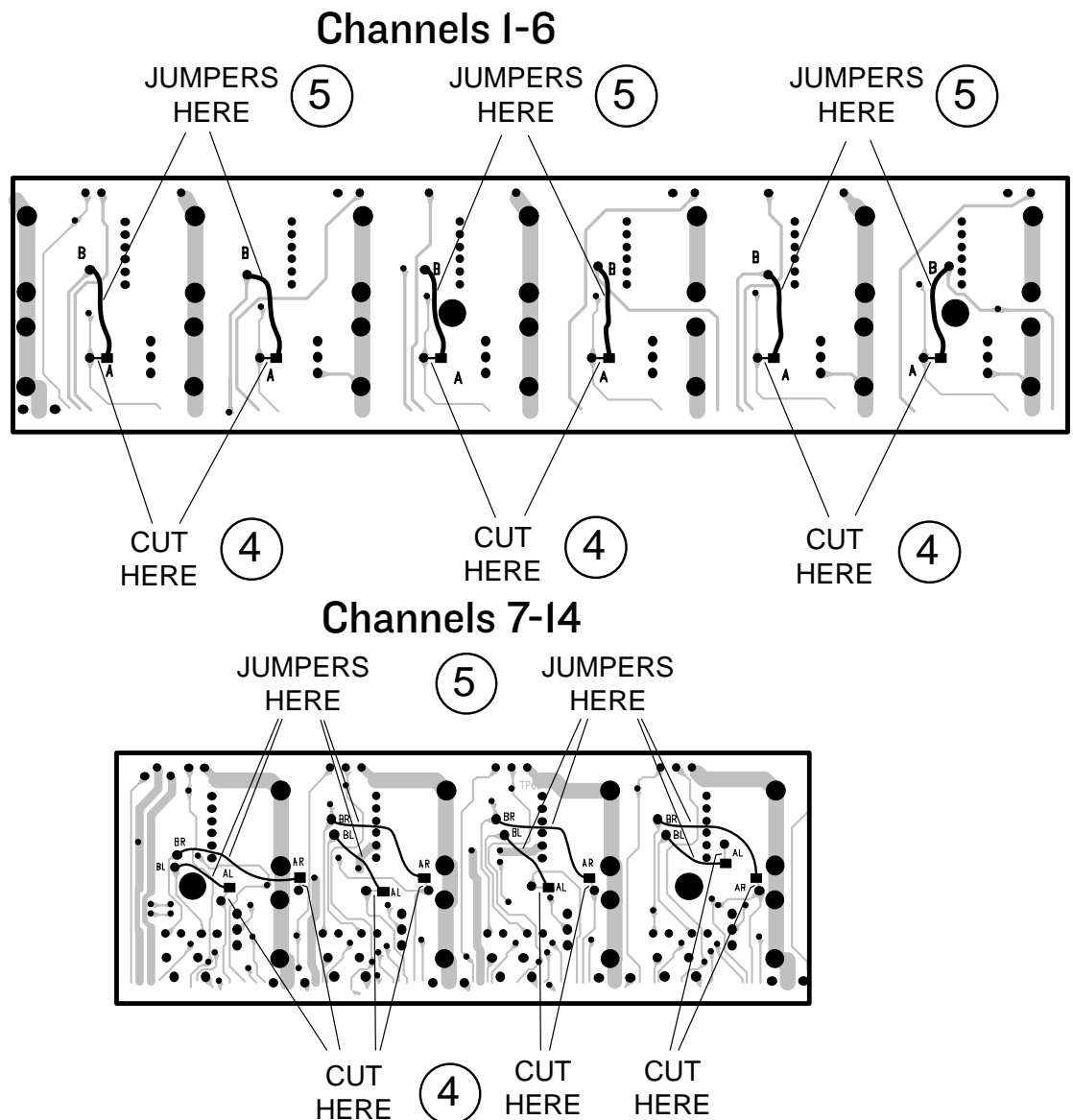
I402-VLZ PRO Modifications

Pre-Fader Mod (Aux To Monitor)

This modification changes AUX SEND 2 to be *pre-fader, pre-mute* instead of *post-fader, post-mute*. (“Mute” refers to the channel’s MUTE/ALT 3–4 switch.) In order to convert the entire mixer, it must be done on each channel, and is slightly more involved for the stereo channels 7–14. The work area is on the underside of the circuit board, near the channel AUX SEND knobs.

1. Remove all cords, including the power cable, from the 1402-VLZ PRO.
2. Place the mixer upside-down on a dry, non-marring surface.
3. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.

4. Using a sharp “X-acto” type knife, cut the conductor at point ‘A’ (channels 1–6) or the conductors at points ‘AL’ and ‘AR’ (channels 7–14). Be careful to cut all the way through the conductor, and do not cut any nearby traces.
5. Add a jumper from point ‘B’ to the square pad at point ‘A’ (channels 1–6) or from points ‘BL’ to ‘AL’ and ‘BR’ to ‘AR’ (channels 7–14).
6. Repeat for all channels.
7. Check your work very carefully, then put the bottom cover back the way you found it. You’re done!

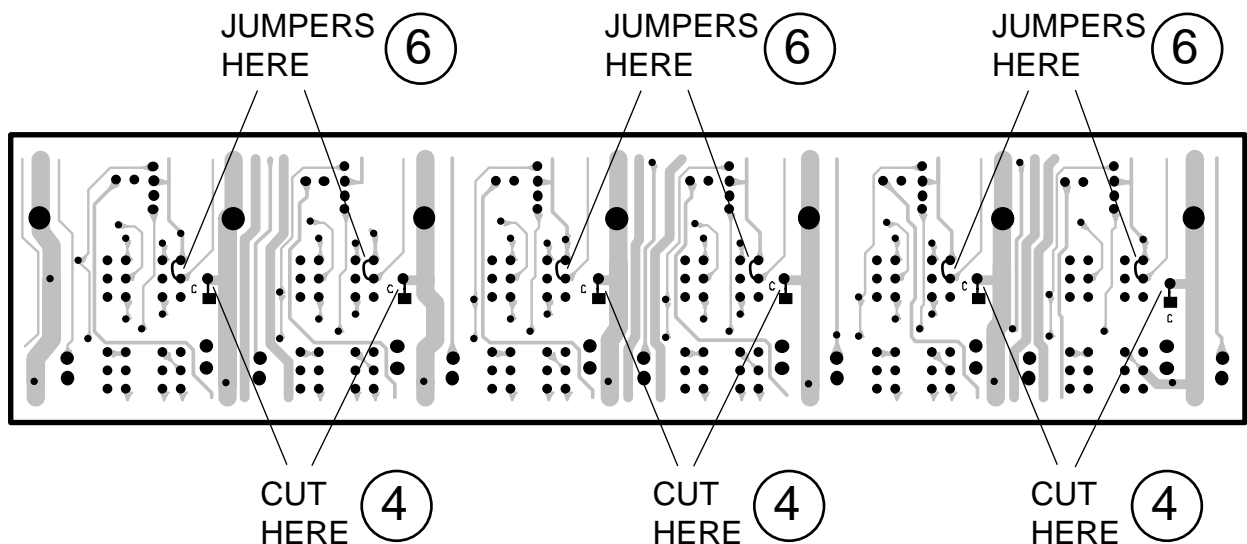


Pre-Mute Mod

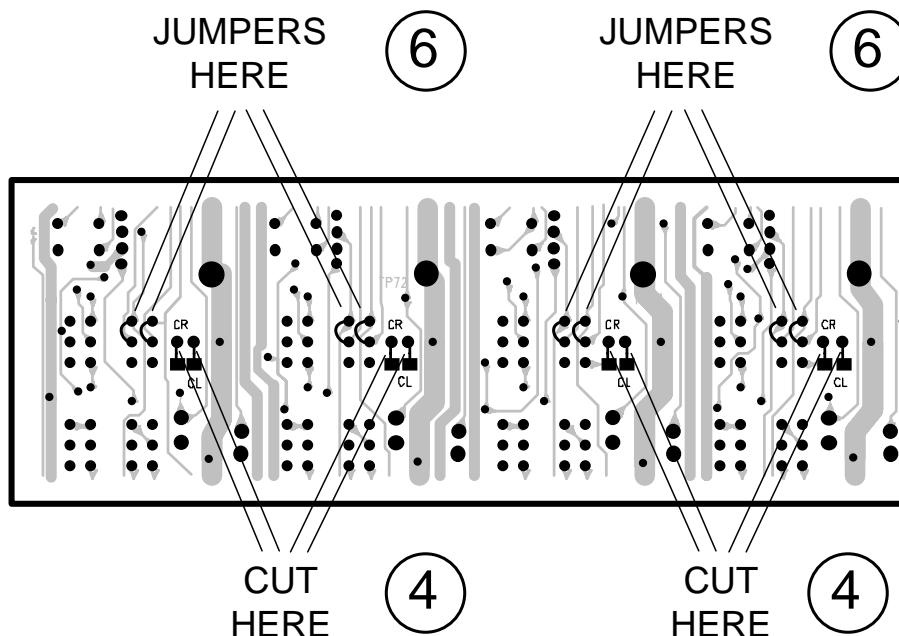
This modification changes AUX SEND 1 (in *post* mode) and AUX SEND 2 to receive signal regardless of the channel's MUTE/ALT 3-4 switch position, but still be *post-fader*. In order to convert the entire mixer, it must be done on each channel, and is slightly more involved for the stereo channels 7-14. The work area is on the underside of the circuit board, near the channel MUTE/ALT 3-4 switches.

1. Remove all cords, including the power cable, from the 1402-VLZ PRO.
2. Place the mixer upside-down on a dry, non-marring surface.
3. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.
4. Using a sharp "X-acto" type knife, cut the conductor at point 'C' (channels 1-6) or the conductors at points 'CL' and 'CR' (channels 7-14). Be careful to cut all the way through the conductor, and do not cut any nearby traces.
5. Locate the 12 pins that comprise the underside of each MUTE/ALT 3-4 switch.
6. Add jumpers as shown on the illustration below—they're not specifically marked on the circuit board itself, so be careful.
7. Repeat for all channels.
8. Check your work very carefully, then put the bottom cover back the way you found it. You're done!

Channels 1-6



Channels 7-14



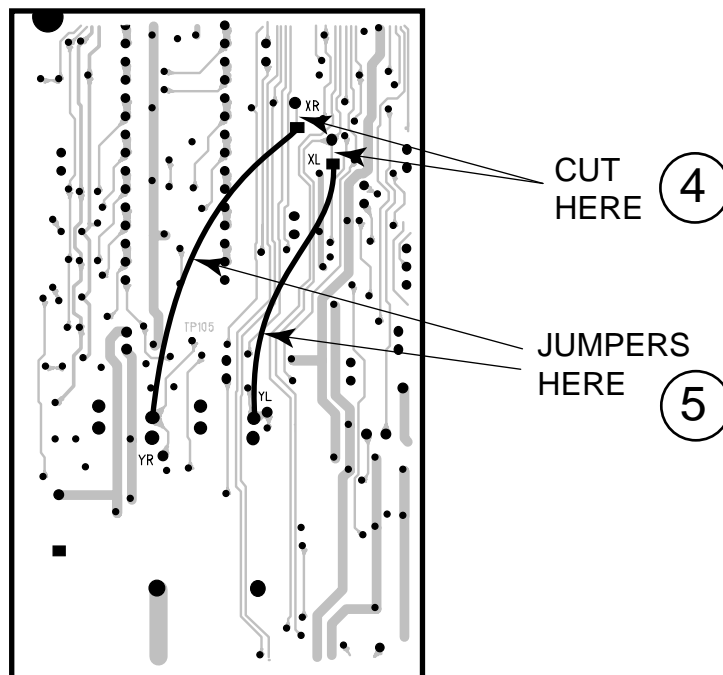
Main Mix Source Mod

This modification changes the SOURCE matrix's MAIN MIX selection to tap the stereo signal before the MAIN MIX level control (*pre*) instead of after (*post*). This could be especially handy for live work where the engineer wants to be able to control the MAIN MIX level (sent to the house system) without changing the level in his headphones. The work area is on the underside of the circuit board, near the MAIN MIX level control.

Caution: This modification also causes the meters to indicate pre MAIN MIX levels. They will not longer indicate the signal level at the MAIN OUTS, but rather the signal level at the PHONES and CONTROL ROOM outputs (when MAIN MIX SOURCE is selected).

1. Remove all cords, including the power cable, from the 1402-VLZ PRO.
2. Place the mixer upside-down on a dry, non-marring surface.
3. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.
4. Using a sharp "X-acto" type knife, cut the conductor at points 'XL' and 'XR'. Be careful to cut all the way through the conductor, and do not cut any nearby traces.

5. Add a jumper from point 'YL' to the square pad at point 'XL' and from point 'YR' to the square pad at point 'XR'.
6. Check your work very carefully, then put the bottom cover back the way you found it. You're done!



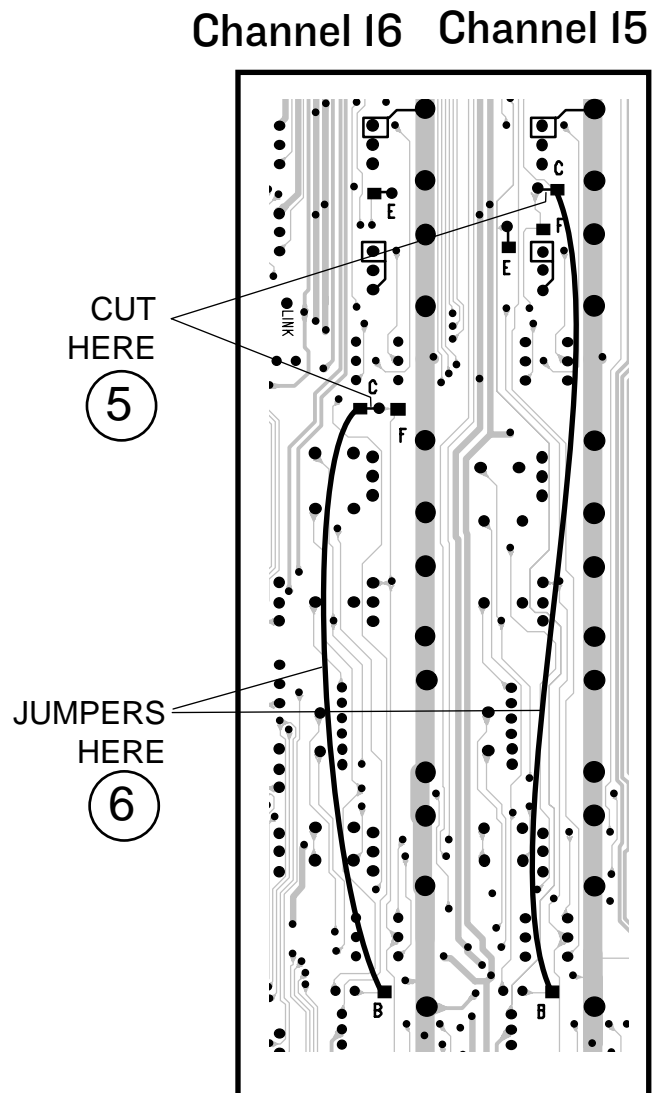
I604-VLZ PRO Modifications

I604-VLZ PRO Post-EQ Mod

This changes AUX SENDS 1 and 2, with the pre switch engaged, to receive their signals *post*-EQ instead of *pre*-EQ. The signal remains post-low cut, pre-mute and pre-fader. With the pre switch disengaged (up), the signals are not affected by the mod. The following must be performed for each channel you wish to modify:

1. Remove all cords, including the power cable, from the I604-VLZ PRO.
2. Place the mixer upside-down on a dry, non-marring surface.
3. If you have converted your mixer to the rack-mount position or have installed a RotoPod, undo those changes and temporarily configure the mixer in the original desktop mode. You do not have to install the pod, just get it out of the way of the bottom cover.

4. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.
5. Cut the conductor at point C, between the square and round pads. Be careful to cut all the way through the conductor, and do not cut any nearby traces. Each channel is slightly different, but this graphic shows Channel 16, which is very different from the others, and Channel 15 (respectively), which is similar to the remaining channels.
6. Add a jumper from the square pad at point B to the square pad at point C.
7. Repeat for each channel you wish to modify.
8. Check your work very carefully, then put the bottom cover back the way you found it.
You're done!



1604-VLZ PRO Source Mod

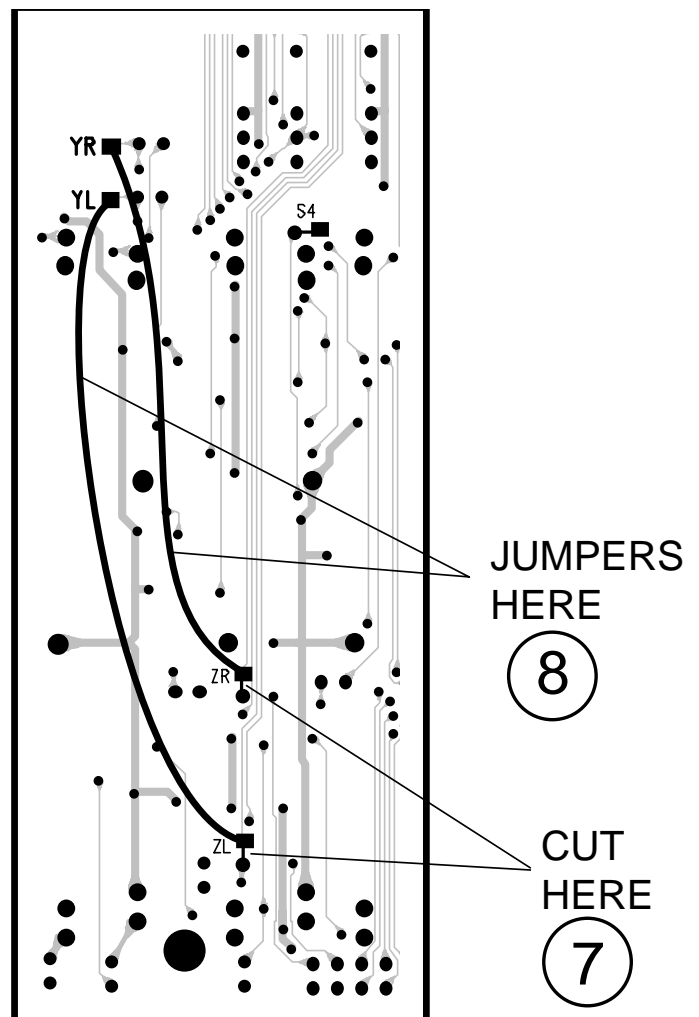
This changes the CTL ROOM/PHONES level control in the SOURCE matrix to receive the main mix stereo signal *pre*-MAIN MIX fader instead of *post*-MAIN MIX fader.

You can accomplish the same result that this modification provides by using two standard 1/4" tip-sleeve "jumper cables" plugged into the MAIN INSERT (L and R) to the *first click* and the other end plugged into STEREO AUX RETURN 4, assigned to C-R/PHNS ONLY. STEREO AUX RETURN 4 level will control the volume as well as CTL ROOM/PHONES level control.

You can also use the TAPE INPUT (L and R) RCA jacks (you will need 1/4" to RCA cables or adapters) and assign the TAPE source button in the SOURCE matrix.

1. Remove all cords, including the power cable, from the 1604-VLZ PRO.
2. Undo the PHONES nut.
3. Place the mixer upside-down on a dry, non-marring surface.

4. If you have converted your mixer to the rack-mount position or have installed a RotoPod, undo those changes and temporarily configure the mixer in the original desktop mode. You do not have to install the pod, just get it out of the way of the bottom cover.
5. Remove the screws that attach the bottom cover. Keep track of what screws go where. Remove the bottom cover.
6. Move the PHONES board to one side so you can get to points YL and YR marked on the board.
7. Cut the conductor at points ZL and ZR, between the square and round pads. Be careful to cut all the way through the conductor, and do not cut any nearby traces.
8. Add a jumper from the square pad at point YL to the square pad at point ZL and another from the square pad at point YR to the square pad at point ZR.
9. Check your work very carefully, then put the Phono board and nut, and bottom cover back the way you found them. You're done!



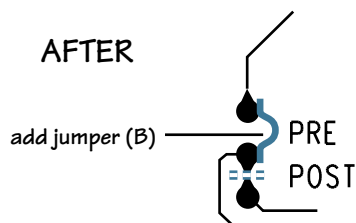
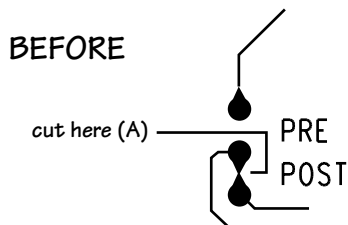
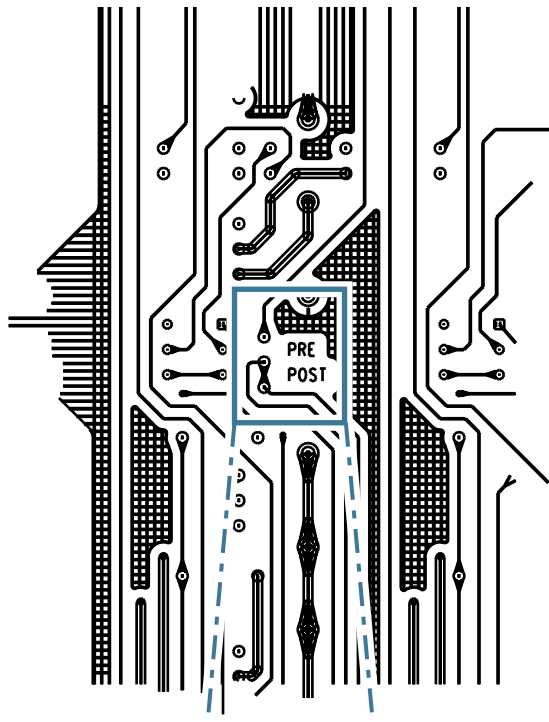
8•Bus Modifications

AUX Send Mod

This modification changes the tap point of all “pre” AUX sends from *post-EQ* to **pre-EQ**. It must be done on each channel. For example, if you have a 24•8, the modification must be done on all 24 input channels.

See Figure below. This modification takes place on each channel strip in an area under the AUX 1/2 Pre/Post switch.

1. Remove power cable.
2. Cut the conductor at Point A.
3. Add a jumper at Point B.
4. Repeat for all input channels.



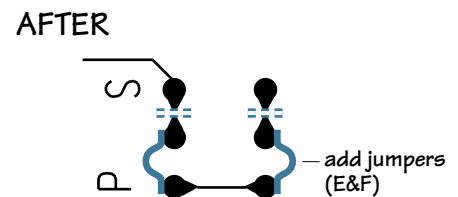
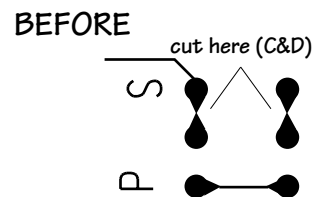
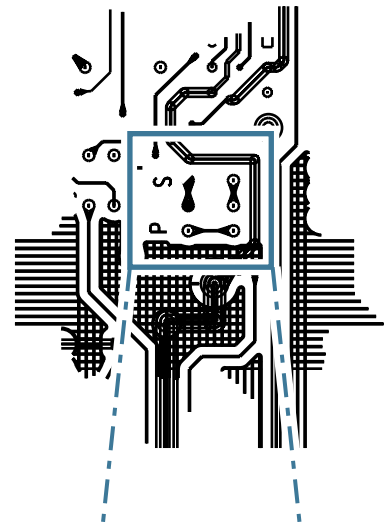
Aux Send Modification

PFL Mod

This modification changes the tap point of the SOLO bus from *post-fader/post-mute* (stereo) to *pre-fader/pre-mute* (mono). It must be done on each channel. For example, if you have a 24•8, the modification must be done on all 24 input channels.

See Figure below. This modification takes place on each channel strip in an area under the channel fader.

1. Remove power cable.
2. Cut the conductor at Point C.
3. Cut the conductor at Point D.
4. Add a jumper at Point E.
5. Add another jumper at Point F.
6. Repeat for all input channels.



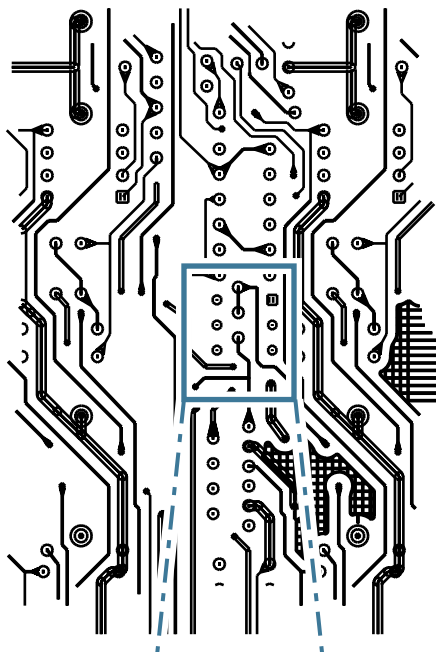
PFL Modification

Mix-B Source Mod

This modification changes the tap point of the Mix-B Source switch (engaged) from *pre-fader/pre-mute* to *post-fader/post-mute*. It must be done on each channel. For example, if you have a 24•8, the modification must be done on all 24 input channels.

See Figure below. This modification takes place on each channel strip in an area under the Mix-B Source switch.

1. Remove power cable.
2. Cut the conductor at Point G.
3. Add a jumper at Point H.
4. Repeat for all input channels.



BEFORE

cut here (G)

AFTER

add jumper (H)

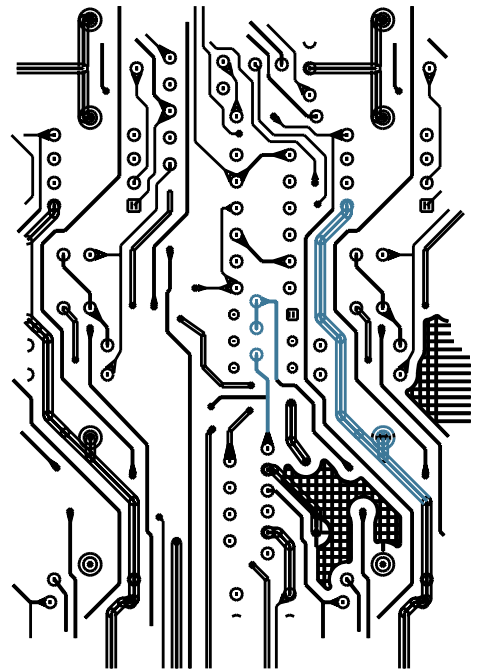
Mix-B Source Modification

Mix-B Mute Mod

This modification converts the Mix-B Source switch to a Mix-B MUTE switch. The switch will no longer be able to source the channel signal. This modification must be done on each channel. For example, if you have a 24•8, the modification must be done on all 24 input channels.

See Figures below. This modification takes place on each channel strip in an area under the Mix-B Source switch.

1. Remove power cable.
2. Cut the conductor at Point G.
3. Add Jumper to point H.
4. Repeat for all input channels.



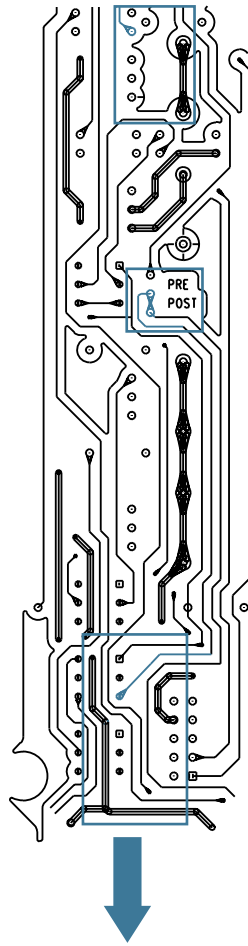
BEFORE

cut here (G)

AFTER

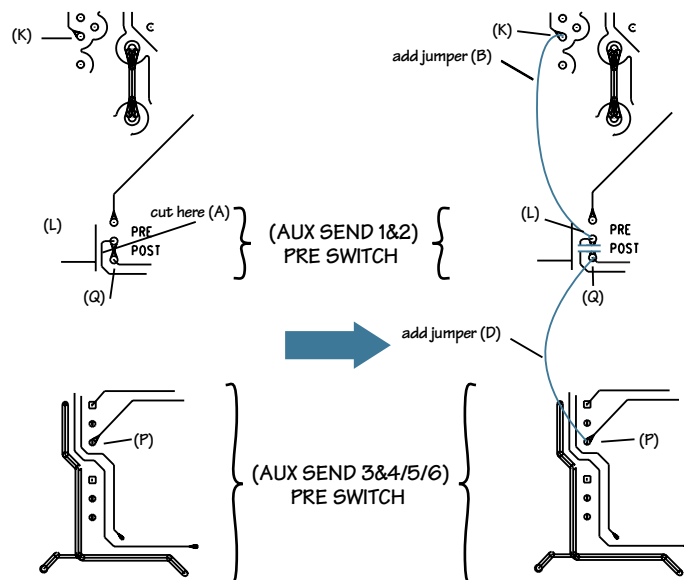
add jumper (H)

Mix-B Mute Modification



BEFORE

AFTER



The infamous and dreaded but-if-we-hadn't-included-it-we'd-just-have-to-fax-it-to-the-foolhardy-anyway modification

AUX 1/2 Source Mod



This modification should only be performed if the console is being used strictly for recording.

A. When the Aux 1/2 Pre Switch is engaged, Aux Send 1 and 2 will tap the Pre-Fader information from the tape returns.

- B. When the Aux 1/2 Pre Switch is *not* engaged, Aux Send 1 and 2 function normally, as post fader channel sends.

All that said, here's the mod.

See Figure above. This modification takes place on each channel strip in the area under the AUX 3/4 Mix-V Source switch and also under and near the AUX 1/2 PRE switch.

1. Remove power cord.
2. Cut two (2) traces at point (A).
Note: The wire used in the next three steps should be 24-28 gauge insulated jumper wire.
Do not strip off any more insulation than is absolutely necessary.
3. Add a 1-1/2" jumper (B), connected at points K and L.
4. Add a 2-1/4" jumper (D), connected at points P and Q.

What follows are Mike's suggestions for headings for each modification. I've currently left them as they are in the manuals. Please let me know what you think of this, and which headings you prefer. Thanks.

I604VLZ-PRO

AUX I-2 Post-EQ

CTL ROOM/PHONES Pre-fader

I402VLZ-PRO

AUX 2 Pre-fader/Mute

AUX I-2 Pre-Mute/Post-fader

Main Mix SOURCE Pre-fader

I202VLZ-PRO

AUX 2 Pre-fader/Mute

AUX I-2 Pre-Mute/Post-fader

Main Mix SOURCE Pre-fader

8-Bus

AUX Send Pre-EQ

PFL Solo

Mix-B Source Post-fader

Mix-B Mute Switch

AUX I-2 Pre From Tape Returns

