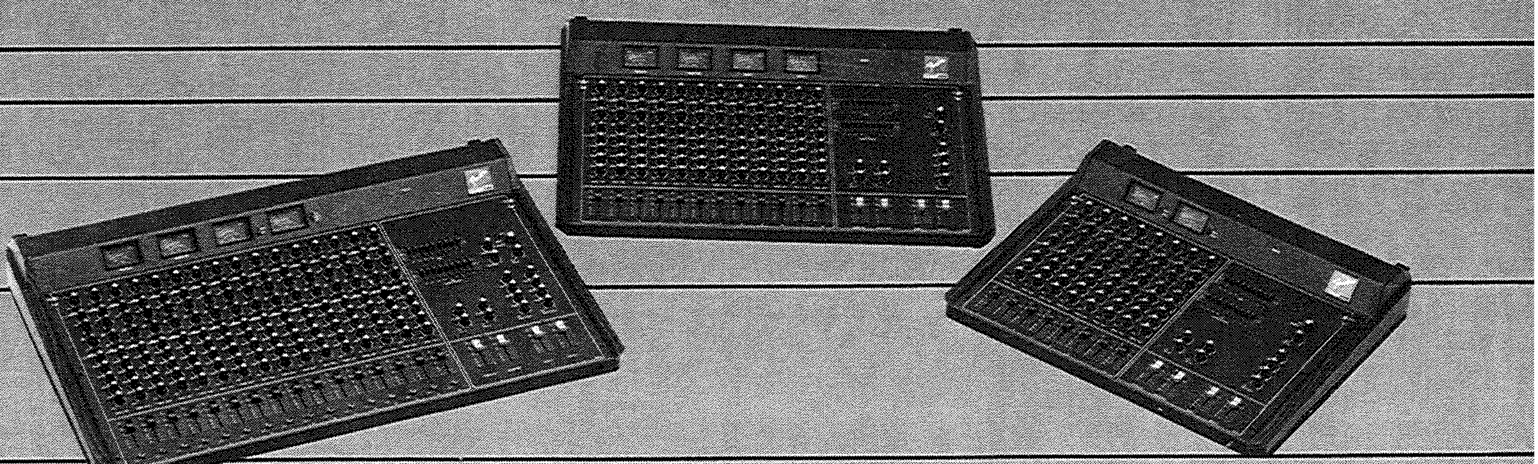
Fender

OWNER/APPLICATION MANUAL

$5.00

Series 4200 MIXING CONSOLES

4208
4212
4216



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Introducing Fender Professional Sound Equipment

The classic sound, look and feel of a Stratocaster, the "punch" of a Twin Reverb. The advanced technology of a Chroma synthesizer, the unmistakable tone of a Rhodes piano — just some of the reasons why you trust us with your music.

But your music doesn't stop with a Rhodes or a Stratocaster. It gets mixed, equalized, enhanced and amplified *long* before it reaches your audience! And with all *that* going on, your sound equipment is as important as your musical instrument.

That's why we build Fender Pro Sound Equipment. We help you perform your music; now we'll deliver it to your audience! And we'll do it in a way that *reinforces your performance and lets your sound become a creative part of your music.*

You'd expect as much from Fender. After all, you trust us with your music. And now, it's good to know you can trust us with your sound!

How to Use This Manual

As a Self-Teaching Guide

Section II is a self-teaching guide to your 4000 Series Fender Mixer. It takes you through the controls and switches on your 4000 Mixer, one by one, and explains not only their operation, but describes *artistic* ways you can use these controls. The Exercises suggested in the Self-Teaching Guide allow you to actually try out each control in a way that simulates an actual live performance.

The Self-Teaching Guide is detailed and assumes little prior knowledge on the part of the reader. If you find that you already understand much of what is explained in the Self-Teaching Guide, you may be able to skip the sections entitled "An Exercise" and concentrate on the section entitled "Block Diagram Closeup" and the various sections that deal with applications of each control and switch.

As a Reference Guide

Once you are familiar with the basic operation of your 4000 Mixer, you may wish to refer every now and then to the Specifications section and to the Block Diagram for your 4000 Mixer. Both of these sections are located near the front of this manual. If you would like a quick review of the operation of a particular control or switch, find its page number in the Contents for Section II, and review the "Block Diagram Closeup" (or other sections as you wish) for that control or switch.

As a Sound System Reference

While the bulk of the material in this manual covers topics directly related to your 4000 Mixer and to mixing in general, Section III, "Special Connections, Biamplification and Other Topics," contains useful information on other topics (not directly related to mixing) — topics like "Impedance and Level Watching" (yes, "watching," not "matching!") and "Grounding and Shielding." Also, some parts of Section II contain useful information about sound systems and mixing in general. Browse through the Contents to find topics that interest you.

Section IV, "Examples of Systems Using the Fender 4000 Mixers," shows such typical applications as portable entertainment systems and instrument mixing systems (keyboard mixing, for example). This section should help trigger your own ideas about the virtually limitless uses for your 4000 Mixer.

Table of Contents

Section I: Specifications and Quick Reference Guides

Introduction to These Specifications

We publish these specifications to help you understand the features and performance of your Fender 4000 Series Mixer.

Refer to the "SPECIFICATIONS" section for information on the performance and features of your mixer.

Refer to the "INPUT IMPEDANCE AND LEVEL" and "OUTPUT IMPEDANCE AND LEVEL" charts for information about the various input and output connectors.

The "MAXIMUM VOLTAGE AMPLIFICATION" chart contains information about the "gain" (more accurately "amplification") from any input connector to any output connector. Using the numbers in this chart, you can determine the level at any output connector from a source at any input connector. Just add the dB Amplification given in the chart to the dB level of the source.

This output level assumes that all faders and the Trim control are at their "maximum" positions. To find the output level with faders or Trim at some lower position, subtract the dB number on the fader or Trim control from the sum of the source level and the dB Amplification number. These numbers are approximate and may vary due to slight differences in control accuracy.

The (Not So) Mysterious "dB"

Our ears are pretty amazing instruments. One of their most amazing attributes is their ability to detect both very quiet and very loud sounds. In fact, the loudest sound we can tolerate is something like 10^{12} times as powerful as the quietest sound we can detect. 10^{12} is 1 followed by 12 zeros; that's one million times one million!

Another interesting fact about our hearing is that it is "non-linear." If you listen to an amplified electric guitar and say that it sounds twice as loud to your ears as a non-amplified acoustic guitar, chances are that a sound level meter would show that the amplified electric guitar is actually producing about 10 times as much sound power as the non-amplified acoustic guitar.

One unhappy result of these two facts about our hearing is that it becomes very cumbersome to describe the things we hear with regular numbers. Imagine, for example, using an 11-digit number and multiplying by 20 just to say that one loudspeaker is four times as loud as another!

There ought to be a better way, and there is. It's a numbering system based on ratios and logarithms (don't let those terms scare you) and its units are the "dB" (the "decibel"). The dB system allows us to use small numbers to express very large ratios with small, easy to remember, numbers. That 1 followed by 12 zeros, for example, becomes 120dB. In addition, because the dB system is based on logarithms (you don't have to understand logarithms to use the dB system), it *automatically* compensates for the non-linearity in our hearing. Thus, one way to think of the dB is that it is a very 'natural" numbering system!

Understanding the dB System

Just a few, simple, rules will allow you to understand the dB system well enough to decipher most specifications and to communicate on a reasonably technical basis with other sound system operators.

Rule 1: **The dB system can be used to describe anything we can measure.** That is, the uses of the dB system are not limited to sound pressure levels or electrical watts. We could use the dB system to describe ratios (see Rule 2) of the heights of buildings or the horsepower of automobile engines.

Rule 2: **The dB is always a *ratio*.** That means we cannot use the dB to directly describe the horsepower of a single engine, for example. We can use the dB to describe the ratio of the horsepowers of two engines, however, and in this way, we can *indirectly* use the dB to describe the horsepower of a single engine. To do this, we choose a *reference* engine, say a 1 horsepower engine. We can then describe the horsepower of any other engine as a ratio to the *reference* engine.

Rule 3: **The dB and sound.** In the sound business, we use the dB system to describe *ratios* of sound pressure level, electrical power and voltage. In addition, there are well-recognized references for each of these quantities that allow us to describe actual sound pressure *levels*, and electrical voltage and power *levels*. In other words, we can use the dB to describe *ratios* ("this amplifier is 3 dB more powerful than that amplifier") or to describe actual *levels* ("this loudspeaker is producing 95 dB SPL").

Rule 4a (The dB for power ratios): For electrical power (or any other kind of power), 3dB means a ratio of 2:1 That is, a 100 watt amplifier is +3dB more powerful than a 50 watt amplifier. Or we could say that a 100 watt amplifier is -3dB less powerful than a 200 watt amplifier. The "+" and "-" signs simpiy indicate "more" or "less." Similarly, 6dB means a ratio of 4:1 in power (a 1000 watt amplifier is +6dB more powerful than a 250 watt amplifier) and 10dB means a ratio of 10:1 (a 10 watt amplifier is -10dB less powerful than a 100 watt amplifier). Any ratio can be described in dB, but these three are the most commonly used. If you are mathematically inclined, the equations for any dB usage are given as rule 5.

Adding dB values is the same as *multiplying* their equivalent ratios. We indicated that 3dB indicates a ratio of 2:1 and 6dB indicates a ratio of 4:1. Similarly, 9dB indicates a ratio of 8:1 and 12dB indicates a ratio of 16:1. Another way to look at this is that since 3dB indicates a ratio of 2:1, anytime you add 3dB to something, you multiply it by two (3dB higher than 200 watts is 400 watts). As another example, if 10dB is a ratio of 10:1, 20dB is a ratio of 100:1 (10 times 10) and 30dB is a ratio of 1000:1. 13dB (10dB plus 3dB) would be a ratio of 2:1 times 10:1 or 20:1 (a 200 watt amplifier is 13dB more powerful than a 10 watt amplifier).

Rule 4b (The dB for voltage and SPL ratios): Voltages and SPL (sound pressure level) ratios are slightly different from power ratios. For voltage or SPL, 6dB means a ratio of 2:1 (remember that for power ratios, 3dB is a ratio of 2:1). That is, 20 volts is +6dB higher than 10 volts. Similarly, 12dB means a ratio of 4:1 and 20dB means a ratio of 10:1. By adding these dB values, we multiply their equivalent ratios as before. Thus, 26dB (20dB plus 6dB) would be a ratio of 20:1 (10:1 times 2:1). This difference from power ratios may seem confusing, but it actually makes things simpler when you get into the technical details of the dB system. Fortunately, outside of those technical details, we seldom have to deal with voltage ratios, and we almost never have to deal with actual sound pressure ratios. Thus, the power ratio rule (4a) is the most important one to remember.

Rule 4c (The dB for power levels): Remember that the dB always describes a *ratio* but that we can indirectly describe a *level* by defining a *reference* level. For power, that reference level is one milliwatt (0.001 watt) which is described as "0dBm." Thus, any power level can be stated in terms of so many "dBm." Two milliwatts would be 3dB higher than one milliwatt. Thus, two milliwatts (0.002 watt) is +3dBm. ½ milliwatt (0.0005 watt) would be -3dBm. 10 milliwatts would be +10dBm and so on. You will hardly ever hear the output of a power amplifier measured in dBm but it could be. One watt is +30dBm, thus, 10 watts would be +40dBm and 100 watts would be +50dBm (each of these values was a ratio of 10:1 times the previous one so we just added 10dB). The dBm is often used, however, to describe the output of a low-level or line-level device, especially those devices that are made to drive 600-ohm lines. In this case, the dBm still describes the *power* output of that device. For example, a limiter that will produce +24dBm into 600-ohms is actually producing ¼ watt into that 600-ohms.

Rule 4d (The dB for voltage levels): Again, we need a *reference* voltage if we want to use the dB system to describe a voltage *level* (versus a voltage *ratio*). A common voltage reference is 1 volt. Voltages given in "dBV" are referenced to 1 volt. Thus, +6dBV is 2 volts; -6dB is ½ volt; +20dBV is 10 volts and so on. The dBV terminology is most often used for microphone sensitivity ratings.

Another voltage reference that is becomming more commonly used is 0.775 volt. The reasons for this reference are described in the following section entitled "The 0.775 Volt 0dB Voltage Reference."

Rule 4e (The dB for SPL levels): The *reference* level for sound pressure levels in dB SPL is 0.0002 dynes/cm² which corresponds to 0dB SPL, the so-called "threshold of hearing." That's pretty academic, however, because you will hardly ever hear a sound pressure level expressed in anything *other* than dB SPL!

Rule 5 (The Formulas):You don't have to know these, but, if you are mathematically inclined, studying them will help you understand the dB system.

Power Ratios:
dB = 10 log(P1/P2) (power in watts)

Voltage or SPL Ratios:
dB = 20 log(V1/V2) (voltage in volts)
dB = 20 log(SPL1/SPL2) (SPL in dynes/cm²)

Power Level in dBm:
dBm = 10 log(P/0.001) (power in watts)

Voltage Level in dBV:
dBV = 20 log(V/1) (voltage in volts)

Voltage Level in dB re 0.775 volts:
dB(0.775) = 20 log(V/0.775) (voltage in volts)

SPL in dB SPL:
dB SPL = 20 log(SPL/0.0002) (SPL in dynes/cm²)

An Example of dB Terminology

A quiet concert hall has an ambient noise level (from outside traffic, air conditioning, etc) of about 0.02dynes/cm². A loud rock concert may have peak sound pressure levels in excess of 112.47 dynes/cm². That means that the rock concert is 5,623.41 times as loud as the ambient noise level in the concert hall. Using dB terminology, the quiet concert hall has an ambient noise level of 40dB SPL; the loud rock concert exceeds 115dB SPL and thus the rock concert is 75dB louder than the ambient noise in the empty hall. The dB values are certainly easier to comprehend and work with.

The Concept of Headroom

Headroom is the difference, in dB, between the average and the peak levels in a performance. A speech only, paging system may have as low as 10dB of headroom. A high-quality musical reinforcement system usually has 20dB or more of headroom.

Headroom is one place where the dB terminology can be misleading. For the power amplifiers in the speech system, 10dB of headroom means that a 200 watt/channel Fender 2224 amplifier will operate at 20 watts average output per channel (10dB headroom indicates a ratio of 10:1). At first glance, the 20dB headroom value for the musical system seems like it is only twice the amount of headroom required for the speech system. However, since 20dB indicates a ratio of *100:1*, the Fender 2224 amplifier will operate at only 2 watts average output per channel, compared to the 20 watts average in the speech system! In other words, while the dB system can make it easier to deal with the very large numbers we encounter in audio, we cannot completely neglect those numbers. Especially when we are calculating headroom, amplifier power, *and loudspeaker power capacity*, we must consider the actual numbers and ratios as well as the dB values.

The 0.775 Volt 0dB Voltage Reference

When we use the term "dB" to indicate a voltage *amplification factor* (gain), no reference is implied (this is a *ratio* as described above). Thus, 84dB Voltage Amplification means that if you take 20 times the Log (base 10) of the output voltage divided by the input voltage, the answer will be 84dB.

When we use the term "dB" to indicate a voltage *level* we are implying a 0.775 volt *reference* voltage level. Thus, a voltage level expressed as "+4dB" was calculated by taking 20 times the Log (base 10) of the actual voltage (1.23 volts) divided by the reference (0.775 volt) voltage.

The choice of a 0.775 volt reference was not arbitrary. dBm is a common method of rating *power* levels (as opposed to our *voltage* level ratings). The "0 dBm" reference is one milliwatt (1/1000 watt). If a mixer produces 0 dBm into exactly 600-ohms, the voltage level is 0.775 volts. Thus, for true 600-ohm lines, the dBm power terminology and our dB voltage terminology are equivalent.

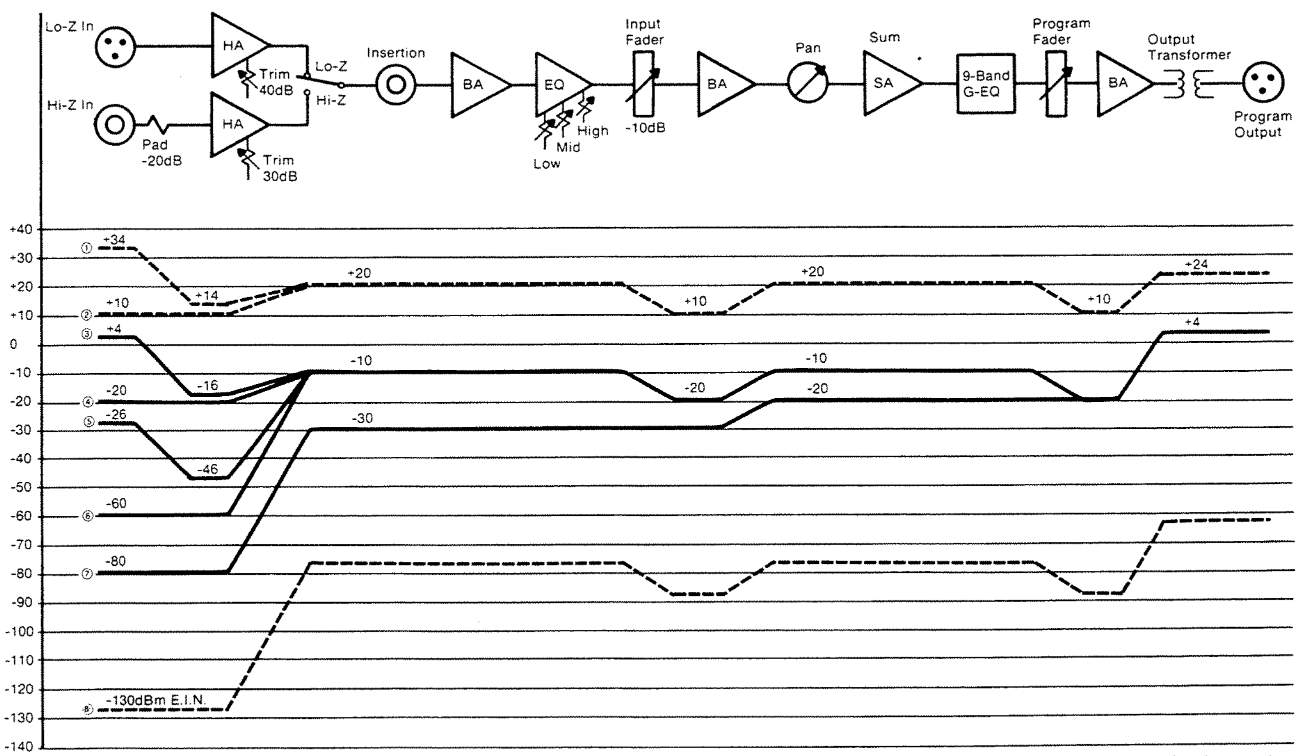
The Gain-Level Diagram

The purpose of this diagram is to express the maximum and nominal voltage levels and the noise floor of each section of the mixer and to show the gain (or loss) between sections.

Once you understand signal flow in the mixer, you can use the GAIN/LEVEL DIAGRAM to help you optimize signal to noise ratios and avoid clipping.

Each section of the GAIN/LEVEL DIAGRAM, from left to right, corresponds to a section of the mixer (sections are shown in the Block Diagrams). The dashed line at the top shows clipping levels in each stage. Below the clipping line, two lines show the level in each section corresponding to maximum and minimum settings of the Trim control. The lower two lines indicate the noise floor in each section for both line-level and mic-level inputs.

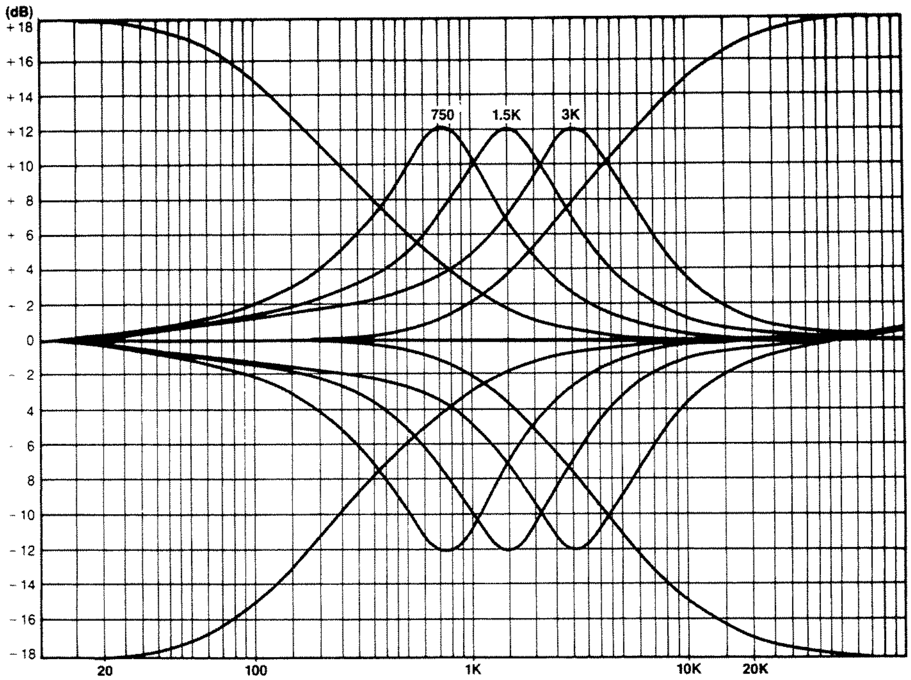
Gain Level Diagram



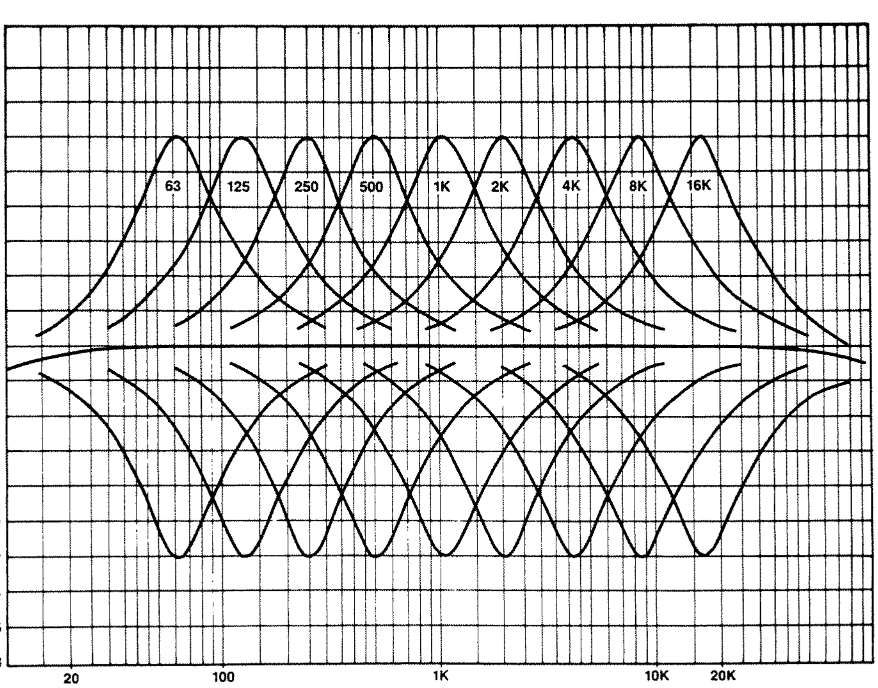
① Hi-Z In Clip Line
② Lo-Z In Clip Line
③ Hi-Z In Nominal Line (Trim at Min)
④ Lo-Z In Nominal Line (Trim at Min)
⑤ Hi-Z In Nominal Line (Trim at Max)
⑥ Lo-Z In Nominal Line (Trim at Max)
⑦ Lo-Z In Max Gain Line (Trim and all Faders at Max)
⑧ Lo-Z In Noise Floor

Note: 0dB reference is 0.775 volt except for -128dBV E.I.N. where 0dBV = 1.00 volt.

Input Channel Equalization Controls



Graphic Equalizers



Specifications, Models 4208, 4212, 4216

FREQUENCY RESPONSE	20 Hz to 20k Hz, +0, -2dB at +24dBm into a 600-ohm load at Program Output jacks.
T.H.D.	Less than 0.2% from 30 Hz to 20k Hz at +24dBm into a 600-ohm load at Program Output jacks, Typically less than 0.05% T.H.D.
NOISE	-130dBm Equivalent Input Noise from 20 Hz to 20k Hz.
SIGNAL TO NOISE RATIO	88dB at Left, Right, Mon 1, Mon 2 Outputs with Master and all input channel faders at minimum.
COMMON MODE REJECTION RATIO	Greater than 70dB at 1000 Hz (external noise rejection).
CROSS TALK	-65dB at 1000 Hz (between input channels).
VOLTAGE AMPLIFICATION	84dB, ±2dB (maximum) at 1000 Hz. Lo-Z Input to Program Output.
MAXIMUM INPUT LEVEL	
Lo-Z Input	Greater than +10dB
Hi-Z Input	Greater than +30dB.
0 VU REFERENCE	+4dB at Program or Monitor or Effect Output jacks.
MAXIMUM OUTPUT LEVEL	+24dB into a 600-ohm load at Program and Monitor Outputs. 2 watts into an 8-ohm load at Left and Right Phones out.
EQUALIZATION	
Input EQ Low	±15dB at 100 Hz, Shelving-Type
Input EQ Mid	±12dB at 750 Hz, 1500 Hz or 3000 Hz (selectable) Peak-Dip-Type
Input EQ High	±15dB at 10k Hz, Shelving-Type
Program 9-Band Graphics	±12dB at 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz 8000 Hz and 16k Hz Peak-Dip-Type
High Pass Filter	80 Hz, 12dB/octave (4216 only).
FADERS	60mm throw, Carbon-type
INDICATORS	
Signal Indicator	Each Input Channel.
Peak Indicator	Each Input Channel. Each VU Meter.
Phantom Power	LED.
INPUTS	See "Input Impedance-Level Chart"
Lo-Z	Balanced (transformerless)
All Others	Unbalanced.
OUTPUTS	See "Output Impedance-Level" Chart
Program Left, Right	Balanced, transformer coupled.
Monitor 1, 2	Balanced, transformer coupled.
All Others	Unbalanced.
MIXING BUSES	L,R Program; Monitor 1,2; Effects (Effects 1 and 2 on 4216).
PHANTOM POWER	+48 volts DC on pins 2 and 3 of each Lo-Z Mic input (pin 1 common) LED "on" indicator and rear panel "Phantom" switch.
CONNECTORS	
Input Lo-Z	3-pin XLR (female) Pin 2 high, Pin 3 low, Pin 1-shield
Insertion (Input)	TRS Phone (tip-ring-sleeve) Tip input, Ring output, Sleeve common (shield)
All Other	TS Phone (tip-sleeve) Tip high, Sleeve low (shield).
POWER CONSUMPTION	120 Volts (±10%), 60 Hz
SAFETY LISTING	UL
DIMENSIONS AND WEIGHT, Model 4208	
Depth	615 mm (24.2 in)
Width	675 mm (26.6 in)
Height	190 mm (7.48 in)
Weight	17.5kg (38.5 lb)
DIMENSIONS AND WEIGHT, Model 4212	
Depth	615 mm (24.2 in)
Width	825 mm (32.5 in)
Height	190 mm (7.48 in)
Weight	21.0kg (46.2 lb)
DIMENSIONS AND WEIGHT, Model 4216	
Depth	695 mm (27.4 in)
Width	975 mm (38.4 in)
Height	195 mm (7.48 in)
Weight	27.5kg (60.5 lb)
FINISH	Simulated Walnut Side Panels, Gun-Metal Grey, Non-Reflecting Face Panel, Color-Coded Controls

0dB is Referenced to 0.775 volts rms.

OUTPUT IMPEDANCE AND LEVEL, MODELS 4208, 4212, 4216

OUTPUTS IMPEDANCE	LOAD TYPE NOMINAL	ACTUAL OUTPUT MAX BEFORE CLIP	OUTPUT VOLTAGES	
Insertion Send L,R	10k-ohm Line	100-ohms	-10dB (245 mv)	+20dB (7.75volt)
Program Out L,R	600-ohm Line	170-ohms	+ 4dB (1.23volt)	+24dB (12.3volt)
Effect Out 1,2*	10k-ohm Line	100-ohms	+ 4dB (1.23volt)	+20dB (7.75volt)
Monitor Out 1,2*	600-ohm Line	100-ohms	+ 4dB (1.23volt)	+24dB (12.3volt)
Phones Out L,R	8-ohm headphone		+14dB (3.88volt)	+14dB (3.88volt)

*The 4208 and 4212 have only one Effect channel.

INPUT IMPEDANCE AND LEVEL, MODELS 4208, 4212, 4216

INPUTS	SOURCE TYPE	ACTUAL INPUT IMPEDANCE	INPUT VOLTAGES NOMINAL	MAX BEFORE CLIP
Insertion (Input)	Lo or Hi-Z Low-level Line	21k-ohms	-10dB (245 mv)	+20dB (7.75volt)
Hi-Z In	Hi-Z Mic or Line Instrument Direct	20k-ohms	-26dB (38.8 mv)	+34dB (38.8volt)
Lo-Z In	Lo-Z Mic or Low-level 600-ohm Line	10k-ohms (bal) -60dB	+10dB (0.775 mv)	(2.45volt)
L,R Direct In (Program)	Lo or Hi-Z Line	110k-ohms	+ 4dB (1.23volt)	+34dB (38.8volt)
Insertion Return (L,R)	Lo or Hi-Z Line	21k-ohms	-10dB (245 mv)	+20dB (7.75volt)
Effect 1,2* Return	Low-level Line	33k-ohms	-20dB (77.5 mv)	+10dB (2.45volt)
Effect 1,2* Direct	Lo or Hi-Z Line	110k-ohms	+ 4dB (1.23volt)	+34dB (38.8volt)
Aux In 1,2	Lo or Hi-Z Low-level Line	33k-ohms	-20dB (77.5 mv)	+10dB (2.45volt)
Monitor 1,2 Direct In	Lo or Hi-Z Line	110k-ohms	+ 4dB (1.23volt)	+34dB (38.8volt)

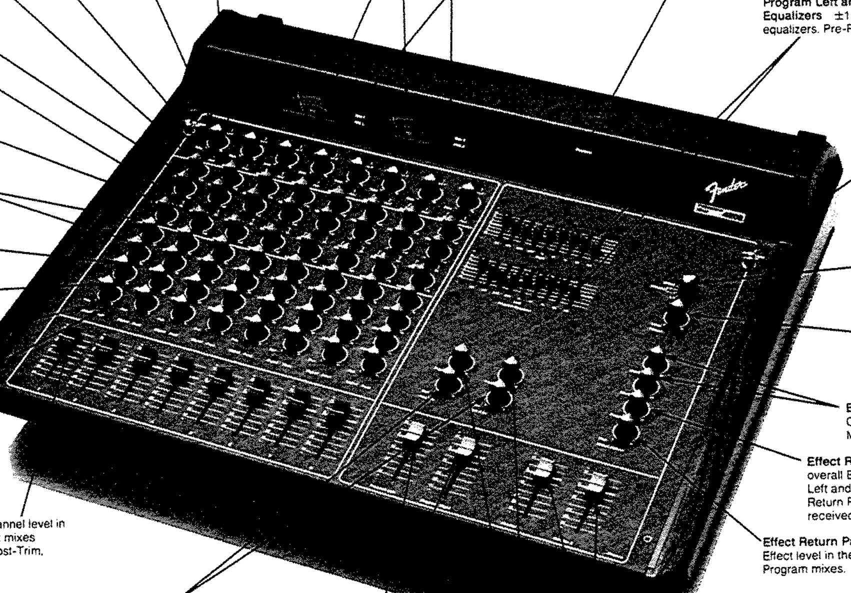
*The 4208 and 4212 have only one Effect channel.

MAXIMUM VOLTAGE AMPLIFICATION, MODEL 4208, 4212, 4216 (20logVo/Vi)

From This Input	To Insertion (Inp Chan)	To L,R Program Output	To L,R Insertion Send	To 1,2* Effect Output	To 1,2 Monitor Output
Insertion (Inp Chan)		34dB	10dB	24dB	24dB
Hi-Z In	16dB	50dB	16dB	40dB	40dB
Lo-Z In	50dB	84dB	50dB	74dB	74dB
Program Direct L,R		10dB	-14dB		
Insertion Return L,R		24dB			
Effect 1,2** Return		34dB	10dB		34dB
Effect 1,2* Direct In				0dB	
Aux In 1,2		34dB	10dB		10dB
Monitor Direct 1,2					0dB

*4208 and 4212 have only one Effect channel.

Quick Reference Guide, Model 4208



Trim Preamp gain. Input Channel preamplifier gain (40dB range on Lo-Z input; 30dB range on Hi-Z input) control. With Fader down and EQ Controls at "0," set Trim so that Peak LED lights occasionally on program peaks. Then set Fader (and Pan) to control Program Left and Right mix levels. Pre-EQ, pre-Fader.

(Left and Right) VU Meters and Peak LEDs Display level of Program mixes or Monitor mixes. 0 VU = +4dB at the Left or Right Program Output jack or Monitor 1 or 2 Output jack. The VU Meter's Peak LED indicates the presence of high-level peak signals in the output channel. This Peak LED should light only very occasionally on normal program peaks.

Pre/Post Insertion (Switch) Places any device connected to Insertion In/Out jack either before (Pre) or after (Post) the Input Channel EQ section.

Hi-Z/Lo-Z (Switch) Selects Hi-Z input or Lo-Z input to Input Channel.

Program and Monitor Left and Right VU Switches Allows VU Meters to display level of Program or Monitor mixes.

Phantom (LED) When "on," the Phantom LED indicates the presence of +48 volt phantom power on all Lo-Z microphone input connectors.

High High-frequency EQ control. ±15dB, 10k Hz, shelving-type high-frequency EQ control. Post-Trim, pre-Fader.

Mid (Control) Midrange EQ control. ±12dB, selectable-frequency, peak/dip-type midrange EQ control. Post-Trim, pre-Fader.

Mid (Switch) Selects center frequency for Mid Control (750 Hz, 1.5k Hz, 3k Hz).

Low Low-frequency EQ control. ±12dB, 100 Hz, shelving type, low-frequency EQ control. Post-Trim, pre-Fader.

Program Left and Right Equalizers ±15dB, 9-Band Graphic equalizers. Pre-Program Faders.

Monitor 1 and 2 Control input Channel level in the Monitor 1 and 2 mixes. Post-Trim, post-EQ, pre-Fader. (Post-Fader option may be installed by a qualified service technician.)

Panel Light Connector BNC type connector supplies 12 volts AC for Fender 12" model 12G gooseneck panel light (optional).

Effect Controls Input Channel level in the Effect mix. Post-Trim, post-EQ, post-Fader.

Phones (Select Switch) Allows phones to receive Program Left and Right (stereo), Monitor 1 (mono), Monitor 2 (mono) or Effect (mono) signals.

Pan Controls relative level of Input Channel in Left and Right Program mixes. Post-Trim, post-EQ, post-Fader.

Phones (Volume Control) Controls the volume level of both left and right channels of a set of 8-ohm headphones connected to the Phones jack.

Signal (LED) Turns "on" when low-level signal is present at either the Lo-Z or Hi-Z inputs. Stays on at higher input levels.

Effect Return Monitor 1, Monitor 2 Control relative Effect level in the Monitor 1 and Monitor 2 mixes.

Peak (LED) Not a clipping indicator. Peak LED lights when high-level signal is present at either the Lo-Z or Hi-Z inputs. Peak LED should light only very occasionally on normal program peaks. Use in conjunction with Trim control to set Input Channel gain. Post-Trim, post-EQ, pre-Fader.

Effect Return Program Controls overall Effect mix level fed to Program Left and Right mixes through Effect Return Pan control. Effect mix is received from Effect Return jack.

Fader Controls Input Channel level in the Program Left and Right mixes through the Pan control. Post-Trim, post-EQ.

Effect Return Pan Controls relative Effect level in the Left and Right Program mixes.

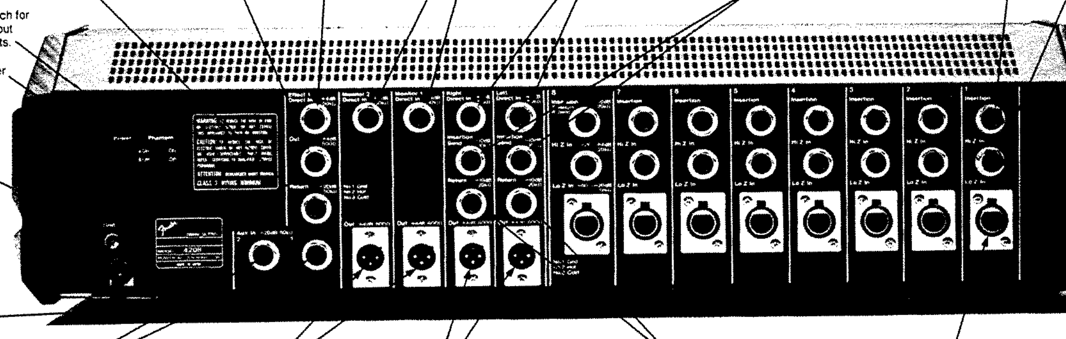
Aux In 1,2 Pan Controls relative level of Aux in device in Program Left and Right mixes.

Phones (Headphone Jack) Accepts a set of 8-ohm stereo headphones (2-watts/channel). ¼" "T/R/S" (tip/ring/sleeve) phone jack.

Program Left and Right (Faders) Control overall Left and Right Program mix levels at Left and Right Program out jacks. Post-EQ (Graphic Equalizer).

Monitor 1 and 2 (Faders) Control overall Monitor mix levels fed to Monitor 1 and 2 Out. Post EQ (Graphic Equalizer).

Aux In 1,2 Controls auxiliary device (connected to PGM Aux In jack) level in the Program Left and Right mixes through Aux In Pan control.



Effect Direct In Accepts +4dB, high or low-impedance unbalanced source. Direct feed to Effect mix. ¼" phone jack.

Monitor 1,2 Direct In Accept +4dB, high or low-impedance unbalanced source. Direct feeds to Monitor 1 and 2 mixes (pre-Monitor 1 and 2 faders). ¼" phone jacks.

Insertion jack Accepts a -10dB auxiliary device into the Input Channel. ¼" "T/R/S" (tip/ring/sleeve) phone jack with ring = send to auxiliary device; tip = return from auxiliary device and sleeve = common/ground.

Effect Out +4dB, unbalanced Effect mix output. Will drive 10k-ohm or higher impedance loads. ¼" phone jack.

Program L,R Direct In Accept +4dB, high or low-impedance unbalanced sources. Direct feeds to Left and Right Program mixes (pre-Program fader). ¼" phone jacks.

Effect Return Accepts -20dB, unbalanced, high or low-impedance source. Direct input to Program L,R and Monitor 1,2 mix (pre-Effect Return and Effect Return Monitor 1,2 controls). ¼" phone jack.

Insertion Send Pre-Graphic-EQ, -10dB, 10k-ohm output for auxiliary devices. ¼" phone jack.

Hi-Z In (Input Channel) Accepts high-impedance microphone or other 26dB to +4dB (line-level), high or low-impedance unbalanced source.

Phantom (Switch) On/off switch for +48-volt phantom power to all Input Channel Lo-Z (microphone) inputs.

Power (Switch) Primary AC Power (on/off) switch.

GND (Screw) Chassis ground screw. Use to connect the ground wire from a phono turntable or other device that requires a chassis ground other than that provided by the third prong on its AC plug.

AC Power Cable Requires 120-volt AC, 60 Hz, grounded outlet. Do not defeat ground pin. See Contents for further safety information.

Aux In 1,2 Accept -20dB, unbalanced, high or low-impedance source. Direct input to Left and Right Program mixes through Aux In Program and Pan controls. ¼" phone jack.

Monitor 1,2 Out +4dB, balanced, transformer-coupled primary Monitor outputs. Will drive 600-ohm or higher impedance loads. XLR (male) type connector with Pin 2 = "hot," Pin 3 = "cold" and Pin 1 = common/ground/shield.

Program L,R Output +4dB, balanced, transformer-coupled Program outputs. Will drive 600-ohm or higher impedance loads. XLR type (male) connector with Pin 2 = "hot," Pin 3 = "cold" and Pin 1 = common/ground/shield.

Insertion Return Pre-Graphic-EQ, -10dB input for auxiliary devices. ¼" phone jack. Normaled to Insertion Send. Inserting a ¼" phone plug into the Insertion Return jack interrupts flow of Program signal from the Insertion Send jack to the Insertion Return jack.

Lo-Z In (Input Channel) Accepts a low-impedance microphone or other -60dB to -10dB (mic-level), low-impedance, balanced source. 3-pin female XLR connector (2-high, 3-low, 1-shield). Switchable +48-volt phantom power on pins 2 and 3 with pin 1 ground.

Quick Reference Guide, Model 4212

Trim Preamp gain. Input Channel preamplifier gain (40dB range on Lo-Z Input; 30dB range on Hi-Z Input) control. With Fader down and EQ Controls at "0," set Trim so that Peak LED lights occasionally on program peaks. Then set Fader (and Pan) to control Program Left and Right mix levels. Pre-EQ, pre-Fader.

(Program L,R; Monitor 1,2) VU Meters and Peak LEDs Display level of Program mixes and Monitor mixes. 0 VU = +4dB at the Left or Right Program Output jack or Monitor 1 or 2 Output jack. The VU Meter's Peak LED indicates the presence of high-level peak signals in the output channel. This Peak LED should light only very occasionally on normal program peaks.

Pre/Post Insertion (Switch) Places any device connected to Insertion In/Out jack either before (Pre) or after (Post) the Input Channel EQ section.

Hi-Z/Lo-Z (Switch) Selects Hi-Z input or Lo-Z input to Input Channel.

High High-frequency EQ control. ±15dB, 10k Hz, shelving-type high-frequency EQ control. Post-Trim, pre-Fader.

Mid (Control) Midrange EQ control. ±12dB, selectable-frequency, peak/dip-type midrange EQ control. Post-Trim, pre-Fader.

Mid (Switch) Selects center frequency for Mid Control (750 Hz, 1.5k Hz, 3k Hz).

Low Low-frequency EQ control. ±12dB, 100 Hz, shelving type, low-frequency EQ control. Post-Trim, pre-Fader.

Monitor 1 and 2 Control Input Channel level in the Monitor 1 and 2 mixes. Post-Trim, post-EQ, pre-Fader. (Post-Fader option may be installed by a qualified service technician.)

Effect Controls Input Channel level in the Effect mix. Post-Trim, post-EQ, post-Fader.

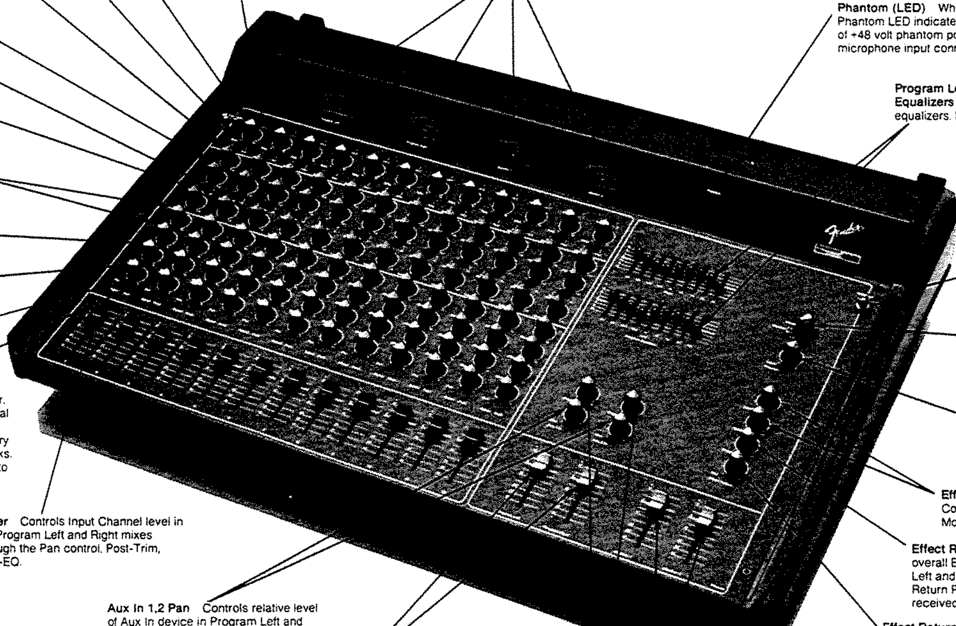
Pan Controls relative level of Input Channel in Left and Right Program mixes. Post-Trim, post-EQ, post-Fader.

Signal (LED) Turns "on" when low-level signal is present at either the Lo-Z or Hi-Z inputs. Stays on at higher input levels.

Peak (LED) Not a clipping indicator. Peak LED lights when high-level signal is present at either the Lo-Z or Hi-Z inputs. Peak LED should light only very occasionally on normal program peaks. Use in conjunction with Trim control to set Input Channel gain. Post-Trim, post-EQ, pre-Fader.

Phantom (LED) When "on," the Phantom LED indicates the presence of +48 volt phantom power on all Lo-Z microphone input connectors.

Program Left and Right Equalizers ±15dB, 9-Band Graphic equalizers. Pre-Program Faders.



Panel Light Connector BNC type connector supplies 12 volts AC for Fender 12" model 12G gooseneck panel light (optional).

Phones (Select Switch) Allows phones to receive Program Left and Right (stereo), Monitor 1 (mono), Monitor 2 (mono) or Effect (mono) signals.

Phones (Volume Control) Controls the volume level of both left and right channels of a set of 8-ohm headphones connected to the Phones jack.

Effect Return Monitor 1, Monitor 2 Control relative Effect level in the Monitor 1 and Monitor 2 mixes.

Effect Return Program Controls overall Effect mix level fed to Program Left and Right mixes through Effect Return Pan control. Effect mix is received from Effect Return jack.

Effect Return Pan Controls relative Effect level in the Left and Right Program mixes.

Fader Controls Input Channel level in the Program Left and Right mixes through the Pan control. Post-Trim, post-EQ.

Aux In 1,2 Pan Controls relative level of Aux In device in Program Left and Right mixes.

Program Left and Right (Faders) Control overall Left and Right Program mix levels at Left and Right Program out jacks. Post-EQ (Graphic Equalizer).

Monitor 1 and 2 (Faders) Control overall Monitor mix levels fed to Monitor 1 and 2 Out. Post EQ (Graphic Equalizer).

Phones (Headphone Jack) Accepts a set of 8-ohm stereo headphones (2-watts/channel). ¼" "T/R/S" (tip/ring/sleeve) phone jack.

Aux In 1,2 Controls auxiliary device (connected to PGM Aux In jack) level in the Program Left and Right mixes through Aux In Pan control.

Effect Direct In Accepts +4dB, high or low-impedance unbalanced source. Direct feed to Effect mix. ¼" phone jack.

Monitor 1,2 Direct In Accept +4dB, high or low-impedance unbalanced source. Direct feeds to Monitor 1 and 2 mixes (pre-Monitor 1 and 2 faders). ¼" phone jacks.

Effect Out +4dB, unbalanced Effect mix output. Will drive 10k-ohm or higher impedance loads. ¼" phone jack.

Program L,R Direct In Accept +4dB, high or low-impedance unbalanced sources. Direct feeds to Left and Right Program mixes (pre-Program fader). ¼" phone jacks.

Insertion jack Accepts a -10dB auxiliary device into the Input Channel. ¼" "T/R/S" (tip/ring/sleeve) phone jack with ring = send to auxiliary device; tip = return from auxiliary device and sleeve = common/ground.

Effect Return Accepts -20dB, unbalanced, high or low-impedance source. Direct input to Program L,R and Monitor 1,2 mix (pre-Effect Return and Effect Return Monitor 1,2 controls). ¼" phone jack.

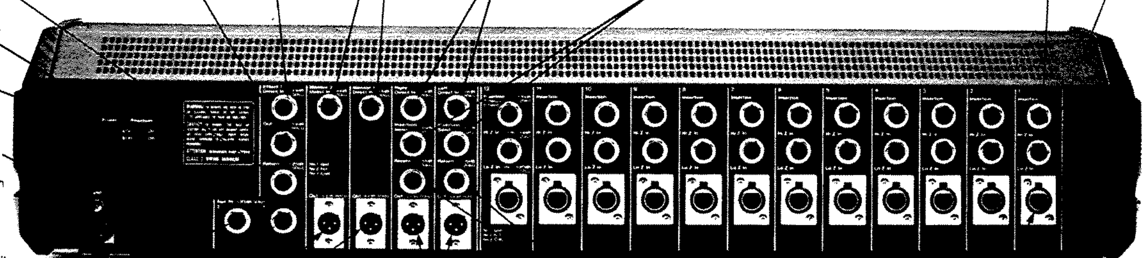
Insertion Send Pre-Graphic-EQ, -10dB, 10k-ohm output for auxiliary devices. ¼" phone jack.

Hi-Z In (Input Channel) Accepts high-impedance microphone or other -26dB to +4dB (line-level), high or low-impedance unbalanced source.

Phantom (Switch) On/off switch for +48-volt phantom power to all Input Channel Lo-Z (microphone) inputs.

Power (Switch) Primary AC Power (on/off) switch.

GND (Screw) Chassis ground screw. Use to connect the ground wire from a phono turntable or other device that requires a chassis ground other than that provided by the third prong on its AC plug.



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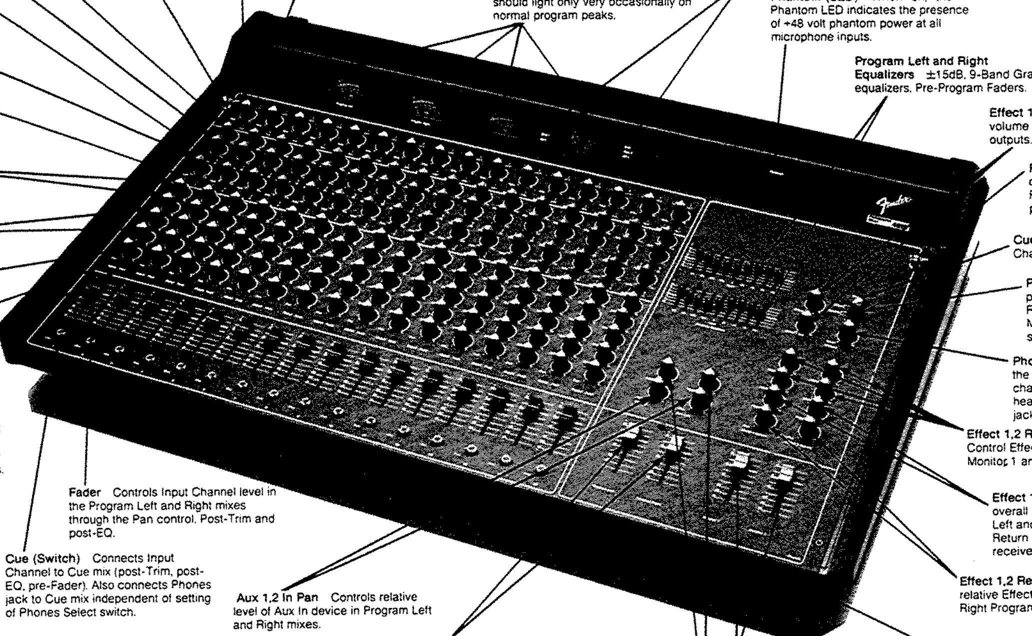
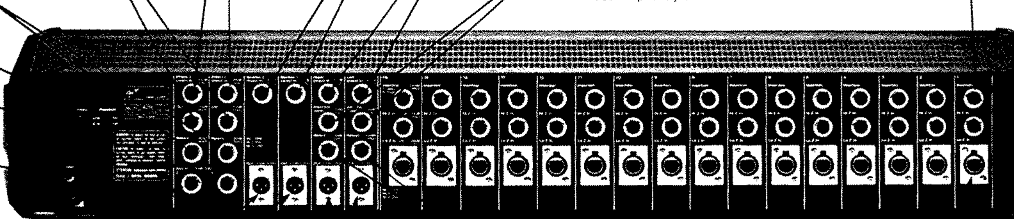
Lo-Z In (Input Channel) Accepts a low-impedance microphone or other -60dB to -6dB (mic-level), low-impedance, balanced source. 3-pin female XLR connector (2-high, 3-low, 1-shield). Switchable +48-volt phantom power on pins 2 and 3 with pin 1 ground.

Quick Reference Guide, Model 4216





Understanding Block Diagrams

The *schematic* diagram of your Fender 4000 Mixer shows every integrated circuit, every resistor, every connection. This kind of detail is useful to a repair technician, but it can actually get in the way of an understanding of the operation of the Mixer *from the user's point of view.*

A block diagram removes all the un-necessary detail from the schematic and leaves only that necessary to understand the way the Mixer operates.

Each section of the Mixer block diagram represents some important function. For example, the triangles represent pre-amplifiers and line amplifiers. The rectangles represent the graphic equalizers. The jagged lines with an arrow through them are controls: pan controls, tone controls and so on. A small vertical rectangle with an arrow through it is a fader. The lines connecting all of these devices represent real wires (or traces on a printed circuit board) inside the Mixer and the vertical lines near the center of the block diagram represent the mix buses (you can always tell a mix bus because it has a large number of inputs connected to it). Additional symbols represent your Mixer's switches, LEDs, VU Meters and input and output jacks.

Let's follow a signal through the block diagram. Start at the Lo-Z input. Notice that, even though your Mixer has at least eight inputs, only one is shown. This simplifies the block diagram considerably but doesn't reduce its usefulness at all (all the inputs are the same anyway). Just past the Lo-Z input jack, you see a vertical line indicating the presence of phantom power on that jack. Then the signal flows through a preamplifier stage. In this preamplifier stage is the Trim control.

From this preamplifier stage, the signal flows through the Insertion switch and, depending on the position of that switch, the signal then flows through the Input Channel Equalization Controls and then out to the Insertion jack (the "post" position of the Insertion switch) or through the Insertion jack and then through the Equalization Controls (the "pre" position of the Insertion switch). If an external device is connected to the Insertion jack, the signal will flow through it regardless of the position of the Insertion switch. Notice also that the Signal and Peak LEDs are located at the output of the Equalization Control section *before the Input Channel fader* which means that they are not affected by the fader.

Now, the signal flows through the fader, through a "buffer amplifier" stage used to isolate the fader from the Pan and Effect controls and through the Pan control to the Program Left and Right mix buses. The signal splits just before the fader to feed the Monitor 1 and Monitor 2 controls which feed the Monitor 1 and Monitor 2 mix buses. Since the signal feeds the Monitor controls *before* it passes through the Input Channel fader, the Monitor controls are "pre-fader," that is, they are not affected by the fader. The Effect control, (Effect 1 and 2 on the 4216) on the other hand, comes after the fader and is therefore "post-fader" (it is affected by the position of the fader). The Monitor controls can be changed to "post-fader" by changing the location of a jumper on the printed circuit board (this modification *must* be performed by a qualified service technician).

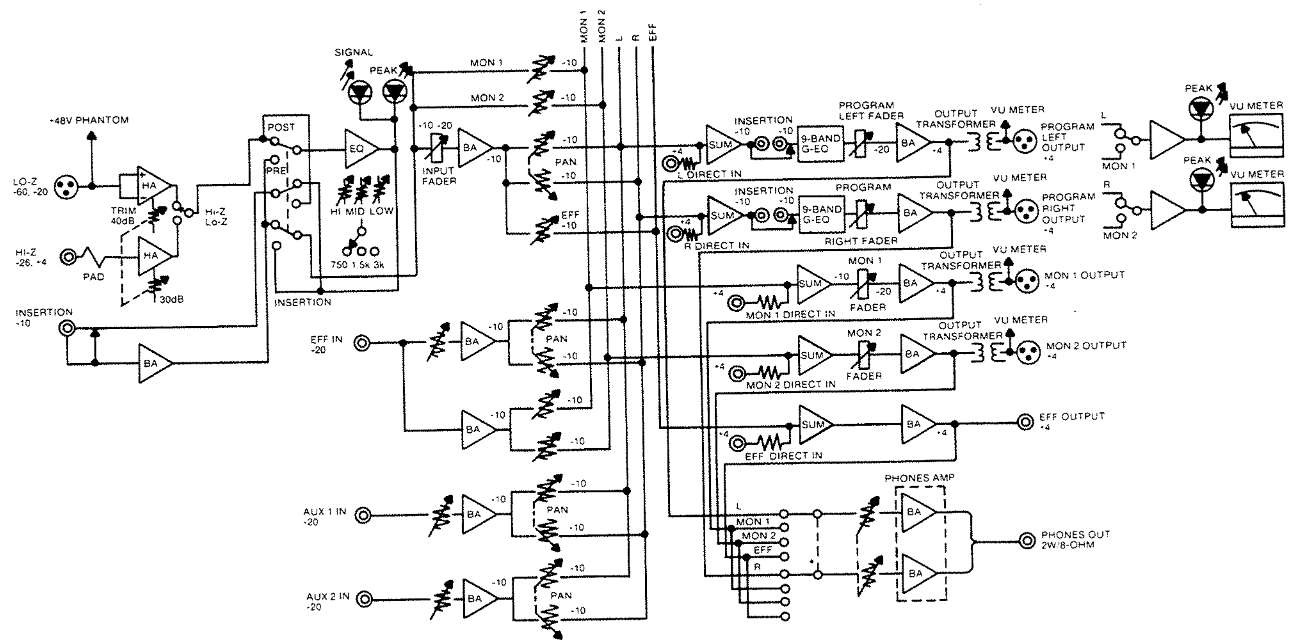
After the Program mix buses, the signal flows through a "summing amplifier." This summing amplifier performs the duty of mixing together the signals from all the Input Channels while keeping them from affecting each other. The Direct In jack is also connected to the input of this summing amplifier stage.

Next, the signal flows through the Insertion Send and Return jacks (and through any device connected between these jacks) and into the Graphic Equalizer. After the Graphic Equalizer, the signal flows through the Program fader which could be called the "master" fader for the entire mixer.

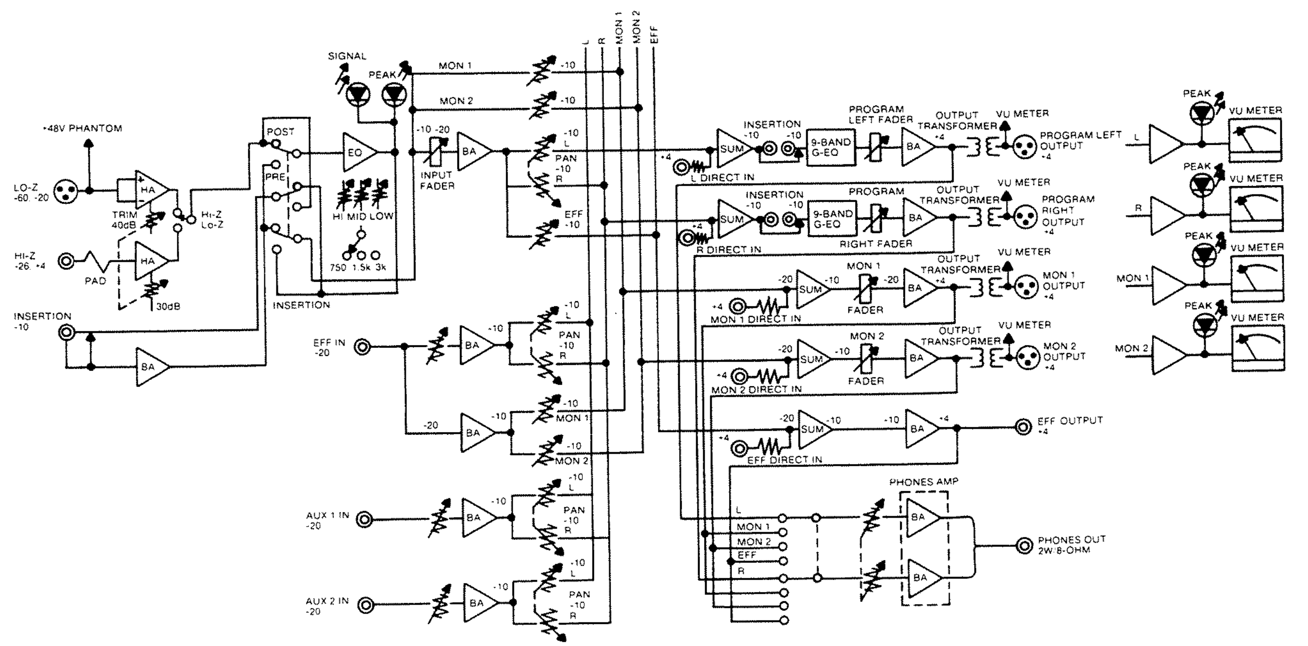
After the Program fader, the signal flows through another buffer amplifier and then, through the output transformer, to the Program Output. The VU Meter is connected to the Program Output so that it directly reads the actual output level.

After following the signal flow through the Mixer, you can see how valuable the block diagram can be. As you learn to read the block diagram of your 4000 Mixer, and, indeed, your entire system block diagram, you will begin to think in terms of the flow of signals through your system. Experienced mixer operators often keep a copy of the block diagram close at hand at all times to remind them of the way the various parts of their system operate and interact with each other. This kind of signal-flow understanding is a very important step in becoming a skilled sound system operator.

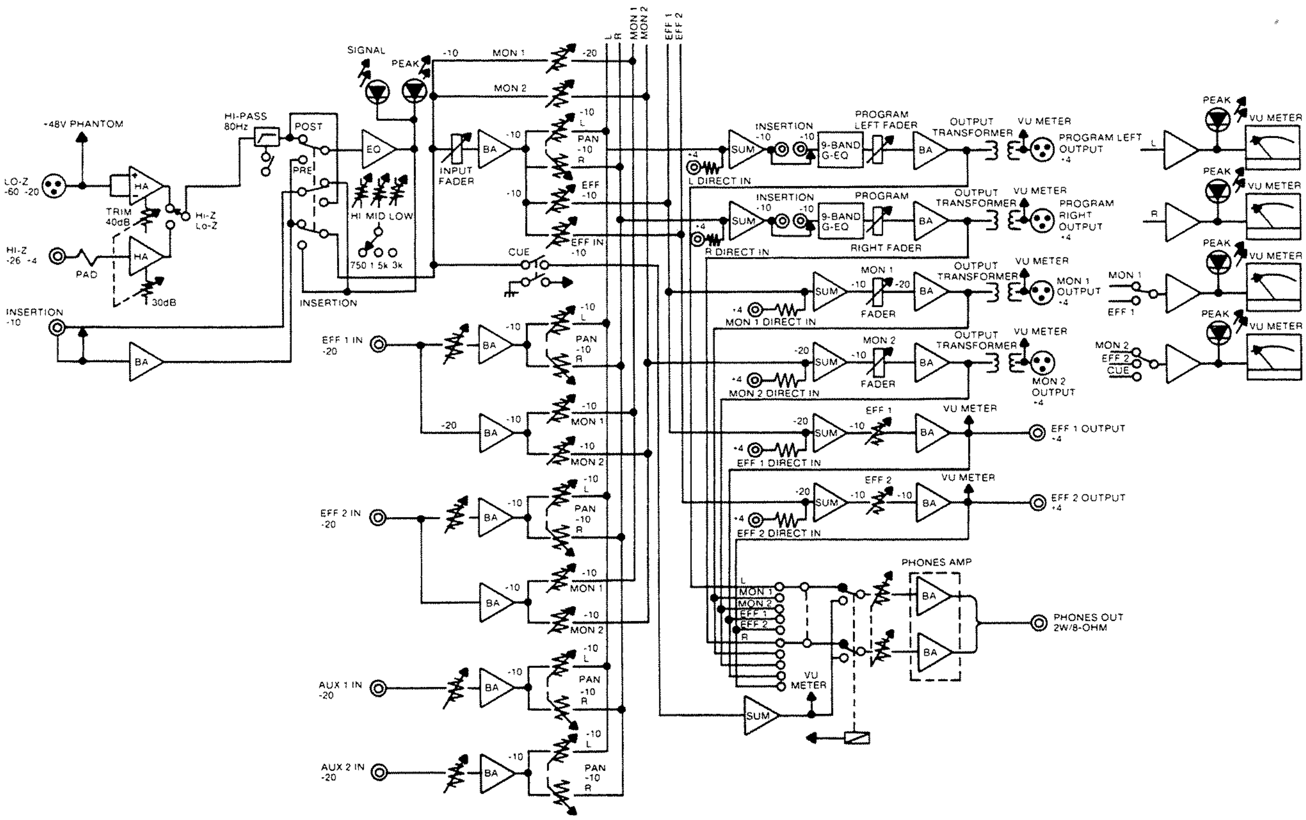
Block Diagram, Model 4208



Block Diagram, Model 4212



Block Diagram, Model 4216



Section II: How to Use Your 4000 Mixer, a Self-Teaching Guide

The Artistry of Mixing

The sound system operator usually has a title having something to do with "technical operations" or "sound crew" or some other title implying behind-the-scenes status. But the sound system operator also deserves to be recognized as an artist — as much an artist, in fact, as the musicians or performers on stage.

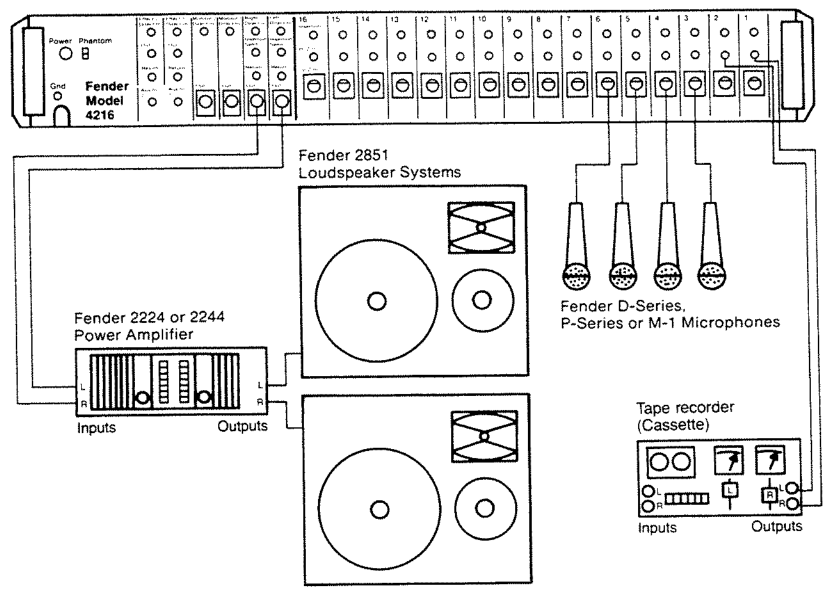
At one time, a performance of any type had to be held in a room (or outdoor area) small enough that the performance could be properly heard by everyone in the (small) audience. That just isn't true anymore. *Most* performances now depend on some type of sound equipment either for sound reinforcement or for sound effects or both. In other words, *the sound system has become an integral part of the performance.* In fact, many performances simply couldn't be held without a sound system.

Knowing this, the sound system operator faces the responsibility of carrying the audio portion of a performance to everyone in the audience. The orchestra balance, once solely the responsibility of the orchestra conductor, is now in the hands of the sound system operator. The tonal character of an instrument, once controlled solely by the musician, is now controlled by the musician *and* the sound system operator. The quality and intelligibility of a voice, once the exclusive responsibility of the vocalist, now depends a great deal on the vocalist's microphone technique *and the abilities of the sound system operator.*

In brief, the sound system operator now shares a significant portion of the *artistic* responsibility for a performance (and that can be a "performance" of any kind, from a live musical drama to a rock concert to a guest speaker at your place of worship).

As you learn to use your Fender 4000 Series Mixer, you will find that it enhances your capabilities and helps you carry out those artistic responsibilities. For that reason, in this manual, we recognize your artistic responsibilities and we comment on the artistic as well as the technical nature of the various connections and controls.

Setup for the Exercises



The Exercises

The Purpose

These "Exercises" to allow you to learn how to use your Mixer's controls and switches and to begin to appreciate the things you can do during an actual performance. And, even though that "performance" may be anything from a large outdoor rock concert to a special choir service at your place of worship, learning the controls and switches now will get you past the "mechanics" stage (what happens when I turn down the "Mid" control?) and farther towards the "artist" stage (how can I improve the vocal quality of that nasal-voiced singer?).

Just reading through these exercises will help you understand your 4000 Mixer, but we encourage you to actually *perform* them if it's at all possible. Real, "hands-on" experience is, *by far*, the best way to learn how to do almost anything. And, you should get that experience *before* you are required to work with an actual performance. The time to learn how to control a skid in your automobile is *before* you have to drive down an icy highway! Likewise, the time to learn how to control feedback or distortion in your sound system is *before* the actual performance takes place!

The Site

There's no reason why these exercises can't be done at home, in your living room. It's possible, of course, that you may want to set up a pair of loudspeakers, one or two microphones and several pieces of external electronics and that you may want to try out the system at higher than living-room sound levels! If your living room doesn't give you the required space or your neighbors (or family) won't put up with the sound levels, we suggest that you practice "on-site," that is, wherever your "performance" will take place.

The Mixer

In these next few sections, we'll discuss the controls and features of the 4216. If you have a 4212 or 4208, you have most of the same controls and features as the 4216 (just fewer Input Channels).

The Equipment

Your Fender 4000 Series Mixer includes just about every piece of electronics you need to perform these exercises. You just add the sources (microphones, etc), a power amplifier (like the Fender 2224 or 2244) and loudspeakers!

You should have at least one microphone, preferably of the type you'll be using "on the job." If you'll be using several types of microphones, try to get one of each type for your exercise sessions.

For your musical sources, use a cassette (or reel-to-reel) machine, preferably a high-quality, stereo player/recorder like you would use in a home stereo system. You may need a pair of "RCA/phono jack to ¼" phone plug" adapters. Ask your Fender Dealer about these adapters.

Get a collection of tapes; search out some with strong solo instruments (and voices). We'll pretend that these tapes are live instruments (which is the reason you should look for lots of good solo passages). Also prepare a tape of the voice of someone you know very well and talk with often (your spouse or a close friend is an excellent choice). If that person sings, ask them to sing. If they feel shy at being recorded, have them read from a book or newspaper. The idea is to get the chance to hear how the controls on your Fender Mixer affect the sound of a voice that you know very well. Your own voice, by the way, is a very *poor* choice for this test! (Remember how *foreign* your own voice sounded the first time you heard it on a tape recorder!)

Imagine how the microphone will be used in a live performance and ask your friend to duplicate those conditions as much as possible. For example, if you will be mixing a live musical performance, the performers will most likely hold a microphone close to their mouths and sing loudly. Thus, you should ask your friend to do the same (watch out for the increased bass response in a cardioid microphone used up close).

Choose a pair of full-range loudspeakers with rated power handling capacity adequate to accept the full output of your power amplifier (220 watts/channel/4-ohms for the Fender 2224; 440 watts/channel/4-ohms for the Fender 2244). It's possible that, for this practice session, you could get along with a pair of lower-powered loudspeakers, like your home stereo speakers. But, if the system unexpectedly goes into feedback (howling), the full output of your power amplifier (and sometimes even more than the amplifier's full rated output) will be sent to your loudspeakers! In other words, *beware!*

If you expect to be using any auxiliary electronic effects devices, such as an external reverb or a compressor/limiter, bring them to your practice site, too.

Making the Connections

Connect the tape recorder's outputs to the Channel 1 and 2 Hi-Z inputs on your Fender Mixer.

Connect the microphones to remaining Input Channels using the Lo-Z inputs for low-impedance microphones and the Hi-Z inputs for high-impedance microphones. If you have a low-impedance microphone with a ¼" phone plug connector, you must use an adapter to connect it to the Lo-Z input. Similarly, if you have a high-impedance microphone with a 3-pin "XLR" type connector, you must use an adapter to connect it to the Hi-Z input. Your Fender Dealer, again, is the most likely source for these adapters. If the adapters come unwired, see the section of this manual entitled "Connectors and Cabling."

If you have any auxiliary devices, you may wish to delay connecting them until the discussion of the Insertion feature and the Effects mix.

Connect the Mixer's AC Power Cable (and the power cables of your auxiliary equipment and power amplifier) to a grounded 120 volt, 60 Hz outlet. Do not defeat the third pin (ground) on the AC Power Cable. If you are using your Fender Mixer outside the USA, confirm that the AC Power voltage, current and frequency are correct for *all* of your electronic equipment.

The Initial Control Setup

Before turning on the AC Power Switch, set all faders (slide volume controls) at their "infinity" (fully down) positions. Set all pan controls and tone controls at their "center" positions (center the slider type graphic equalizer controls, too). Set all other rotary type volume controls (Effect, Monitor, Trim, Program) at their "0" positions (fully counterclockwise). Set all "Cue" buttons in their "off" (up) position. Don't worry about the settings of the other front panel switches for now.

Turning It On

Some types of audio electronics produce a sharp "thump" when they are first turned on. This is called a "turn on transient." some audio devices also produce a similar noise when they are turned off (a "turn-off transient"). While these transients are usually harmless, in some cases, they can reach high levels and represent a danger to your loudspeakers. To protect against such loudspeaker damage, it's a good idea to turn on your mixer and all other low-level and line-level electronics first, *and then turn on your power amplifier.* This allows any turn-on transients from the low-level and line-level electronics to occur at a time when they cannot reach your loudspeakers. When the performance is finished, *turn off the power amplifier before turning off the rest of the system* to keep turn-off transients out of your loudspeakers.

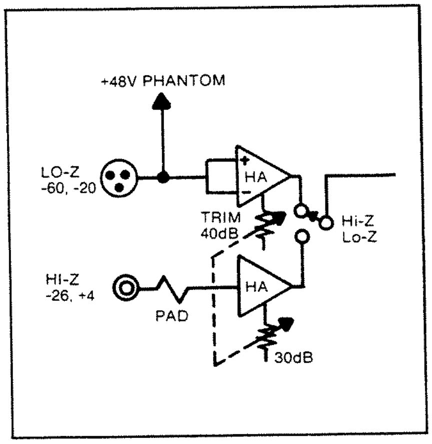
Fender 2224 and 2244 Power Amplifiers include a delay relay which helps prevent turn-on and turn-off transients from reaching your loudspeakers. Other power amplifiers, however, may not include such relays and may even contribute their own turn-on and turn-off transients. If you are uncertain about the effects of turn-on and turn-off transients in your system, it may be a good idea to unplug your loudspeakers when turning the system on or off.

If you're ready for the following "Exercises," make sure your power amplifier is "off" and turn on your Mixer. (The Mixer's AC Power Switch is located next to the AC Power Cable on the rear panel.) Turn on any auxiliary equipment (watch the VU Meters, and LEDs on your 4200 Mixer to see the turn-on transients of the auxiliary equipment) and then turn on your power amplifier.

The "Phantom" Switch

Block Diagram Closeup

+48 volt Phantom power is supplied to pins 2 and 3 of all Lo-Z In connectors with Pin 1 common. The "Phantom" switch (rear panel, near AC Power switch) turns the Phantom power on and off for all Lo-Z In connectors.



An Explanation of Phantom Power

Your Fender 4000 Series Mixer provides "phantom power," that is, the DC power required for most condenser microphones such as the Fender M1 and P1, (some electret condenser microphones, such as the Fender M1, also run on batteries). If *any* of your chosen microphones require phantom power, turn this switch "on." Don't worry, the phantom power supply won't harm any non-condenser microphones, nor will it cause any performance changes. In other words, you can leave the Phantom switch "on" all the time if you want to.

Adapters and Phantom Power

You might use an adapter to connect a microphone with a ¼" phone plug to the XLR Lo-Z input connector on your 4000 Mixer. While, on some mixers, this adapter would "short out" the phantom power, *it is perfectly acceptable on your 4000 Mixer* (the Phantom power will not work on the channel that has the adapter plugged in, however). In other words, go ahead and use adapters as needed, even when you are using your 4000 Mixer's Phantom power feature.

Auxiliary Equipment and Phantom Power

Some types of direct-coupled auxiliary equipment could be harmed by being connected to the Lo-Z Inputs of your 4000 Mixer when the Phantom power is turned "on." While this is unlikely, it's a good idea to check with the manufacturer of the auxiliary equipment if you have doubts about this connection (or plug the auxiliary equipment into some other input, such as the Hi-Z or Auxiliary Inputs).

What is "Headroom?"

The speedometer on your car probably has a maximum of about 100 miles per hour. Of course, you never drive that fast, but, theoretically anyway, the car has enough power to be capable of that kind of speed. Why? Because you need that power for those brief acceleration periods when you pass another car.

Headroom in a piece of audio electronics is very similar. It's unlikely, for example, that you will run your 4000 Mixer at its full +24dB output for any length of time. You will seldom need that much output level. Yet, you need it occasionally for *the peaks in music and speech*. Examples of these peaks include the sharp attack of an electric guitar or of a drum stick on a wood block. These peaks may be as high as 10 to 20dB above the average level of your program which means they require 10 to 100 times as much power!

This difference between the *average* and *peak* levels in your program is known as *headroom*. Maintaining adequate headroom is important to avoid what is known as "clipping."

Clipping is a form of distortion that happens when the signal level is too high in one or more sections of the mixer. Clipping can happen in any piece of audio electronics but it occurs most frequently in the input sections of a mixer or preamplifier and in power amplifiers. The Trim control and Peak LED can help you adjust the gain of the Input Channels in your mixer to avoid Input Channel clipping.

Clipping distortion adds a very "raspy" sound quality to a signal, and, because it causes a power amplifier to produce excessive power levels, extreme clipping can actually damage your power amplifier or loudspeakers.

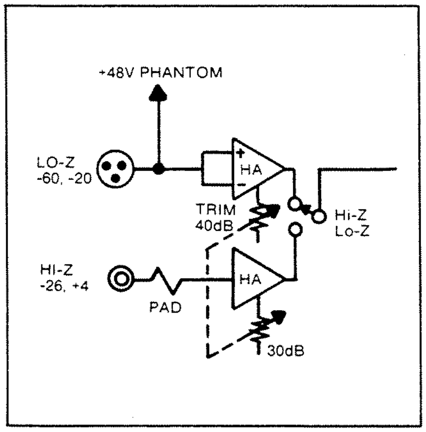
Other terms you will hear in connection with clipping distortion are "squaring up," which means the same as "clipping," and "hitting the rails" (the power supply "rails" or voltages) which means the signal is so high its voltage level is in excess of the power supply voltages. "Hitting the rails," again, is the same thing as "clipping."

In most cases, about 10dB of headroom is considered adequate to avoid noticable clipping. In your 4000 Mixer, of course, you don't have to think much about clipping or headroom. Just adjust the Trim controls properly, watch the Peak LEDs and the VU Meters and *listen*. If your ears tell you that the sound quality is good, that's a good indication that your adjustments have been made correctly.

The Hi-Z/Lo-Z Switch

Block Diagram Closeup

The Hi-Z/Lo-Z Switch appears *after* the Hi-Z and Lo-Z preamplifiers and Trim control.



Using the Hi-Z/Lo-Z Switch

Place the Hi-Z/Lo-Z Switch in the "Hi-Z" position when you are using the Hi-Z input jack. Place the Hi-Z/Lo-Z Switch in the "Lo-Z" position when you are using the Lo-Z input jack. It is not possible to use both the Hi-Z and Lo-Z inputs simultaneously but you can use both jacks and switch between them. For example, you could connect a pair of low-impedance microphones to the Lo-Z input jacks on Channels 1 and 2 and the outputs of a tape machine to the Hi-Z jacks on the same 2 Inputs. Then, during a break in a performance, you could play taped music through Input Channels 1 and 2 by simply switching the two Hi-Z/Lo-Z Switches to their "Hi-Z" positions. If you do this, set up the Trim control for the microphones and use the output volume controls on your tape machine to compensate if necessary.

The "Signal" LED, "Trim" Control and "Peak" LED

Block Diagram Closeup

The Signal LED lights when a low-level signal is present at the Lo-Z In or Hi-Z In connector. The Trim control optimizes the gain of the Input Channel for almost any input (source) level. The Peak LED lights when a high-level peak signal is present at the Lo-Z In or Hi-Z In connector.

What the Trim Control Does

The Trim control allows you to optimize preamplifier gain for different sources. By doing this, you minimize preamplifier "hiss" noise and maximize preamplifier headroom. Because the Trim control is a *continuous* adjustment (unlike the preamplifier gain *switches* on other mixers), you can optimize the gain for almost any type of input from a very low level microphone to a high line-level synthesizer.

What the Peak LED Does

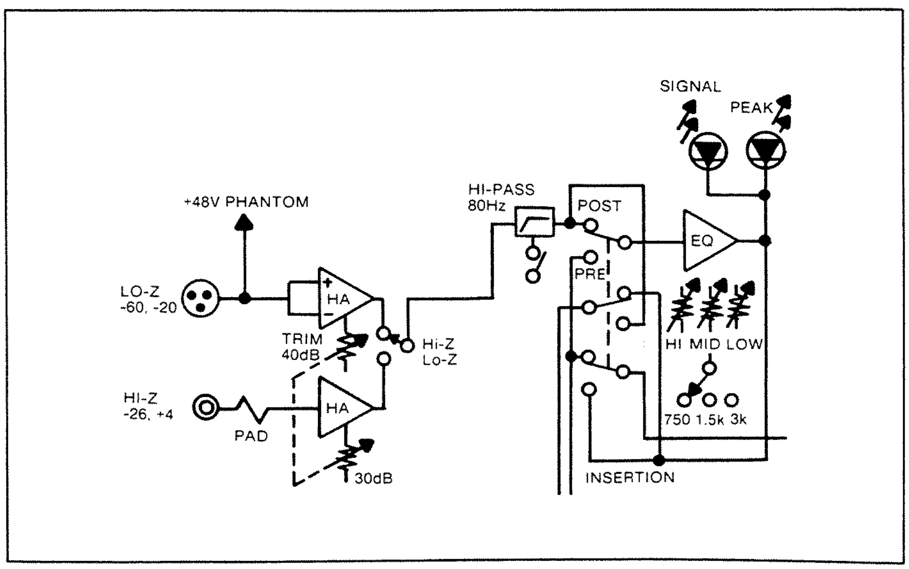
The Peak LED guides you through the process of adjusting the Trim control. After you have adjusted the Trim control, the Peak LED gives you important information on the incoming signal strength.

What the Signal LED Does

The Signal LED indicates the presence of a signal in the Input Channel. The Signal LED lights when a low-level signal is present at the Lo-Z In or the Hi-Z In connector. The Signal LED will stay on when signal levels are higher, that is, the Signal LED *is supposed to be on during normal operation of the mixer* as long as there is some signal coming into your microphone (or whatever device you have connected to the Input Channel).

An Exercise

If necessary, see "The Exercises" for instructions on setup and connections. (Keep the Program and Monitor master faders all the way down for now.) Choose any tape from your collection and start your tape machine. Now, while watching the Peak LED, turn up the Trim control (clockwise) until the Peak LED begins to blink on and off regularly. Now turn the Trim control back down, just a bit, so that the Peak LED blinks only occasionally.



That's all there is to it! You have just optimized preamplifier gain for maximum headroom and minimum noise! Now repeat the process for all the other inputs you're using. For microphones, talk or sing into the mic while you are adjusting the Trim control (or have a friend do the talking/singing). Talk or sing at approximately the same level you would expect in a performance (for a musical performance, this may be much higher than normal speaking voice).

Once you have set the Trim controls on all channels, you should not have to reset them unless you plug something different into the input or there's a drastic change in input level (like a strong-voiced singer replacing a very weak-voiced singer).

Using the Trim Control

Before an actual performance, you should perform a Trim control adjustment on each Input Channel. As you become familiar with your equipment, you should be able to judge the proper settings for the Trim controls from experience. If the same singer always uses the same microphone, for example, you'll be able to set the Trim control in the same place each time (and probably just leave it there if you always use the same Input Channel).

Watch the Signal and Peak LEDs. The Signal LED can help you determine whether or not a microphone is working (if someone is talking/singing into the microphone and the microphone is working, the Signal LED will be on). The Peak LED can help you determine the relative level of two different inputs (whichever input is loudest will light the Peak LED more often). However, if the Peak LED stays on for more than an instant, or if it lights frequently, you may be experiencing some "clipping distortion" and you should probably readjust the Trim control downward slightly.

For more information on the LEDs see the section on VU Meters.

The Input and Output Channel Faders

Block Diagram Closeup

The Input Channel "faders" are post-EQ, that is, they appear in the block diagram after the signal has passed the Input Channel equalization (tone controls). The Program and Monitor (Output Channel) "faders" are also post-EQ, that is, they appear in the block diagram after the signal has passed the Graphic Equalizers.

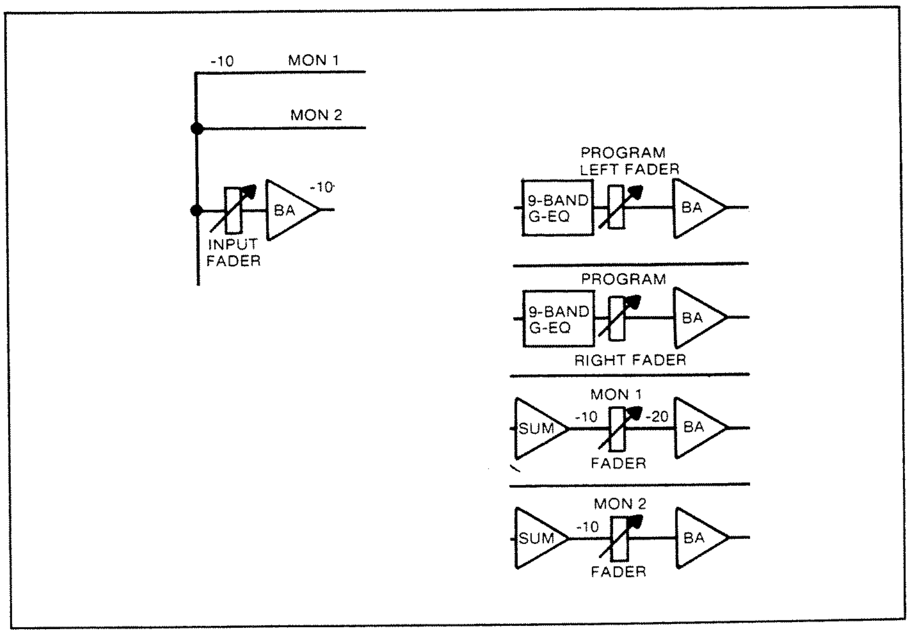
What is a "Fader?"

The term "fader" comes from the stage lighting business where a slide-type control "fades" the light level up or down. In audio, a "fader" is a slide-type control that "fades" the sound volume up or down. Faders are more desirable than rotary volume controls on a professional mixer since you can see their individual positions, *and their relative positions (the "mix")* at a glance. In addition, faders are a better "human interface" than rotary controls. It's easy to bring several faders up and down with one hand but almost impossible to do the same with several rotary controls. For this reason, rotary controls are used for functions that you can pretty much "set and forget" (like the Trim control). Faders are used for more active functions (like mixing!).

An Exercise

See "The Exercises" if you haven't already done so. Then, set the Program Left and Right (master) faders to their "0" positions. Set your power amplifier's volume controls at about the half-way position for now. If, during this exercise, you find that the volume level from your loudspeakers is either too high or too low when you have both the Input Channel Fader and the Program Faders at their "0" (nominal) positions, you may wish to change the volume control settings on your power amplifier. For more information on power amplifier settings see the next section entitled "The VU Meters."

Now, *slowly*, bring up the fader on Input Channel 1 and listen to the music from your tape machine come out of your loudspeakers! Go ahead and bring the fader up to a comfortable listening level but avoid turning the level up loud enough to cause distortion and keep the VU meter



indication at or below the "0 VU" position for now (we'll talk about the VU meters later.)

Note that, even though only one Input Channel is operating (which is connected to the left output from your tape machine), *both* of your loudspeakers are making music. This is because of the centered position of the Pan control. Leave the Pan control centered for now, we'll get to change its position in a minute or two.

Now, slowly bring up the fader on Input Channel 2 to the same position as the Channel 1 fader. Now, both the left and right outputs of your tape machine are being routed to your loudspeakers, but they are mixed "in mono." To confirm this, turn down one Program (output) fader at a time and note that the sound from both loudspeakers is, indeed, a mix of the left and right channels from your tape machine.

Next, bring the level of the Program Right fader all the way down. Now, alternately, mix the Input Channel 1 and Input Channel 2 faders up and down which will alternately bring the left and right outputs of your tape machine into the left loudspeaker. If you wish, try the same procedure with the Program Left fader all the way down. In a very real way, you are "mixing" these two "sources" (the tape machine outputs) to your loudspeaker.

Establishing a Balance

The 0 position, also called the "nominal" position should be considered the ideal setting for the Program (master) faders. Of course you will change the position of the Program faders as a normal part of mixing a performance. Yet, keeping the Program faders at or just below their 0 position has two important benefits. First, it gives you "maneuvering room" to mix the output levels up or down at will. If you keep the Program faders too near the top or bottom of their travel, you limit your ability to increase or decrease the output level. Second, keeping the Program faders near their 0 positions, along with careful setting of the Trim controls allows you to mix the Input Channel faders at positions that are reasonably near their 0 positions. *This mixing technique establishes a balance between Trim control, input fader and Program fader positions that gives you maximum maneuverability and also optimizes the signal to noise ratio inside the mixer.*

What is a "Mix?" What is a "Mix Bus?"

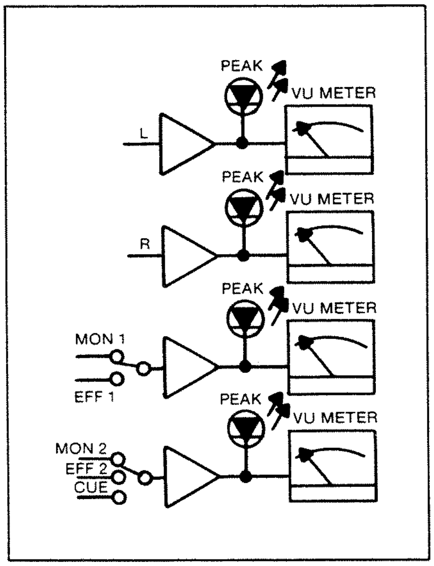
In Latin, the word "omnibus" means "all." In audio (in the English language), the word "mix" usually is used to describe the way an operator adds together *all* of the inputs and routes them to an output. For example, if we are talking about the Program Left output channel, we may refer to that as the Program Left "Mix" because all the inputs have been "mixed" into the Program Left Output (before they get to the Program Left Output, they are "summed" onto the Program Left "mix bus," see next paragraph). Similarly, when we talk about the Monitor 1 "Mix" we mean that group of inputs which, by use of the Monitor 1 control on the Input Channels, have been "mixed" into the Monitor 1 Output (via the Monitor 1 "mix bus").

A "bus" or "mix bus" is a *physical connection point* where the outputs from of a group of Input Channels (or other signals) are physically connected together (we often say the signals are "summed" on the mix bus). The mix buses in your Fender 4000 Mixer are shown as continuous vertical lines on the block diagram. If you study the block diagram for your Mixer you can see how all of the Input Channels are connected to the Program Left and Program Right mix buses, for example.

The VU Meters

Block Diagram Closeup

The VU Meters read the *average* signal level at the Program (Left or Right) Output jacks and the Monitor (1 or 2) Output jacks. The VU Meter Peak LEDs read the *peak* signal level at the Program (Left or Right) Output jacks and the Monitor (1 or 2) Output jacks.



Differences

The 4208 has two VU Meters with two VU Meter Peak LEDs. The 4208's VU Meters can be switched to read either the Program (Left and Right) Outputs or the Monitor (1 and 2) Outputs. The 4212 has no VU Meter switches but its four VU Meters (and four VU Meter Peak LEDs) allow simultaneous viewing of the Program Left and Right and the Monitor 1 and 2 Outputs. The 4216 also has four VU Meters and four VU Meter Peak LEDs. One of the 4216's VU Meters can be switched to read either the Monitor 1 or the Effect 1 output. Another of the 4216's VU Meters can be switched to read the Monitor 2 Output, the Effect 2 Output or the Cue mix which is part of the Headphone System (see "The Cue System" and "The Headphones").

An Exercise

If you haven't already done so, read "The Exercises." Then, play a tape through Input Channels 1 and 2. As you mix the Channel 1 and 2 inputs to the Program Left and Right outputs, watch the VU meters. (If you have a 4208 you may have to switch the VU meters from Monitor to Program.) The VU Meter Peak LEDs will light occasionally on normal program peaks; their operation is similar to the Input Channel Peak LEDs.

Understanding VU Meters

VU Meters have been around since the early days of audio. The mechanical and electronic design of a *true* VU Meter (like those on your Fender mixer) is such that it automatically smooths out the peaks and shows the *average* power level in the signal. That's good because it approximates the way our ears work. In other words, if the *average* level of a program increases, the program *sounds* louder. If, on the other hand, the *peak* levels in the program increase, but the *average* level stays the same, the apparent loudness (to our ears) will probably not increase. The kind of peaks we're talking about, by the way, are things like the sharp attack of a guitar pick on a string or the peaks caused by a drum stick hitting a wood block. These sounds can have a very high peak level and still not sound very loud.

The 0 or "Nominal" Position

The VU Meters on your 4000 Mixer read '0" when the signal level at the Program (Left or Right) Output jacks (or Monitor (1 or 2) Output jacks) is +4dB. *The signal level at the Program (Left or Right) Output jacks may rise to as high as ×24dB, however, before 'clipping" occurs. That is, there is a full 20dB of 'headroom" (×24dB -×4dB) at the Program (Left or Right) Output and Monitor (1 or 2) Output jacks.* This +4dB output level, along with its full 20dB of headroom, allows your 4000 Mixer to be compatible with a vast array of auxiliary audio devices designed for these levels — tape recorders, limiters, equalizers, special effects devices and so on. When you connect one of these auxiliary devices (with a +4dB nominal input level) to the Program (Left or Right) Output or the Monitor (1 or 2) Output jacks, the '0" position on your 4000 Mixer's VU Meters becomes the 'nominal" position.

Some auxiliary devices, like some of the newer low-cost multi-track tape recorders, are designed for a nominal input level that is somewhat below the +4dB output level of your 4000 Mixer. A -10dB input level is common. (+4dB) - (-10dB) means the tape machine's nominal input level is 14dB below the nominal output level of your 4000 Mixer. In this case, the "nominal" position on your 4000 Mixer's VU Meters must be 14dB lower or the "-14" position on the face of the VU Meter. *Alternately, to keep the "0" position as the "nominal" position, you may insert a 14dB pad between the Program (Left or Right) Output jack and the input of this -10dB tape machine.* (See "Pads and Transformers.") In many cases, however, a pad won't be necessary. That's because the input volume controls on many auxiliary devices function like "continuously variable input pads." If the input volume controls on your auxiliary device function in this manner, you can simply turn them down far enough to accomplish the required amount of attenuation (-14dB for the tape recorder example). If the volume controls are not labeled in dB, try the "half-way" position as a good starting point and adjust up or down if necessary. (If your auxiliary device has VU Meters, you can use them as a guide.) The goal is to allow the VU Meters on your 4000 Mixer to swing around the "0" position and yet avoid overdriving the auxiliary device.

Most professional power amplifiers are designed to produce their *full output power* when they receive an input signal of between 0dB and +8dB. The Fender 2224 and 2244, for example, produce their full rated output of 240 watts and 440 watts, respectively, when they receive an input of 0dB. Similarly, the internal power amplifiers in Fender's 3000 Series Mixers reach their full output of 200 watts into 4-ohms when they receive a +4dB signal (at the Power Amp In jacks).

On the Fender 2244, a *full output* of 440 watts for an input of 0dB means that the 0dB input level is the *maximum* (not the *nominal*) input level to the Power Amp In jacks. Remember that the +4dB output level of the Program Left or Right or the Monitor 1 or 2 Output jacks is not their *maximum* but their *nominal* output level. (The *maximum* output level from these jacks is +24dB which is 20dB above the nominal of +4dB: 20dB of "headroom.")

This may seem a bit confusing at first, but it is an audio industry standard to publish the *maximum* input level to a power amplifier and the *nominal* output level of a preamplifier (or other low-level or line-level device)! That is, when you read that the output level from a professional graphic equalizer is +4dB that means that the *nominal* output level from this equalizer is +4dB. Its *maximum* output level may be anywhere from 10dB to 20dB higher than +4dB. Yet, when you read that the input level to a professional power amplifier is +4dB, that is the *maximum* input level (which produces full output from the power amplifier). The *nominal* input to that power amplifier will be from 10dB to 20dB *lower* than the +4dB rating.

The result is that preamplifier and line-level devices, like your 4000 Mixer, have from 10dB to 20dB more output level than is needed to drive a professional power amplifier to full output! The reason for this difference is that you will commonly insert several auxiliary devices between the output of your Mixer and the input of a power amplifier. *Some of these auxiliary devices may cause a loss in level amounting to between 10dB and 20dB.* This loss can be overcome, however, by the additional output level available from your 4000 Mixer.

Your 4000 Mixer's Program (Left or Right) and Monitor (1 or 2) Output jacks have a maximum output of

+24dB for exactly this reason. You can insert auxiliary devices having as much as 20dB of loss (a passive graphic equalizer, for example) and still have enough output level to drive almost any power amplifier to its full output.

When you are *not* using any auxiliary devices, however, the Program (Left or Right) and Monitor (1 or 2) Output jacks are capable of as much as 20dB more output level than is needed to drive the input of most professional power amplifiers. This means that the "nominal" VU Meter position must be lowered by the amount of "gain overlap" (the difference between the +24dB output of your 4000 Mixer and the rated input level for the power amplifier). For example, for the Fender 2244, with its rated "0dB" input, the "nominal" position on the 4000's VU Meters would be the -24 position. *Alternately, to keep the "0" position as the "nominal" position on the VU Meters, you can insert a pad (again, equal to the 24dB "gain overlap") between the Program (Left or Right) or Monitor (1 or 2) Output jacks and the power amplifier's input jacks.* (See "Pads and Transformers.") *In most cases, however, a pad won't be necessary. That's because the input volume controls on most professional power amplifiers function like "continuously variable input pads." If the input volume controls on your power amplifier function in this manner, you can simply turn them down far enough to accomplish the required amount of attenuation.* If the volume controls are not labeled in dB, try the "half-way" position as a good starting point and adjust up or down if necessary. (If your power amplifier has VU Meters, you can use them as a guide.) *Your goal is to allow the VU Meters on your 4000 Mixer to swing around the "0" position and yet avoid overdriving the power amplifier.*

Using the VU Meters

In general, it's a good idea to keep the VU Meter swinging around the "nominal" position or below. (The "nominal" position may change, of course, as explained in the previous section.) Occasional swings above nominal are acceptable, but frequent swings above nominal probably mean that you are overdriving any auxiliary equipment connected to the Program (Left or Right) or Monitor (1 or 2) Output jacks (or you may be overdriving your power amplifiers). When this happens, you may experience the kind of distortion known as "clipping," which you will hear as a very raspy, irritating sound quality.

The VU Meters give you an idea of the *loudness* (the average power level) of the signal. While the VU Meter shows *average* power level, the VU Meter Peak LED indicates the presence of high-level (but normal) program peaks. In addition, unlike the purposely slow response of the VU Meters, the Input Channel and VU Meter Peak LEDs respond *very* fast. This means that they may turn on occasionally even when the VU Meter is at or below its "nominal" position. As long as you don't *hear* any distortion, it's probably okay for the Input Channel or VU Meter Peak LEDs to light frequently. If they stay on longer than an instant, however, or are "on" more than they are "off," you are probably experiencing some clipping distortion.

Severe clipping distortion is almost always audible and, provided that it is taking place in your 4000 Mixer (it could be taking place in your power amplifier and not in your Mixer), severe clipping distortion will light the Peak LEDs (and they will probably stay "on" for as much as several seconds). This kind of distortion is one of the most common causes of loudspeaker damage. In fact, the clipping distortion produced by *overdriving a small power amplifier* can actually be more dangerous to a loudspeaker than a higher level of unclipped power from a larger power amplifier! Ask your Fender dealer for help in choosing a power amplifier that's the right size for your loudspeakers and the type of performances and rooms you will work with.

Using the LEDs and VU Meters as Artistic Mixing Tools

The indicators on your Fender 4000 Mixer can do a lot more than just tell you whether or not a signal is present and how loud it is (or whether it's *too* loud).

For example, you're mixing an unfamiliar singing group of all female voices. One voice stands out and needs to be lowered in level, but *which one is it?* With a little practice, you can tell from how often the Peak LED is lighting. The Input Channel Peak LED will light more often, and stay on longer, on the channel with the loudest signal (the stand-out voice).

As another example, imagine that the audience is arriving and you still haven't managed to get a full sound check (no matter how well you plan. . .). You know the performance but *are all the mics working?* You don't want to disturb the audience by having "Check, One, Two!" coming through the loudspeakers so you simply turn down all your faders (just turning down the Program and Monitor faders will do) and have a helper talk into the microphones one by one *while you watch the Signal LEDs* on each channel! If all the Signal LEDs come on at the appropriate mic check, you are reasonably assured that these Input Channels will work properly when your performance starts. There are, of course, other things that could cause the Signal LEDs to light — like bad cables, for example, but we'll cover that in the section on Troubleshooting, near the end of this Manual.

If you have a 4216, of course, you could easily solve both of the above problems by simply using the Cue and Headphones system! For more information on these functions, see the section entitled "The Cue and Headphones Systems."

As a final example, assume that you've been couped up in an enclosed room (perhaps with the lighting crew) and told to mix a performance. This, of course, is an extremely undesirable situation because you can't hear what the audience hears and your ability to do your job well has been dramatically impaired. How do you mix the performance? By listening to the system (using a tape) before the performance to get an idea of how loud it will be for a given VU Meter reading. Then, during the performance, you can watch the VU Meters and have a reasonable idea of the actual loudness in the audience. Even watching the VU Meters closely, however, and even if you have a good set of control room monitor loudspeakers, it's still a good idea to move out into the audience area as often as possible to get a "live" viewpoint of the sound.

Information Galore! A Review of the LEDs and VU Meters

In addition to its VU Meters, your Fender 4000 Mixer has three different types of LED indicators! As you learn to read them and understand their meanings, you will realize that they tell you a lot about the signals making their way through your mixer.

In brief review, then, the "Signal" LED (Input Channel) indicates the *presence* of a signal in the Input Channel. The Signal LED comes on even at very low input levels and will stay on, almost continuously, when signal levels in that Input Channel are normal to high. The Input Channel "Peak" LED indicates the presence of a high-level input signal. The VU Meter Peak LED functions much like the Input Channel Peak LED except that the VU Meter Peak LED indicates the presence of high-level peak signals in one of the Program or Monitor (output) channels. It is normal (and even desirable) for either of the Peak LEDs to light occasionally, even frequently (the Peak LEDs should not stay on continuously).

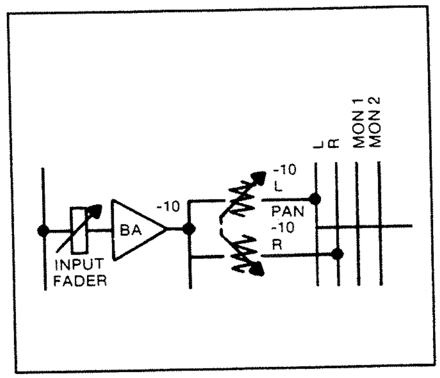
The VU Meter shows a continuously varying display of output signal level in a way that approximates the way we hear. Thus, the VU readings are a good indication of the apparent *loudness* of the signal level.

Again: "Signal" means low level signal (it will stay on when signal level gets higher); "Peak" means high-level peak signals (Peak can light frequently but not continuously). The VU Meters show *average* signal level which corresponds to apparent *loudness*. Keep the VU Meters swinging at or below the "nominal" position (it's okay to allow them to swing above nominal on occasion).

The (Input Channel) "Pan" Control

Block Diagram Closeup

"Pan" controls appear in each Input Channel and in the Aux In and Effect Return sections. In each case, the Pan control "pans" a signal between the Program Left and Right buses.



What is a "Pan" Control?

The term "pan" is adapted from "panoramic," as the term is used in the movie business. To "pan" a camera, while filming a movie, means to swing it from one side of a scene to the other to show the audience a "panoramic" view. In audio, to "pan" an audio signal means to vary the volume of the signal from one loudspeaker to another which makes the apparent source of the audio move from one loudspeaker to the other. A "Pan" control performs this function.

Using the Pan Control

Besides its ability to move the apparent source of a sound from one place to another, the Pan control may also be used to "position" an instrument (more or less permanently) at some point between two loudspeakers. This "widens" the apparent size of the sound source, at least for people sitting in an area where they can hear both loudspeakers. The Pan control can also be used to send entirely different mixes to the Program Left and Program Right outputs. You might do this if your two loudspeakers were pointed at entirely different areas in a room (an "L-shaped" club, for example).

If, *as in most situations,* at least some part of the audience cannot hear both loudspeakers well, it is a good idea to avoid panning an Input Channel entirely to one side or the other. That would cause it to disappear from one loudspeaker and part of your audience would then not hear that input. This same situation, where at least part of your audience cannot hear one of the loudspeakers well, often prevents you from doing a true stereo mix (a mix where the apparent placement of instruments in the mix corresponds to their physical placement on stage).

The sound system operator is, in a sense, the representative of the subjective tastes of *each member* of the audience. This is one reason experienced operators always try to position themselves (and their mixer) in an "average" seat. Typically, this "average" seat will be about 1/3 of the way back from the performance area and slightly off center (off-center to avoid the frequent bass-frequency buildup near the center of an audience area). Positioned in this average seat, you can *hear* the results of changing the Pan control setting, for example, and you can be certain that that action has enhanced, not degraded, the mix.

An Exercise

If you haven't already done so, read "The Exercises." Then, while playing your tape machine, fade the Channel 2 input all the way down. Set the Program Left and Right faders at equal levels. Now, turn the Channel 1 Pan control all the way counterclockwise toward the "L" (left) side. Notice that the sound is now coming entirely from your left loudspeaker and that the Left VU Meter is the only one showing any activity. Try turning the Pan control all the way clockwise to the "R" (right) position. The sound is now coming entirely from the right loudspeaker and the Right VU Meter is doing its thing while the Left VU is quiet.

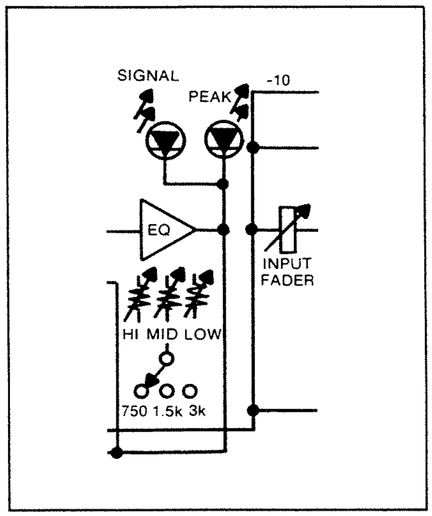
"Pan" back and forth for a while to get the "feel" of the control. Then set the Channel 1 Pan all the way left and set the Channel 2 Pan control all the way right. Now bring up the Channel 2 fader. Presto! Stereo! Channel 1 (the left channel of your tape machine) is feeding *only* the Program Left output (your left loudspeaker). Channel 2 (the right channel of your tape machine) is feeding *only* the Program Right output (your right loudspeaker).

Try a "cross-fade" by simultaneously turning the Channel 1 Pan control all the way right and the Channel 2 Pan control all the way left. The sound mixes to mono and centers, then splits again with the image reversed from its original position. This could turn out to be fun!

The Input Channel Equalization Controls

Block Diagram Closeup

The Input Channel Equalization (Tone) controls appear in the block diagram before the Input Channel fader. Thus the Equalization controls are "pre-fader." The Equalization controls are also pre-Monitor and pre-Effect. What that means is that any settings of the Equalization controls affect not only the Program mixes but also the Monitor and Effects mixes.



What the Input Channel Equalization Controls Do

The Equalization controls have much the same effect on the frequency response of the Input Channel as the tone controls on your stereo system have on its frequency response. The purpose of these controls is to give you the ability to alter the frequency response of a voice or instrument to improve its subjective sound quality. Here, in fact, is *the* place you really show your skill as an audio artist!

The straight lines in Chart A show the frequency range of an electric bass guitar and a flute. Don't confuse this

Chart A

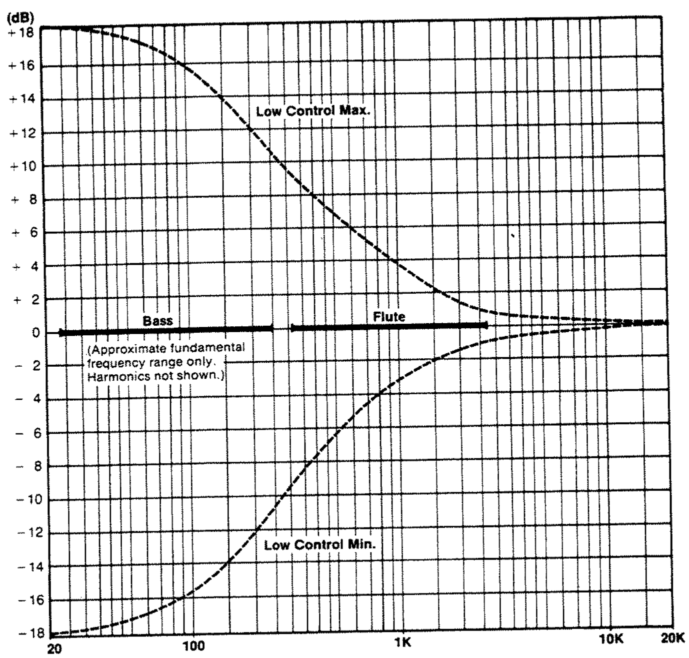


Chart B

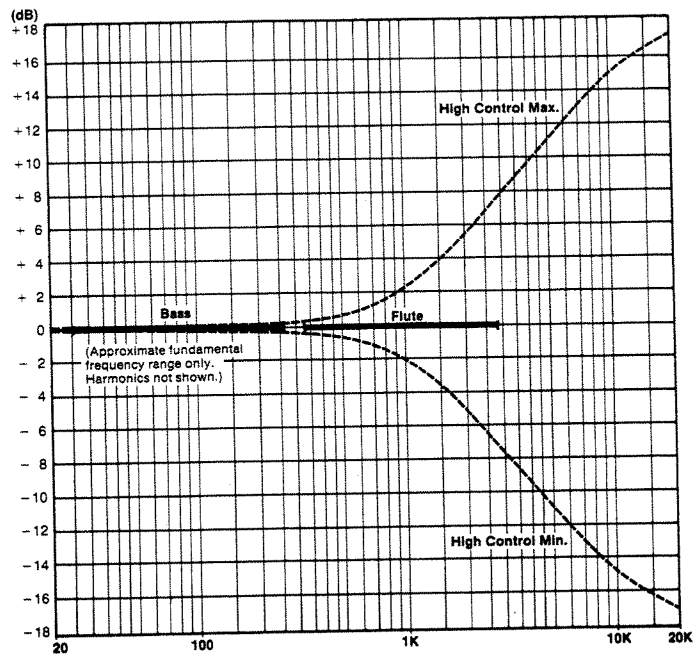


Chart C

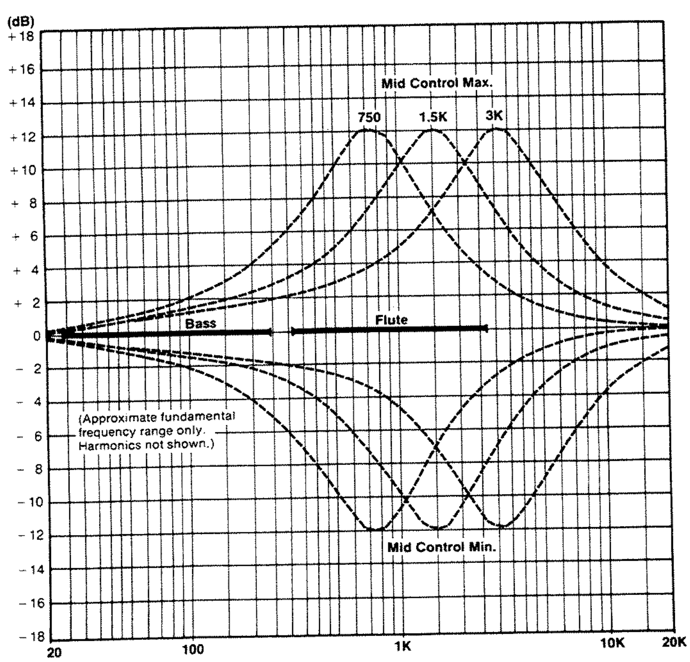
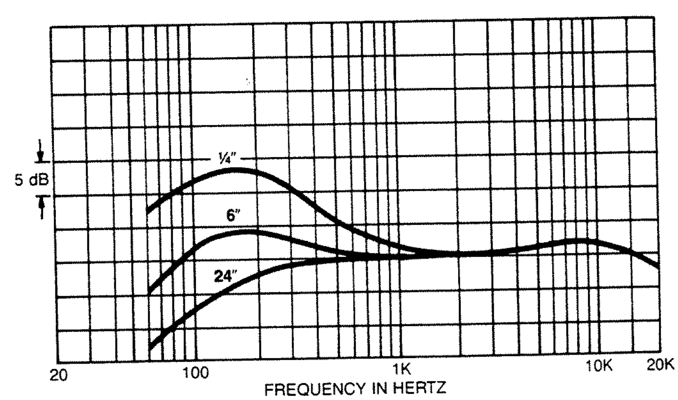


Chart D. Proximity Effect at various working distances — cardioid dynamic microphone.



term "frequency range," which shows the highest and lowest notes that can be played on the instrument, with the similar term "frequency response," which normally applies only to electronic devices and not to musical instruments. Also keep in mind that the range of upper harmonics will vary from instrument to instrument (thus, these charts are not exact).

The dashed line shows the changes in Input Channel frequency response produced by the "Low" control (bass tone control). You can see from these lines that the Low control can have a significant effect on the sound of an electric bass but will have much less effect on the sound of a flute.

The dashed line in Chart B shows the changes in Input Channel frequency response produced by the "High" control alongside the same electric bass and flute frequency ranges. Here, it's apparent that the High control can have considerably more effect on the sound of the flute than the electric bass.

The dashed lines in Chart C show the changes in Input Channel frequency response produced by the "Mid" control. There are three dashed lines corresponding to the three settings of the Mid switch. The solid lines show the frequency range of a typical male and female voice as well as the electric bass and flute shown in Charts A and B.

Here, it's apparent that the Mid control can cause significant changes in the sound quality of all these instruments. In other words, of the three controls, the Mid is the most powerful and probably the most important. Its three switch settings also make it the most versatile of the three controls. Compare this Mid control and the important part it plays to the *complete lack of a Mid control on many competitive mixers!*

Finally, the solid lines in Chart D show the changes in frequency response of a Fender D-1 cardioid microphone for different microphone-to-voice distances. The increase in bass response of a cardioid microphone at short microphone-to-voice distances is known as "proximity effect" and is discussed in more detail in the section on Choosing and Using Microphones. In some cases, you might want to use the Low control to reduce this proximity effect (in other cases, you would welcome the increased "warmth" the proximity effect can add to a voice).

Also note the slight rise in the high-frequency response of this typical cardioid microphone. This is a planned "feature" of the microphone, known as "presence," and it can help bring out the sibilants (consonant sounds) in a voice. In some cases, this presence is desirable, in other situations, you may wish to reduce the "presence" of a microphone with the High control.

An Exercise

If you haven't already done so, read "The Exercises." Center the Pan controls again and bring the Channel 2 fader all the way down. Select a series of tapes with lots of different instruments, again, use solo instruments whenever possible. Try a low-frequency instrument first, like an electric bass or organ.

Set the Program Left and Right faders for a comfortable volume level from your loudspeakers. Now, while playing the tape, experiment with the Channel 1 Low control. Can you make the instrument sound more or less "mellow?" Notice that the Low control, because it affects most of the frequency range of the instrument, almost acts like a volume control for this instrument.

Keep the same tape running and reset the Low control back to its center position. Now, try out the Mid control with the Mid switch set at "750" (which means that the peak of its effect is at 750 Hz). What effects does the Mid control have on the "sharpness" of the sound quality? Try again with the Mid switch set at 1.5k and 3k (1500 Hz and 3000 Hz). As you move the Mid switch up in frequency, the Mid control will have a different *and decreasing* effect on the bass instrument.

Now, reset the Mid control to its center position, and, with the same tape still running, try out the High control. You probably won't notice much effect on the sound of the bass instrument. You may notice an increase in tape hiss! (Tape hiss is primarily a high frequency phenomenon.) You may also notice an increase in "presence" in some bass instruments which have an appreciable high-frequency content (lots of high harmonics), an electronic organ, for example.

You may wish to try these same experiments with a high-frequency instrument like a flute and with several midrange instruments like saxophone, violin and piano. You can also do some pretty amazing things to the sound of a good drummer with the Input Channel Equalization controls. In other words, the best way to understand the operation of the Input Channel Equalization controls is to find tapes of as many different instruments as possible and experiment!

The Input Channel Equalization Controls and the Human Voice

One "instrument" you should work with in this exercise is the human voice. If for some reason, you didn't get a tape of a friend's voice, find a tape from your collection with a solo singing voice. You want one with as little reverberation and effects as possible. Ditto instrumental backup — an acappella voice would be ideal. If you have a choice of male and female, choose one and then do this exercise over again with the other.

If there are two of you doing this practice session, now is the perfect time to try out your microphones. Do these exercises with a microphone instead of your tape machine. And do something more original than "testing, one, two." If you don't want to sing (the best way to try out this set of controls) then at least read something from a book or magazine to get some variety into these tests!

Try to make the voice sound "warmer" (or less "warm") by using the Low control. Notice the effects the Mid control can have on the relative "harshness" of the voice. Try all three Mid switch settings. Depending on whether the voice is male or female and on the particular voice qualities, you will probably find that one of the Mid switch settings is optimum for controlling *this particular voice*. Some other voice, of course, might respond better to a different Mid switch setting. Now, try the High control and notice its effect on the "presence" of the voice. You can emphasize or de-emphasize the sibilants (high-frequency consonant sounds) in the voice with the High control. You can also affect the sibilants, to a lesser degree, with the Mid control when the Mid switch is set to the "3k" position.

If at all possible try out the Input Channel Equalization controls with a *live* voice (other than your own) and your various microphones. Not only will you discover the difference between live and recorded signals, you will hear the differences among the various microphones. Listen, in particular, for "proximity effect," an increasing bass-boost noticable in many cardioid microphones as the talker moves closer to the microphone. How would you counter this effect if it was excessive? Also listen for the difference in "presence" (high-frequency response) in the microphones. How would you increase presence if it were lacking in a microphone (or voice)?

The Input Channel Equalization Controls and Mixed Instruments

Now that you have a good idea of the effects of the Input Channel Equalization controls on *solo* instruments, it's time to try out your skills on an *ensemble*! Find a tape with a group of instruments, and at least one voice. An "uncluttered" piece of music like a folk song would be ideal. Avoid a piece with lots of reverb and complex effects. Run both channels of your tape machine and use both Channel 1 and Channel 2 on your Fender Mixer. Bring the faders up to a comfortable listening level.

This is a simulation of a real performance. The only differences are that you don't have individual control over the various inputs and that there will always be some differences in the sound of live versus recorded sources. Never-the-less, you can try out some "live-performance" techniques here. In particular, you should attempt to isolate particular instruments and increase their apparent level. Since you don't have individual volume controls for each instrument, on the tape, you are limited to using the Input Channel Equalization controls (and, perhaps the Pan controls if the instrument is primarily on one channel of your tape machine). For example, you should be able to bring out the voice(s) with the Mid control (try the various Mid switch positions for the best results). You might even be able to emphasize the voices on Channel 1 and the bass instruments on Channel 2 and then use the Pan controls to completely alter the original mix! The importance of this exercise is that, in a live performance, you can use these techniques to emphasize a particular voice or instrument *without increasing the volume level*. There's a lot more to "mixing" than just fader settings!

In a live performance, you may find that the Input Channel Equalization control settings you used for an individual instrument during a practice session just don't sound the same when there are other instruments (or voices) present. This is a normal effect of a live mix. The point is that there is no *right or wrong* way to set the Input Channel Equalization controls for a particular instrument or voice. What is important is the subjective sound quality you achieve during an actual live mix. *Think mix!* If the piano sounds like a piano (and the other instruments and voices also sound "correct") during the actual performance, then you've done your job right.

A Precaution About Using the Input Channel Equalization

Equalization is a very powerful tool. Used carefully, it can significantly enhance your artistic capabilities. Used to excess, it can actually hinder the process of sound reinforcement. The trick is to use the Input Channel Equalization controls in a *subtle* way, like an artist uses a fine-line paint brush. Over-use these controls, by turning them too far up or down, and you risk excessive noise, distortion and a very un-natural sound quality.

We'll discuss equalization in more detail in another section. For now, don't hesitate to use the Input Channel Equalization controls, but remember that it is very rare to need more than 3dB to 6dB of boost or cut for the vast majority of voices or instruments.

The High-Pass Switch (Model 4216 Only)

Block Diagram Closeup

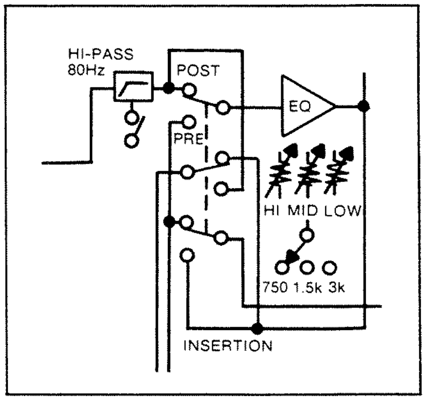
The High-Pass circuit is "post" both the Hi-Z and Lo-Z preamplifiers (and thus affects both inputs). It is also "pre-Fader," "pre-EQ" and "post-Trim."

What the High-Pass Switch Does

The High-Pass Switch inserts an 80-Hz, 12dB/octave high-pass filter into the Input Channel. A "high-pass" filter *passes* frequencies above its "cutoff" frequency (80 Hz in the 4216) and *attenuates* frequencies below its cutoff frequency. 12dB/octave is the "rate of attenuation" which means that frequencies are attenuated 12dB for every octave they are below the cutoff frequency. Thus, 40 Hz will be attenuated by 12dB and 20 Hz will be attenuated by 24dB when the High-Pass Switch is in the "In" position.

When to Use the High-Pass Switch

Using the High-Pass can help protect your system (especially your loudspeakers) from possible damage due to high-power, low-frequency transients caused by dropped microphones or faulty cables. Because it reduces these low-frequency transients, the High-Pass will also help you avoid clipping distortion and can reduce intermodulation distortion in loudspeakers. Finally, by reducing unwanted low-frequencies, the High-Pass can "clean up" the sound quality of individual Input Channels and your entire system. For example, the High-Pass will attenuate "breath noises" (the pops caused by the "P's" and "B's" in our language). As another example, the High-Pass will also help attenuate "turntable rumble" noises when you have a phonograph turntable (and RIAA preamplifier) connected to your Mixer. For all these reasons, it's a good idea to get in the habit of leaving the High-Pass Switch in the "In" position unless you have a specific reason for switching it "Out."



The Effects of the High-Pass on Music

Except for low-bass instruments like bass guitar, low organ notes and bass drum, most of the frequency content of music is above 80 Hz. Thus, switching the High-Pass circuit "in" will not appreciably affect the sound quality of most instruments. In most cases, even a deep male voice will not be noticably affected by switching the High-Pass "in."

An Exercise

If you haven't already done so, read "The Exercises." Then, connect one of your microphones to Input Channel 3 (choose a cardioid dynamic type like the Fender D-1 if possible). Keep the Input Channel EQ controls at their "center" positions for now, and switch the High-Pass "Out." While talking into the microphone, bring up the Input Channel Fader and the Program Right Fader until you reach a comfortable listening level. (Keep the levels well below the "feedback" point.)

Now, *carefully*, tap the microphone with the palm of your hand. Listen to the "thump" from your loudspeakers and watch the "Peak" LEDs (which will probably light). Then switch the High-Pass "In" and repeat the test. Notice how effectively the High-Pass filter attenuates the "thumping" noise.

Try the test again but this time talk into the microphone. Say the word "pie" over and over, speaking directly into the microphone so that you can hear the effects of the "P" sound. Then switch the High-Pass "in" and try again. The "P" sounds should be noticably reduced.

Now, reduce the level on Channel 3 and play a tape with lots of bass instruments through Channels 1 and 2. Switch the High-Pass "In" and "Out" on both channels and listen to the difference in the musical quality. Since most cassette machines have poor low-frequency response, you probably won't hear much difference. Even the bass instruments will come out clearly, with little or no change in sound quality. You may even find that the sound quality is *improved* by switching the High-Pass filter "In" since this will reduce "wow and flutter" noises from your tape machine.

Using the High-Pass in a Live Performance

Live music will have considerably more bass energy than your tape machine. Thus, you should experiment with the High-Pass Switch to see if it degrades the subjective sound quality of any given instrument. In most cases, it won't. Thus, again, it's a good idea to get in the habit of leaving the High-Pass Switch in the "In" position unless you have a specific reason for switching it "Out."

One way to use the High-Pass filter on a bass instrument is to switch High-Pass "In" and also boost the "Low" control slightly. This will give a "false bass" quality to the sound that, in many cases, will actually sound "cleaner" and "tighter" than the unmodified signal. Try this on a bass drum microphone, for example. Your loudspeakers will be happier, too, (and they will produce less distortion) since, unless you are using separate subwoofers, most loudspeaker systems have a hard time accurately reproducing frequencies much below 80 Hz.

The Program Graphic Equalizers

Block Diagram Closeup

The Program Graphic Equalizers appear in the block diagram before the Program faders. That is, they are "pre-fader." In the Program Output Channels, the Insertion Send jacks are "normaled" (normally connected) to the Insertion Return jacks. Connecting an external device to the Insertion Return jack automatically disconnects this normaled connection.

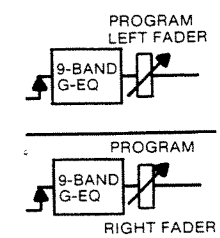
An Introduction to the Program Graphic Equalizers

The Program Graphic Equalizers perform tone-control-like functions for the Program Outputs in a similar manner to the way the Low, Mid and High controls work on the Input Channels. There are, however, two, important differences.

First, the Program Graphic Equalizers are 'graphic" equalizers (graphic equalizers are also discussed in the section entitled "Equalization"). This means that there are more controls and each control affects a narrower frequency range. It's like having three Low, three Mid and three High controls.

Second, the Program Graphic Equalizers affect *an entire mix*. That is, the Program Left Equalizer affects the frequency response (and therefore the sound quality) of every input that is affected by the Program Left fader.

Because they operate on individual Input Channels, you will normally use the Input Channel Equalization controls to affect the tonal character of an *individual source* (a microphone or instrument pickup



connected to one of the Input Channels). Because they operate on an entire mix (an entire Program output channel), you will normally use the Program Graphic Equalizers to affect the tonal character (the frequency response) of one channel of your *entire system*. That is, the Program Graphic Equalizers are used to "EQ the system."

An Exercise

If you haven't already done so, read "The Exercises." Then play a tape of a group of instruments and/or voices. Experiment with the sound quality changes you get when you boost or cut the various sections of the Program Left and Right Graphic Equalizers. Notice that each control causes similar effects to those you experienced when using the Input Channel Low, Mid and High controls. The Program Graphic Equalizer controls, however, affect a narrower frequency band, and you can hear that by first trying the Input Channel Low control, (then set it back to its center position) and then trying the Program Graphic Equalizer "125" control (which means 125 Hz). Experiment with the other controls, comparing them to the Input Channel controls if you wish.

Using the Graphic Equalizers

Because they affect an entire mix, you wouldn't use the Program Equalizers to try to enhance the "presence" of a lead singer's voice because your actions would affect everything else in the Program mixes! You might, however, use the Program Equalizers to enhance the presence of the entire sound system in a "dull" sounding room full of carpeting, draperies and over-stuffed furniture (some hotel lounges are like this).

In other words, use the Input Channel Low, Mid and High Equalization controls when you need to affect an individual instrument or voice. Use the Program Graphic Equalizers when you need to alter the sound quality of an entire mix (or your entire sound system).

Like the Input Channel Equalization controls, the Program Graphic Equalizers are powerful tools. And, like the Input Channel controls, the Program Equalizers can enhance a performance or detract from it. Remember that, in almost every case, 3dB to 6dB of boost or cut on any individual control should be sufficient. Don't hesitate to use the Program Equalizers, as much as you need them (that's why we put them there!). But think of them as artist's tools and use them with an artist's touch!

The Monitors

Block Diagram Closeup
The Input Channel Monitor controls are pre-fader and post-EQ. The Output Channel Monitor faders are completely separate from the Program Output functions. The Input Channel Monitor controls may be changed to post-fader. This modification *must* be performed by a qualified service technician.

What the Monitor Mixes Do
The Monitor 1 and 2 mixes give you the ability to send a group of inputs to a separate amplifier and loudspeakers to *monitor* the progress of a performance and the effects your mixing actions have on that performance. Alternately, you may set up a Monitor system which allows a performer to monitor their performance and to hear other performers better.

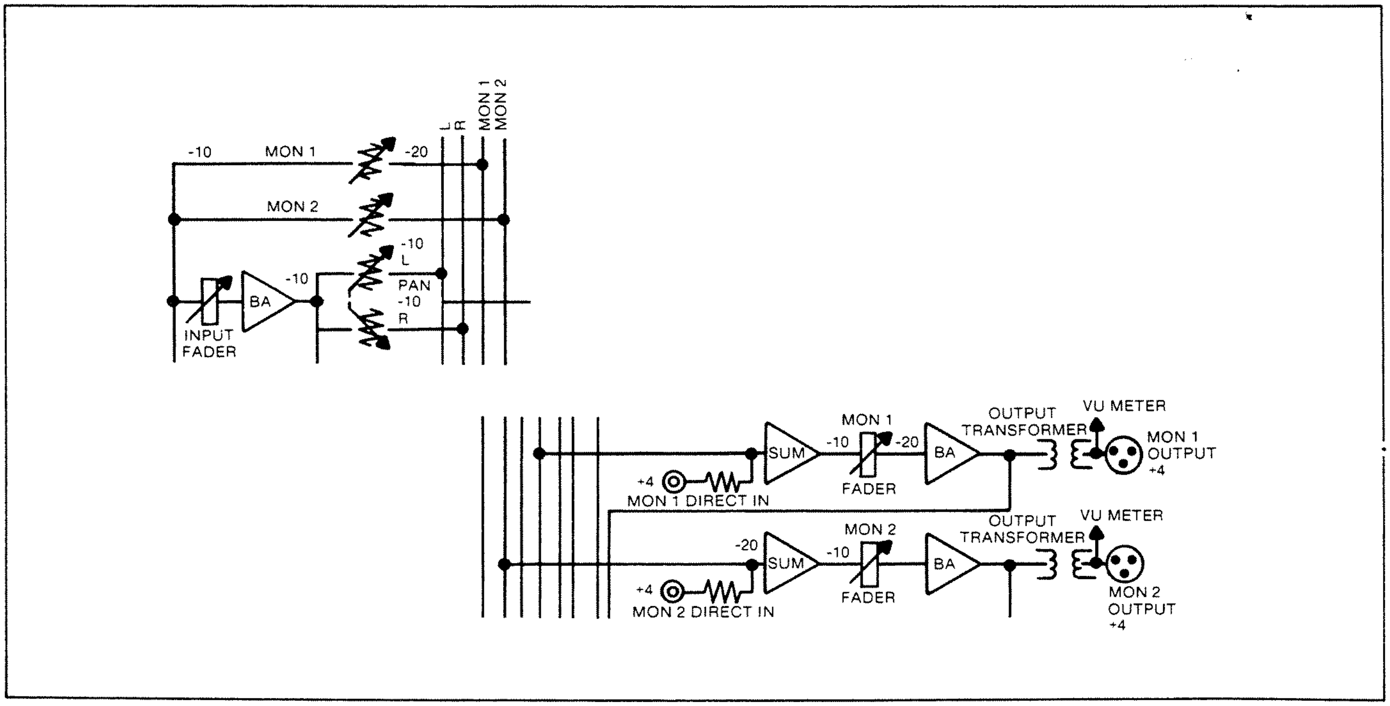
An Exercise
If you haven't already done so, read "The Exercises." You'll need additional connections to try out the Monitor mixes (do it; it's worth while to find out how these important mixes work). You can either use a separate power amplifier or you can reconnect the power amplifier you have been using for the previous Exercises (which have, so far, delt with the Program Outputs). Connect the power amplifier's inputs to the Monitor 1 and Monitor 2 Out jacks and connect its outputs to your loudspeakers.

Turn the Monitor 1 and Monitor 2 output faders all the way down and turn on your tape machine. The Signal LED should light and the Peak LED may light occasionally, as usual. Bring up the Monitor 1 control on Channel 1 to its center position. Now bring up the Monitor 1 output fader until you reach a comfortable listening level. Readjust the two controls if you wish to bring them both into their "nominal" positions (about midpoint for the Monitor 1 Input Channel control).

Now, move the Channel 1 fader up and down. Notice that the Monitor level does not change! In other words, the Monitor 1 function is independent of the position of the Input Channel fader. We say that Monitor 1 is "pre-fader" which means that the connection to the Monitor 1 control comes at a point in the block diagram *before* the signal has passed through the Input Channel fader. The Input Channel Monitor 1 and Monitor 2 controls can be changed to post-fader. This modification *must* be performed by a qualified service technician.

Turn down the Monitor 1 control and try the same experiment with the Monitor 2 control and the Monitor 2 output fader. Is Monitor 2 also pre-fader? (Yes, it is.)

Try altering the tonal quality of the Monitor 1 or Monitor 2 mixes with the



Input Channel equalization controls. These controls *do* affect the Monitor 1 and 2 mixes! We, therefore, say that the Monitor mixes are "post-EQ" which means that the connection to the Monitor 1 and 2 controls comes at a point in the block diagram *after* the signal has passed through the Input Channel equalization controls.

You may also wish to experiment with the Trim and Pan controls. The Pan control affects only the Program mixes and does not affect the Monitor mixes. The Trim control affects the Monitor mixes in the same way it affects the Program mixes. That means you can set up the Trim control just once and it will be right for both the Monitor and Program mixes.

Be cautious about altering the setting of the Trim control during a performance, however. You (the system operator) will normally be unable to hear the on-stage monitor loudspeakers. Yet, a change in a Trim control setting will affect your monitor mix. In other words, a change in the Trim control could significantly alter your on-stage Monitor mix or even increase the possibility of feedback and yet, in your position as the sound system operator, you would not be able to *hear* these changes. (You can, of course, use the Phones system to listen to what's going on in the Monitor mix. See "The Que and Headphones System.")

Using the Monitor Mixes

The primary purpose of the Monitors is to give you the ability to send a *separate* mix, independent of the main Program mixes, to a set of monitor amplifiers and loudspeakers. Those monitor loudspeakers may be on stage to give the performers the ability to hear themselves and the other performers better. The monitor loudspeakers may also be in your "control room" if you are not located in a place where you can hear the performance directly. The idea, in any case, is that the Program mix may not be ideal for these monitor loudspeakers.

On stage, for example, a drummer may want a "mix" that emphasizes the vocals and de-emphasizes the drums. A vocalist may want a "mix" that de-emphasizes the instruments but emphasizes the vocals and has just enough drums to help the vocalist keep on time. In a control room, your monitor mix will bring up electric instruments more than you would bring them up in the audience area (since the electric instruments will carry much of their own sound to the audience but this sound will not reach you in the control room).

In some cases, your 4000 Mixer may be used *exclusively* for monitors. That is, the Program Left output will feed one set of monitors, the Program Right will feed another set and the Monitor 1 and 2 outputs will feed a third and fourth set of monitors! This kind of diversity in monitor mixing is especially common in large entertainment systems and it illustrated in one of the Example Systems later in this manual.

Actually, the Monitor 1 and 2 mixes are unique mixes which can be used for just about any purpose. In a night-club system, for example, you might set up a separate mix, using Monitor 1, to feed a set of amplifiers and loudspeakers in a separate room of the club. By using a separate mix you could balance the system properly for both the main room and the separate room.

Because the Monitor 1 and 2 mixes are pre-fader, they are truly independent of the Program mixes. The Effect mix (discussed next) could be used as a post-fader monitor mix if you wanted to be able to set the monitors just once and then have the monitors automatically "mixed" along with the Program mixes. Another way to get a post-fader monitor mix would be to have a qualified service technician perform a post-fader modification on the Input Channel Monitor 1 and 2 controls (the modification affects both Monitor controls and *must* be performed by a qualified service technician.)

One minor precaution about the Monitor 1 and 2 mixes. At the end of a performance, or during a break, you will probably bring down the Program Left and Right faders to keep stage noises from reaching the audience. Remember to bring down the Monitor 1 and 2 (master) faders, too! Since the Monitor mixes are entirely independent of the Program mixes, they will keep operating normally, even when the Input Channel faders and the Program Left and Right (master) faders are all the way down! This means, for example, that the audience near enough to the stage to hear the stage monitors could still hear stage noises or even feedback if a technician moves a microphone to the wrong position. If you are using wireless microphones and a performer walking off stage forgets to turn off their transmitter, their off-stage conversations will continue to come through the monitors, too! Remember that the primary reason you have to think about these potential problems because you, as sound system operator, are normally unable to *hear* the on-stage monitor loudspeakers.

One way, of course, to avoid having to remember to turn down the Monitor mixes is to have a qualified service technician perform the Monitor "post-fader" modification to your 4000 Mixer. That way, when you turn down the Input Channel faders, the Input Channel Monitor controls (because they are now "post-fader") are also effectively turned down. This modification is probably undesirable for experienced operators who like the idea of a totally independent Monitor mix. It could be desirable, however, if your 4000 Mixer will be operated by inexperienced personnel or if it will be used as the "house" mixer in a nightclub, for example, where a different operator may be present for each new performing act.

The Effects

Block Diagram Closeup

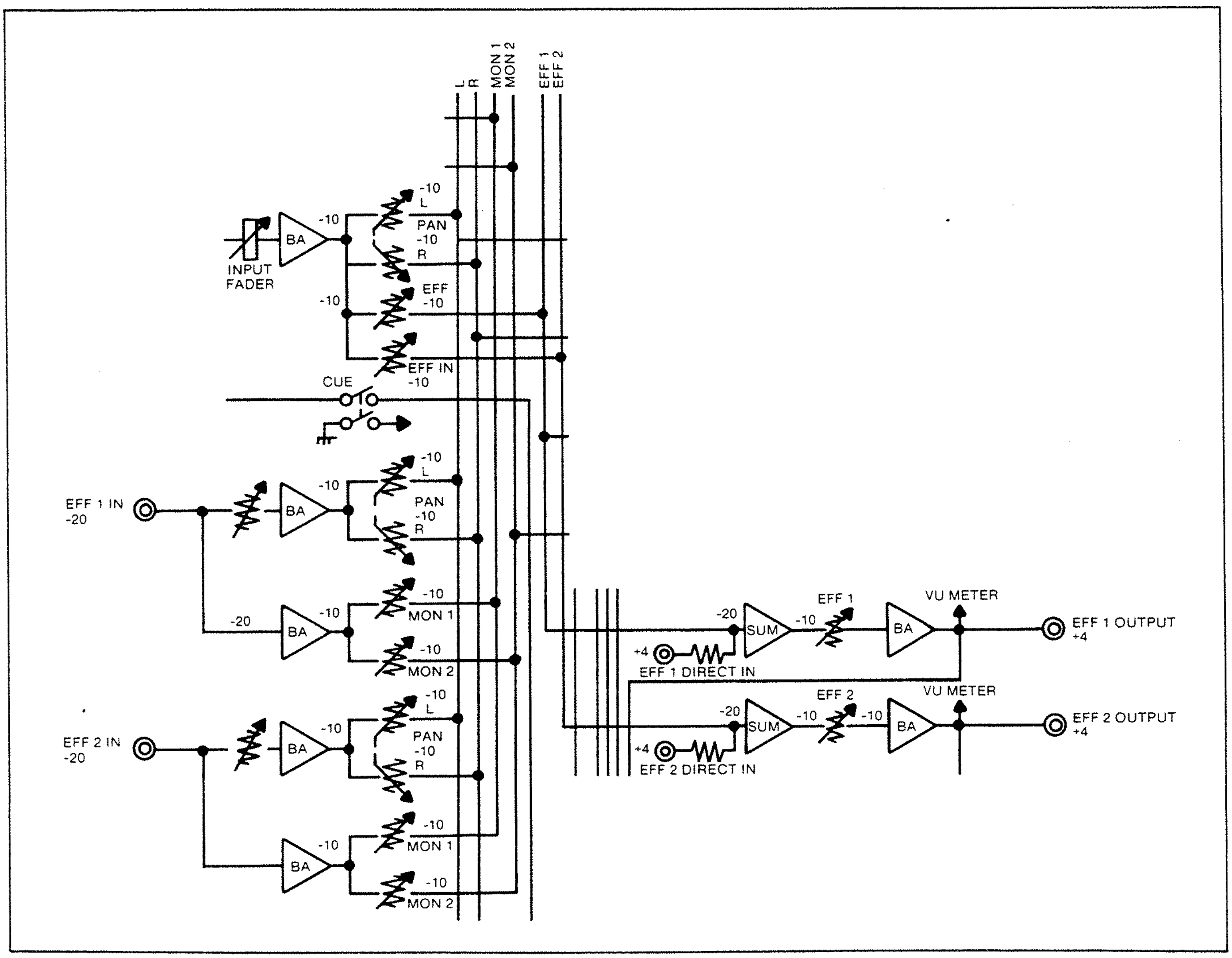
The Input Channel Effect control is post-EQ and post-fader. The 4216 has two Effect Outputs (with two Input Channel Effect controls labeled "Effect 1" and "Effect 2") and the 4216 has Effect "master" controls (labeled "Effect 1 Out" and "Effect 2 Out"). The 4208 and 4212 have a single Effect Output and no Effect master control.

What the Effect Mixes Do

The Effect mixes operates very much like the Monitor mixes, that is, they mix each Input Channel onto the Effect buses in a similar manner to the way the Input Channel Monitor 1 and 2 controls mix each Input Channel onto the Monitor 1 and 2 buses. There are a couple of important differences, however.

First, the Effect mixes are *both post-EQ and post-fader* (the Monitor mixes are post-EQ but *pre*-fader). This means that when you make changes in the settings of any of the three Input Channel Equalization controls, that action will also change that Input Channel in the Effect mixes, and, unlike the Monitor mixes, when you change the setting of the Input Channel fader, that action also changes the level of that Input Channel in the Effect mixes.

Second, *except on the 4216*, there are no "master" controls for the Effect mixes (like the Monitor 1 and Monitor 2 "master" faders). Instead, there are four "Effect Return" controls clustered just above the Program Left and Right



faders. These controls feed the Effect mixes into the Program Left and Right mixes and the Monitor 1 and 2 mixes, but *only after the Effect mixes have passed through any external effects device.*

An Exercise

If you haven't already done so, read "The Exercises." It's easy to connect an external effects device: just connect the device input to the Effect Out jack and the device output to the Effect Return jack. If you are using a 4216, connect your external effects device to the Effect 1 Output and Effect 1 Return. We'll assume, for this Exercise, that your external effects device is some type of reverberation system.

Play a tape of a group (again, an "uncluttered" tape with the least possible amount of reverb and effects) and set the Input Channel faders and the Program Left and Right faders for a comfortable listening level. Bring up the Channel 1 Effect control about half-way (Effect 1 if you have a 4216). Center the Effect (1) Return Pan control (the cluster of four controls above the Program Left and Right faders) and bring up the Effect (1) Return Program control slowly. *Reverb!* Try moving the Effect (1) Return Pan control from "L" to "R" and *the reverberation* will move from your Left loudspeaker to your Right Loudspeaker. Bring up the Effect (1) control on Channel 2 and you'll notice a subtle change. When you only had effects on Channel 1, only those instruments and voices coming into Channel 1 had any reverberation added to them. Now, with both Effect (1) controls up, you have added reverberation to the instruments and voices on both Channel 1 and Channel 2.

Now for another subtle effect. Turn down both Effect (1) controls on the Input Channels. Pan Channel 1 fully Left and Channel 2 fully Right so that you have true stereo coming from your loudspeakers. Center the Effect (1) Return Pan control and place the Effect (1) Return Program control about half way up. Now bring up the Channel 1 and 2 Effect (1) controls until you hear reverberation in your loudspeakers. The *reverberation* is a *mix* of the "reverberated" signals from *both* Channel 1 and Channel 2. In other words, the reverberation is in *mono*.

Turn the Effect (1) Return Pan control from left to right. This places the reverberation in either the Left or the Right loudspeaker (but does not cancel the monophonic nature of the reverberation signal).

If, for some reason, you sometime need a true stereo effects mix on a 4212 or 4208, you can use the Monitor 1 and Monitor 2 mixes, with two external effects devices, and bring the outputs of these effects devices back in through the Program Left and Program Right Direct In jacks. A stereo effects mix, of course, requires a pair of external effects devices (or a stereo effects device).

In the case of reverberation, however, a monophonic effects mix may actually be better than a true stereo mix. This is because *natural* reverberant sound from music in a room comes from random acoustic reflections which *have no apparent source.*

There are two more controls in the Effect Return cluster. These two controls, "Monitor 1" and "Monitor 2," mix the signal from the Effect bus into the Monitor mixes. If you'd like to experiment with these controls, set up your system the way you did when we first discussed the use of the Monitor mixes. Then, try mixing some reverberation into Monitor 1 and Monitor 2 using the Effect Return Monitor 1 and Effect Return Monitor 2 controls. In a live performance, you might do this when you are using one of the Monitor mixes to feed a system in a separate room. You might also do it when you are mixing monitors for a performer who asks for "a little reverb in my monitor."

How about putting a little reverberation (or some other effect) on an individual Input Channel? There's an easy way to do it! It's called the Insertion feature (see next section).

The Input Channel Insertion Feature

Block Diagram Closeup

The Input Channel Insertion feature includes an Insertion jack and an Insertion switch. An external device, connected to the Insertion jack, will be "inserted" into the Input Channel block diagram. The Insertion switch actually changes the position of this external device in the block diagram as shown in Figure A and Figure B. Thus, with the Insertion switch in the Pre position, the external device is pre-EQ. When the Insertion switch is in the Post position, the external device is post-EQ.

What Do We Mean by Terms Like "Pre-EQ" and "Post-Fader?"

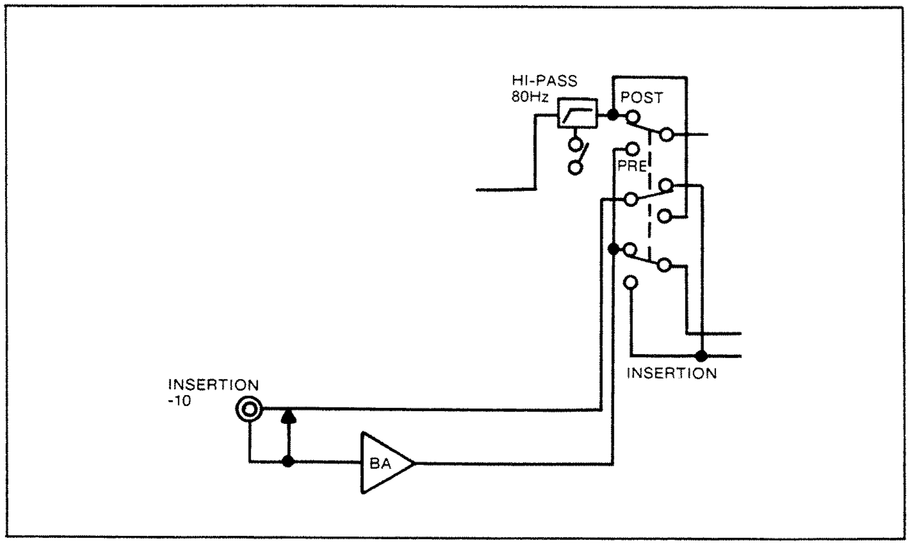
These terms, and others like them, refer to the position of a feature (usually a control) within the block diagram. For example, the Effect control is "post-fader" in the Input Channel. This means that it appears "post" (after) the fader in the block diagram. "Pre" and "post" refer to *the direction of signal flow* in the block diagram. (Signal flow is usually from left to right but may reverse direction or flow from up to down, etc.)

What the Insertion Feature Does

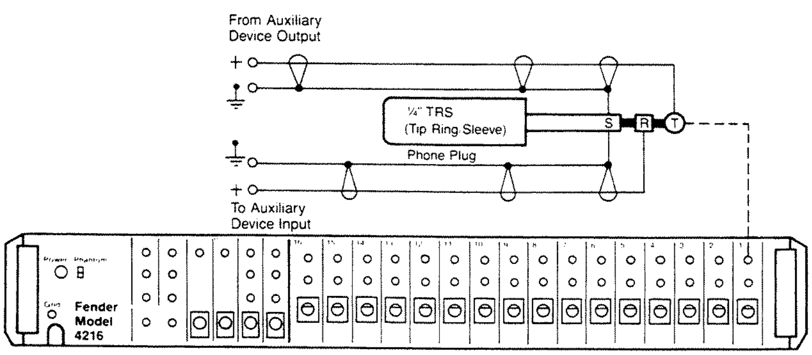
The Insertion jack accepts a low-line-level effects device and inserts it at a point in the Input Channel block diagram. When the Insertion switch (front panel, just below the Input Channel Trim control) is in the "Pre" position, the external device is placed after the Input Channel preamplifier but *before* the Input Channel Equalization control section ("pre-EQ"). When the Insertion switch is in the "Post" position, the external device is placed *after* the Input Channel Equalization controls ("post-EQ"). In both cases, the external device is "pre-fader."

An Exercise

Here, you get a chance to try out one of your effects devices. We'll assume it's a reverberation device that you



Connecting an Auxiliary Device to the Insertion Jack



want to use on an individual Input Channel. First, if you haven't already done so, read "The Exercises."

Be sure the external device will accept a +4dB input level and that its nominal output level is about -20dB. The device should have a 10k-ohm or higher input impedance (it will probably be much higher which is fine) and an output impedance of 10k-ohms or lower. Connect the device as shown. If you need a special cable for this connection, contact your Fender Dealer.

Before you turn on the external device, turn down the Program Left and Right and Monitor 1 and 2 faders. Many external devices produce a large "turn-on transient" (pop) when first turned on. Bringing down the faders prevents this potentially harmful transient from reaching your loudspeakers.

Now turn on the device and place the Input Channel Insertion switch in the "Post" position. Set the Input Channel fader and Program Left and Right faders at the levels you have been using in previous examples.

You may have to turn up the controls on your external device before hearing anything from your tape. This is because the external device has been "inserted" into the Input Channel and the signal from your tape machine has to pass through the external device as well as the Input Channel! Go ahead and experiment with the controls on your external device until you get a pleasing amount of reverberation at a comfortable listening level.

Now try changing the Input Channel Equalization controls and note the effect on the signal. With the Insertion switch in the "Post" position, the signal is equalized *before* it enters the external device. Now place the Insertion switch in the "Pre" position and try the Input Channel Equalization controls. In this position, the signal is equalized *after* is passes through the external device.

Pre and Post

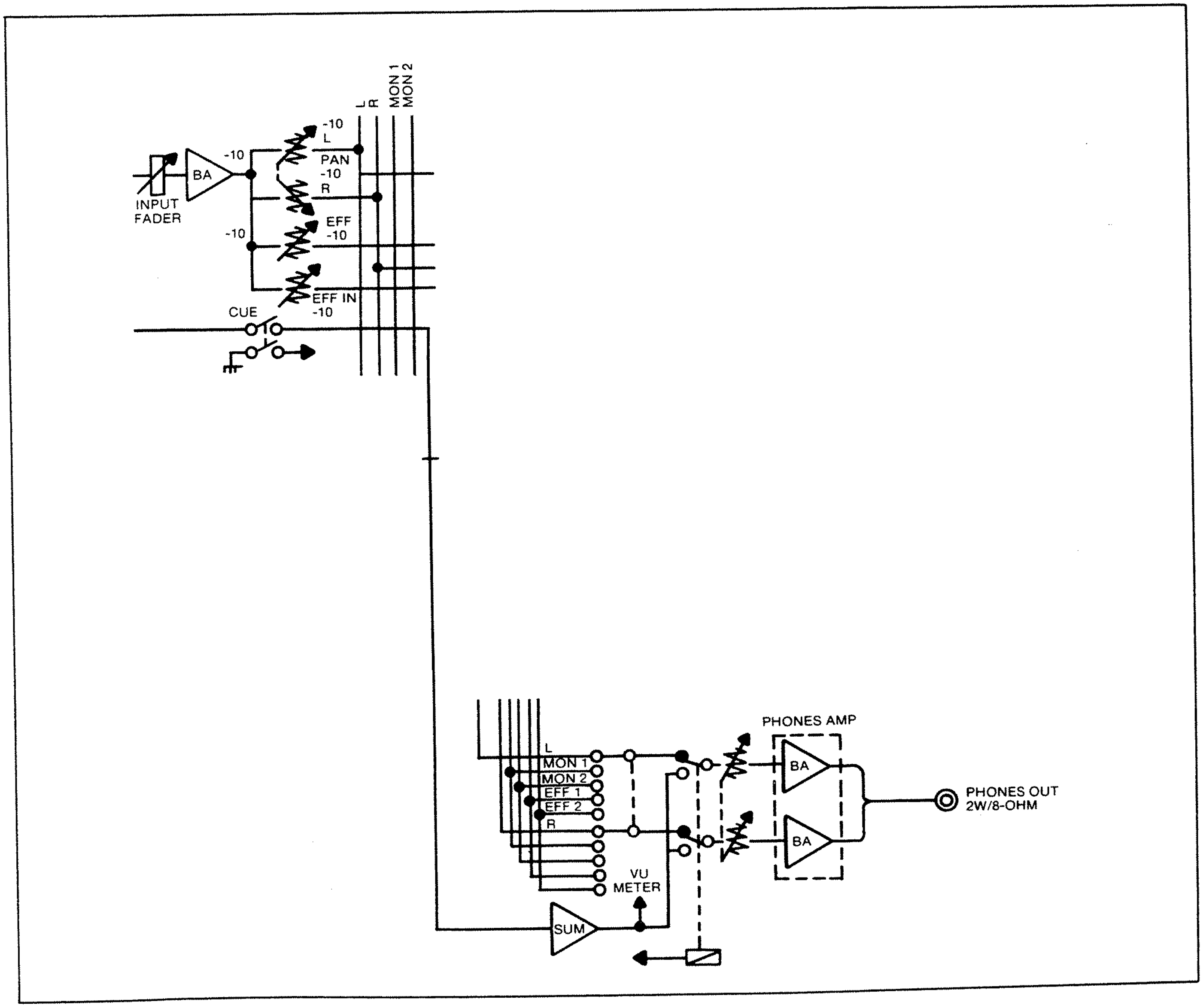
The differences between the Pre and Post positions of the Insertion switch may be subtle, but they could be important in some situations. For example, if the external device were a limiter, and you placed the Insertion switch in the "Pre" position, the signal would be equalized *before* it entered the limiter. If you then boosted the Low control, for more bass response, the limiter might begin tracking the increased bass frequencies rather than the entire signal. If you wanted to use the limiter to prevent Input Channel overload, this would be a useful effect. If you were using the limiter as a compressor to keep the overall level constant, this would probably be an undesirable effect. By placing the Insertion switch in the "Post" position, the equalization is done after the signal passes through the limiter which means that changes in the settings of the Input Channel Equalization controls will not have any effect on the operation of the limiter.

The Phones and Cue Systems

Block Diagram Closeup

The Phones (headphones) system is a specialized "monitor" that allows you to listen to the signals on any mix bus in your 4000 Mixer. Because the Phones system gets its signals directly from each mix bus, it is not affected by the "master" controls on any of the output channels. The Phones system includes a Phones Selector Switch and a Phones Level control and a two-channel power amplifier capable of up to two watts into each side of a pair of 8-ohm headphones (or a pair of monitor loudspeakers).

Because it has two Effect outputs, Model 4216 includes an extra position on its Phones Selector Switch. In addition, Model 4216 has a "Cue" System. The Cue system allows you to selectively listen to any single Input Channel or group of Input Channels through the headphones by simply depressing the Cue button on the Input Channel (below the Fader). Depressing the Cue button on any Input Channel activites a relay that disconnects the Phones amplifier from the Phones Selector Switch and connects it to the Cue bus.



Applications for the Phones System

You can use the Phones system, with a pair of 8-ohm headphones, to listen to the signals on any of your 4000 Mixer's mix buses. If you are listening to the Program buses (Phones Selector Switch set to "L,R") the Phones System feeds Program Right to the right channel of your headphones and Program Left to the Left channel of your headphones. Phones is not affected by the Program Left or Right (master) Faders nor is it affected by the Program Left or Right Graphic Equalizers. Thus, you can use the Phones system to "preview" the Program Left and Right outputs, even when the Program Left and Right Faders are fully down. (Just set the Phones Level control for a comfortable listening level.)

When the Phones Selector Switch is in the Monitor 1 position, *both* channels of your headphones will receive the signals from the Monitor 1 bus (in "mono"). The Monitor 2 and Effect positions work the same way (Effect 1 and 2 on the 4216). In the Monitor or Effect positions, Phones is not affected by the Monitor 1 or 2 (master) Faders or, on the 4216, by the Effect 1 or 2 Out controls. Thus, you can "preview" the Monitor (or Effect) buses, even when their Faders (controls) are all the way down.

This ability to monitor any of your 4000 Mixer's buses, even when their respective master controls are fully down, is an extremely valuable feature of your 4000 Mixer. For example, you could use the Phones System to check out the microphones before a performance without alterting your audience that something "technical" was going on. Keep the Program Left and Right Faders all the way down and have a helper talk into each microphone while you listen to the Phones (with the Phones Selector in the "L,R" position). You can even talk back to the person on stage, through your Monitor loudspeakers, by connecting a microphone to an unused Input Channel and bringing up only the Monitor 1 control on that Input Channel (plus the Monitor 1 master Fader). (This assumes you have a monitor amplifier and loudspeakers connected to the Monitor 1 Out jack.)

You can also listen to what's happening on stage. For example, by listening to the Phones you could find out when performers were beginning to assemble behind drawn curtains before a performance. And, the Phones System can allow you to listen to the Monitor mixes even when the location of your 4000 Mixer prevents you from actually hearing the monitor loudspeakers (a frequent occurance).

The Phones System can also be very useful if you are forced to mix from an enclosed space (like a lighting booth), where you can't hear the performance directly. Even with Phones, however, in a case like this it's still a good idea to get out into the audience area as often as possible to hear the actual sound the way the audience hears it.

Applications for the Cue System (Model 4216 Only)

The Cue System is actually a totally separate mix bus. You can connect any single Input Channel or any group of Input Channels to the Cue bus by depressing the "Cue" switch on any of the Input Channels. (You know when the Cue switch is "on" because a colored dot appears in the clear plastic top of the switch and the Cue LED, above the Phones Selector Switch, turns "on.") There is no "Cue" position on the Phones Selector Switch because depressing *any* Cue switch *automatically* connects the Phones system to the Cue bus! Releasing *all* of the Input Channel Cue switches automatically returns the Phones System to whatever bus you have previously selected with the Phones Selector Switch.

The Cue switch on each Input Channel is post-EQ and pre-Fader (just like Monitor 1 and Monitor 2) so you can listen to an Input Channel on the Cue bus even when the Input Channel Fader is all the way down. This makes the Cue system valuable for checking out microphones before a performance. Use a helper to talk into the microphones and simply depress the appropriate Input Channel Cue switch while listening to the Phones. You can leave the Input Channel Faders all the way down since they don't affect the Cue bus.

You can also use the Cue system to check out an individual microphone (or any other input) during a performance. For example, assume that you are mixing an unfamiliar singing group with one voice that stands out above the rest. Which one is it? Just depress the Cue switches, one by one, until you are listening to the stand-out voice on your headphones! If you want to reduce that Input Channel's Fader, you may wish to release the Cue switch and move the Phones Selector Switch to the "L,R" position to monitor the effects of the Fader change (if you keep listening to the Cue bus, you won't hear the effects of changes in Input Channel Fader position). However, if you want to change the Input Channel EQ settings to soften this stand-out voice, you can continue listening to the Cue system. That's because Cue is "post-EQ" and any change you make with the Input Channel Low, Mid or High controls (or the Mid switch) will change what you hear on the Cue bus.

Cue would also be useful if you were using your 4000 Mixer in a disco type system. A "disc jockey" normally "cues" one record while playing another to avoid interruptions in the program. Your 4216 Mixer's Cue System is set up for exactly this kind of operation. Simply keep the Faders all the way down on the Input Channels for your second turntable. Then depress the Cue switches for these Input Channels to preview the record on your second turntable while your audience is listening to the record playing on your first turntable. One of the examples at the end of the Manual describes the connections for such a system.

An Exercise

If you wish to try out your Phones or Cue system, we suggest that you play your tape machine through Input Channels 1 and 2 and try listening to the Phones system while adjusting the Input Channel Faders, the Input Channel EQ controls and the Input Channel Montor controls. Notice that the Program Left and Right and Monitor 1 and 2 Faders have no effect on the Phones level. Neither do the Effect 1 and 2 Out controls on the 4216.

To try out the Cue system on the 4216, select "L,R" on the Phones Selector Switch, pan Input Channel 1 fully "Left" and Input Channel 2 fully "Right." While listening to the stereo output on your headphones, depress the Cue switch on Input Channel 1. You are now listening to Input Channel 1 only, through both sides of your headphones. Then release the Input Channel 1 Cue switch and try connecting a microphone to Input Channel 3 and listen to the combination of your two channels of tape plus the microphone on the "L,R" Selector Switch Position (talk into the microphone yourself if you don't have a helper around). Now, depress the Input Channel 3 Cue switch. The stereo music goes away and you are listening to the microphone by itself. To confirm that you can, indeed, listen to more than one input at a time, depress the Input Channel 1 Cue switch and you will now have one channel of taped music plus your microphone.

The Phones Output and Loudspeakers

You can connect a pair of 8-ohm loudspeakers to the Phones jack using an adapter cable like the one shown here. A small pair of loudspeakers will work fine but larger loudspeakers are usually more efficient and therefore produce more sound level. The two-watt output from the Phones jack will produce a surprisingly loud SPL level from a pair of efficient loudspeakers which means that you may be able to use the Phones amplifier and a pair of efficient loudspeakers as "remote location" control room monitors, for example.

The Auxiliary Inputs

Block Diagram Closeup

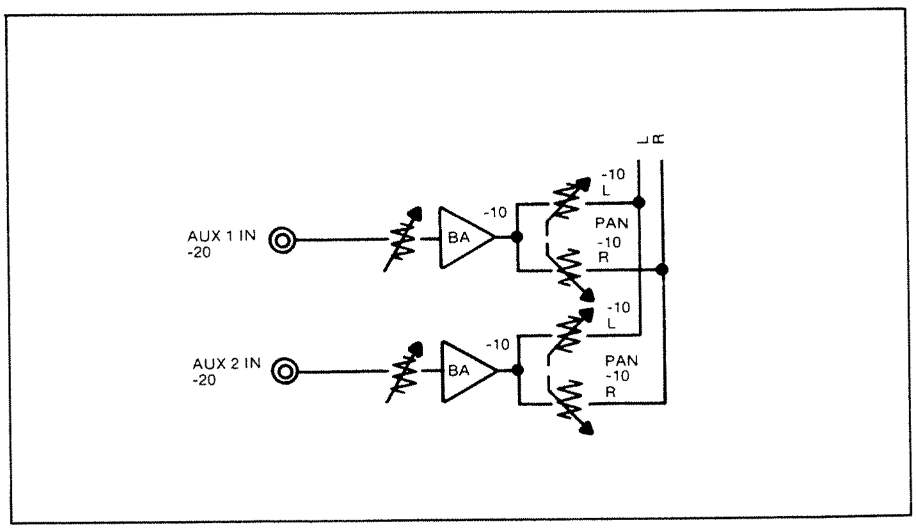
The Auxiliary Inputs ("Aux In" jacks) feed a low line level signal directly to the Program and Monitor buses.

Using the Auxiliary Inputs

You would probably connect a tape machine to the Aux In jacks unless it needed the Equalization or Effects functions of the Input Channels. You might also connect a tuner (radio) to the Aux In jack. Another use of the Aux In would be to bring a submixer into the Program buses (see the section entitled "Submixing with Your 4000 Mixer"). In brief, the purpose of the Aux In is to allow you to feed a low line-level signal into your 4000 Mixer without "using up" an Input Channel.

An Exercise

If you haven't already done so, read "The Exercises." Then, reconnect your tape machine to one of the Aux In jacks on the rear of your Fender 4000 Mixer. Place the Program fader at its "nominal" (0) position, start the tape and turn up the Aux In Program control until you reach a comfortable listening level. Try out the Aux In Pan control (which works just like the Input Channel Pan controls). Also notice that the Aux In signal is affected by the Program Graphic Equalizer, that is, the Aux In is "pre-EQ."



The "Insertion Out/Insertion In" Jacks

Block Diagram Closeup

The Insertion In jack is normaled to the Insertion Out jack.

What the Insertion Out/Insertion In Jacks Do

The Insertion Out/Insertion In jacks function as "insertion" points similar to the Insertion jack on each Input Channel. The Insertion Out/Insertion In jacks are "pre-EQ and pre-fader."

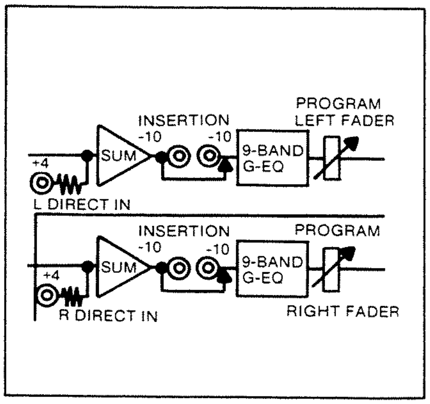
The Insertion In jacks can also function, alone, as inputs to the Mixer, although plugging a connector into either jack interrupts the normaled connection and thus interrupts the signal flow from the Program mix bus.

The Insertion Out jacks can function as outputs from the Mixer. Provided you "watch" impedances and levels, using these jacks will not affect the normal operation of the Mixer. (See "Impedance and Level Watching" and "The VU Meters.")

Using the Insertion Out/Insertion In Jacks

The Insertion Out/Insertion In jacks jacks can be used to insert an external device, such as a limiter, into the Mixer. Keep in mind that the Insertion Out/Insertion In jacks are low line level, -10dB. Choose your external devices appropriately. (See "Impedance and Level Watching" and "The VU Meters.")

If you are using an external limiter like a compressor, to help keep overall levels more constant (for different talkers, for example), you may wish to use it in the Insertion Out/Insertion In



jacks. In this position, the limiter will be both pre-EQ and pre-fader and the limiter will operate purely on the output from the Program mix bus and will not be affected by the Graphic Equalizer or the Program Fader.

Inserting an external effects device into the Insertion Out/Insertion In jacks may not work the way you expect. The output of most external reverberation devices, for example, is pure reverberation with no direct signal mixed in. The Effects mix, on the other hand, allows you to mix in a desirable amount of reverberation along with the direct, un-reverberated, signal. Thus, your external effects device should allow you to mix a desirable amount of effects with the direct, un-modified signal. Otherwise, you should use it in the Effects system of your 3000 Mixer.

An Exercise

If you wish to try out one or both of these connections, you may connect an external device, such as a limiter, and adjust the limiter while playing your tape machine. A limiter may respond very differently on live music or voice, so if you can, use a live source for this exercise, or try both your tape machine and a live source.

What is a "Normaled" Connection?

In a recording studio, the various electronic devices are usually connected to each other through a "patch bay" (a group of connectors in a panel). "Patch cables," inserted into appropriate connectors in the patch bay, connect the devices together.

Most of the time, the studio will use the same connection scheme, called the "normal" connection scheme. For this connection scheme, the patch bay is wired in a special way so that, *when all patch cables are removed, the patch bay is automatically set up for the "normal" connection scheme.*

To achieve this, recording studios use special connectors *which include a switch*. When no patch cables are inserted into the connectors, the switches connect the patch bay in the normal connection scheme. When the studio engineers desire to alter the normal scheme, they merely insert patch cables in the appropriate connectors which automatically opens the switch and breaks the normal connection. The concept and terminology for a "normaled" connection comes from this recording studio connection scheme and a "normaled" connection simply means any two jacks which are normally connected together but which are automatically disconnected when a plug is inserted into one of the connectors.

Submixing With Your 4000 Mixer

Block Diagram Closeup

In Example 1, a Fender 4208 Mixer is used as a submixer to a 4216. In Example 2, submixing is performed within a 4216 Mixer (no external submixer is used).

What is Submixing?

Sometimes it is useful to take a group of microphones (like drum mics or background vocals) and mix them together, through a separate fader, into the main mix. This process is called "submixing."

Example 1, Submixing With an External Mixer

Using an external mixer is probably the most common and most desirable way to submix. In this example, the Program Left and Right faders on the 4208 Mixer become "sub-masters" for the sources (mics or other inputs) connected to the 4216 Mixer. The Program Left and Right faders on the 4216 Mixer are the "master" faders and these faders *control all Input Channels on both mixers.*

Example 2, Submixing Within a 4000 Mixer

When you have only a few inputs to submix, it may be easier (and less costly) to do submixing within your 4000 Mixer. In the Example 2 block diagram, we show the Monitor 1 mix on a 4216 used as a submix. (Submixing within a 4208 or 4212 can be done the same way but these two mixers have fewer Input Channels and are less likely to be used as shown in this example.)

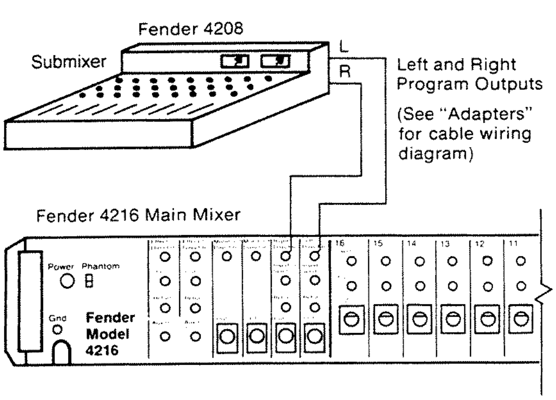
To use a 4216 Mixer this way, make the connections shown. Then, on the Input Channels you will use for the submix, turn the Input Channel faders all the way down. This keeps these Input Channels out of the Program Left and Right mix buses (until after they have been submixed). You then control the individual level of these Input Channels using their Monitor 1 controls. Do not use the faders on these Input Channels. Input Channel #16 is used, in this example, as the "sub-master," so you would use its fader is the sub-master fader to mix the chosen group of Inputs to the Program Left and Right Outputs. You can also use the Pan control on this last Input Channel to pan the sub-mixed channels into the Program Left and Right buses. *This setup will not work correctly if the "post-fader modification" has been performed on your 4216 Mixer.* (The post-fader modification must be performed by a qualified service technician.)

Alternately, you could bring the submixed channels from the Monitor 1 Out jack back into the Aux 1 In jack through a 24dB pad. This connection would free the last Input Channel for other uses.

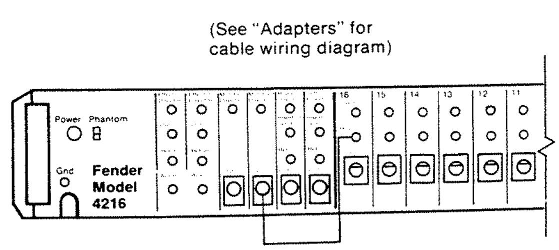
Other submixing connections are possible. For example, in a monophonic mix, you could pan a group of Input Channels all the way left and the rest of the Input Channels all the way right. Then patch the Left Program Out back into the Program Right Direct In jack. Now, the Program Left fader is a sub-master for those Input Channels which were panned left and the Program Right fader is the master fader for all the Input Channels.

In a similar way, you could do a monitor submix by patching the Monitor 1 Out jack into the Monitor 2 Direct In jack. Input Channels to be submixed are mixed into the Monitor 1 mix bus with their Monitor 1 controls. The Monitor 2 controls on the submixed Input Channels are kept all the way down. (The rest of the Input Channels use their Monitor 2 control with their Monitor 1 control all the way down.) The Monitor 1 fader, then, is the submaster, and the Monitor 2 fader is the Monitor master fader.

Submixing Example 1



Submixing Example 2

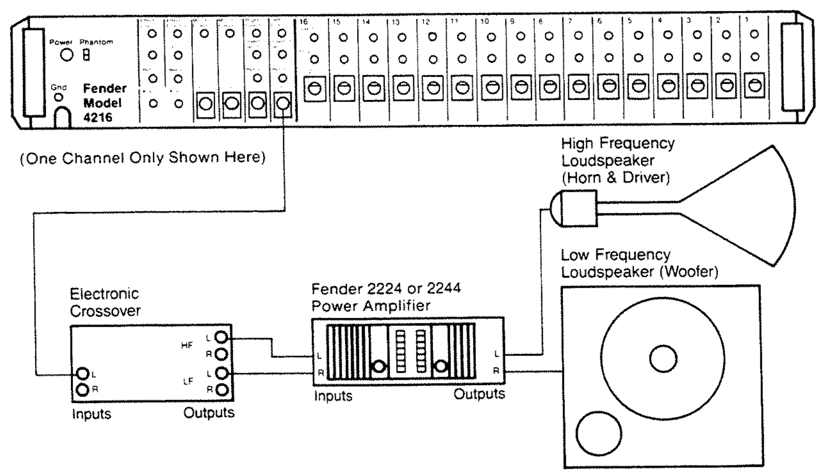


Section III, Special Connections, Biamplification and Other Topics

Special Connections

Many special connections are possible on your 4000 Mixer and you will doubtless come up with your own. Please observe two precautions, however. First, watch impedances and levels carefully (see "Impedance and Level Watching" — yes we mean "watching," not "matching"). Second, do not make a connection that forms a "feedback loop" as this could cause potentially damaging internal oscillations in your 4000 Mixer. A feedback loop is formed whenever you take a signal from the output of a section (such as the Program Left Output) and feed it back to the input of that same section (such as the Program Left Direct In jack or even one of the Input Channel input jacks).

A Biamplified System



Biamplification

In most music, especially modern music, the bass frequencies predominate. Thus, the bass frequencies use up most of the power in a power amplifier, "hogging" it away from the high frequencies. In a low to moderate level system, this causes few, if any problems and a non-biamplified loudspeaker system can probably serve your purposes very well. In a higher level system, where, for example, you normally use more than one power amplifier and more than one set of loudspeakers, the problem of the bass frequencies "hogging" the amplifier power can be significant. In this case, biamplification can help solve the problem. "Triamplification," of course, splits the frequencies into low, mid and high power amplifiers and loudspeakers. The same benefits apply to triamplification (or multi-amplification) as to biamplification.

The reason, of course, that biamplification works is that it allocates an entire power amplifier to those greedy bass frequencies so that they don't interfere with the midrange and high frequencies. The result can be a significant improvement in headroom (for lower distortion), *even when the total power output of your biamplified system is the same as a non-biamplified system.* More headroom, of course, translates into lower distortion and an overall "cleaner" sound quality.

There's another advantage to biamplification. Clipping distortion causes a lot of unwanted upper harmonics. If the bass notes in a non-biamplified system are high-level enough to cause the (single) power amplifier to clip, the upper harmonics (high frequencies) from the clipping distortion will pass through the passive crossover *and be reproduced by the tweeter.* Besides adding to the audibility of the distortion, in extreme cases, *this bass-note clipping can actually damage the tweeter!* In a biamplified system, even if the low-frequency amplifier is pushed into clipping, that clipping distortion *can never reach the tweeter* (reducing the possibility of tweeter damage). In addition, the bass loudspeaker cannot reproduce the upper harmonics of the clipping as well as the tweeter; thus the audibility of the distortion is reduced.

Can biamplification help in smaller systems? It will give you the same headroom improvements as in a larger system but there are disadvantages for smaller systems, too. First, biamplification costs more because it requires an added electronic crossover and a second power amplifier. Second, if you are planning to remove the passive crossover in an existing loudspeaker system so that you can biamplify that system consider that most loudspeaker system manufacturers include a certain amount of equalization in that passive crossover network to smooth out the response of the loudspeaker system. Thus, when you remove the passive crossover, you remove the equalization and the loudspeaker system's response may become ragged. You can probably use the Graphic Equalizers in your 4000 Mixer to help smooth out the response of your loudspeakers, but this complicates the use of the Graphic Equalizers for other purposes (such as compensating for a very "dead" room sound). Third, if you are using your system at a low to moderate level (the Clip LEDs almost never light), then the additional headroom provided by biamplification may not produce audible improvements in your sound.

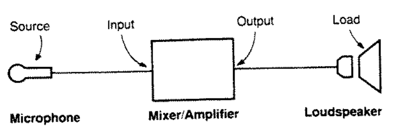
Thus, biamplification is a way to improve sound quality in a moderate to high level system. In fact, almost all high-level sound systems, like those used at large concerts, are biamplified. For smaller, low-level systems, however, a traditional non-biamplified system should provide excellent results at a lower overall cost.

Impedance and Level Watching

Impedance *Watching?*

Whatever happened to impedance *matching?* Don't worry; impedance matching is alive and well. Most active devices, however, do not require "matched" impedances. What they *do* require is impedance *compatibility*. In addition, all audio devices require (signal) *level compatibility*. Thus, "impedance and level *watching* means establishing and maintaining that impedance and level compatibility and that's what this section is all about.

Origin of Terms



Terms: Source, Input, Output, Load

In the "Origin of Terms" diagram, the microphone is the "source," the "input" is the input to the mixer/amplifier, the "output" is the output from the mixer/amplifier and the loudspeaker is the "load," but these four terms are relative. For example, the input to the mixer/amplifier can be called a "load" from the viewpoint of the microphone. And, the mixer/amplifier output can be called a "source" from the viewpoint of the loudspeaker.

Thus, the input impedance of the mixer/amplifier can be called the "load" impedance for the microphone and the output impedance of the r can be called the "source" impedance for the loudspeaker.

These four terms: source, input, output and load, and their relative nature are important to an understanding of impedance and level watching. As an example, consider a microphone whose "impedance" is 200-ohms. That impedance is actually the microphone's *internal* impedance and should be called the microphone's "source" or "output" impedance (the microphone is a source from the viewpoint of the mixer/amplifier).

That same microphone should probably be "loaded" with an impedance of 1500-ohms or higher. That "load" impedance is actually the "input" impedance of the mixer/amplifier (the input of the mixer/amplifier is a load to the microphone).

Thus, when you see any of the four terms "input, output, source or load," try to determine the device that is being used as a reference. If it is a microphone, the "load" impedance will be a mixer or mixer/amplifier input. If the reference device is a power amplifier, the "load" impedance will be a loudspeaker.

Impedance Compatibility

Impedance watching just means making sure that when we connect two devices together, they are *compatible* from an impedance viewpoint. Here are some rules to help you "watch" your impedances:

1) **Passive Devices** In the special case of a passive filter, like a loudspeaker crossover network and some (rare) passive graphic equalizers, you must *match* impedances. These devices are the origin of the familiar term "impedance matching." Impedance matching means that if the device is a loudspeaker crossover network and it has an 8-ohm low-frequency output impedance and an 8-ohm high-frequency output impedance, then you *must* connect an 8-ohm low-frequency loudspeaker and an 8-ohm high-frequency loudspeaker to that crossover network. Any other impedance, either higher or lower, will degrade the performance of the crossover network. (The *input* to a modern loudspeaker crossover network is designed for the very low actual output impedance of a modern power amplifier.)

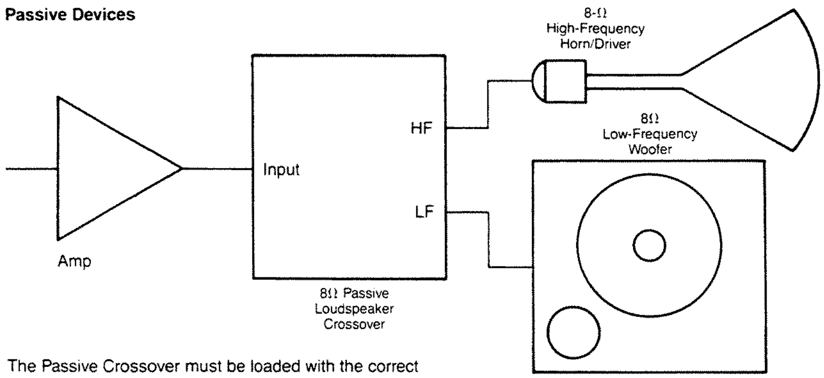
Those increasingly rare passive graphic equalizers have similar requirements. If such a device has a 600-ohm *input* impedance, then you must supply a *source* impedance of exactly 600-ohms. The same goes for the output. If the passive graphic has a 600-ohm *output* impedance, then you must supply a *load* impedance of exactly 600-ohms to assure proper operation of the graphic equalizer. In many cases, you will have to add "build-out" and "termination" resistors to match these impedances. For information on how to go about adding build-out or termination resistors, get a copy of "Sound System Engineering" by Don and Carolyn Davis or "The Audio Cyclopedia" by Howard M. Tremaine, both published by Howard W. Sams.

2) **Passive Sources** Impedance watching for a passive source like a dynamic microphone or guitar pickup simply means supplying a *compatible* load impedance for that device. The device specifications should guide you to the proper load impedance. A good rule of thumb for dynamic microphones is that the microphone load impedance (which is probably the *input* impedance of a mixer or preamplifier) should be at least 10 times the microphone's rated *source* impedance. Thus, for a 150-ohm (source impedance) microphone, the optimum load impedance would be 1500-ohms or higher. This requirement is satisfied by the input of almost all "lo-Z" mixer inputs including the Lo-Z inputs on your Fender 4000 Mixer. Note that the load impedance required by a "high-impedance" microphone is many times higher than the load impedance required by a low-impedance microphone. High-impedance microphones, therefore, can only be used with mixers having special inputs designed for these high impedances (like the "Hi-Z" inputs on your 4000 Mixer).

3) **Active Sources** Active sources like battery or phantom-powered condenser microphones should receive the same treatment as any other active device (next rule).

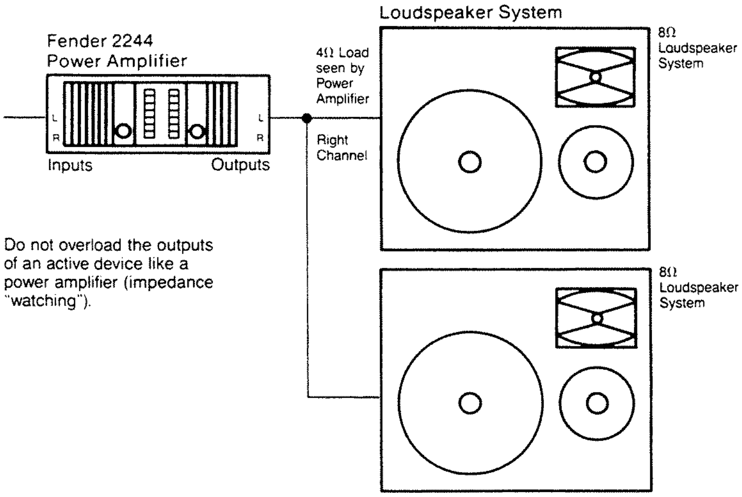
Impedance Watching

Passive Devices



The Passive Crossover must be loaded with the correct impedance loudspeakers (impedance matching).

Active Devices



Do not overload the outputs of an active device like a power amplifier (impedance "watching").

4) **Active Devices** An active device is one that uses batteries or AC power and has one or more tubes, transistors or ICs. Impedance watching for an active device means *not overloading its output*, that is, not connecting *too low* a "load" impedance to the output of the active device. A too-low impedance is an overload because the lower the impedance, the closer it is to a short circuit.

It's usually very easy to follow this rule because almost every active device comes with a set of specifications that tells you the value, in ohms, of the lowest allowable load impedance. This is usually called the "rated" or "minimum" load impedance. Incidently, in almost every case, it's okay to connect a *higher* than rated load impedance to any active device.

For the Fender 2224 and 2244 power amplifiers, for example, the "minimum" load impedance is 2-ohms. That means you can connect any impedance *down* to 2-ohms to these two powr amplifiers. Since an 8-ohm loudspeaker is greater than 2-ohms, it is an acceptable load. So is a 16-ohm loudspeaker. Two, 8-ohm loudspeakers in parallel equal a 4-ohm load, so this is also acceptable. Four, 4-ohm loudspeakers in parallel equal a 1-ohm load, so this is definitely *not* acceptable. (See "Calculating Series and Parallel Impedances.") Connecting a too-low load impedance to a power amplifier will cause the power amplifier's protection circuits to operate (which increases distortion) and may, in extreme cases, cause damage to the power amplifier or loudspeakers.

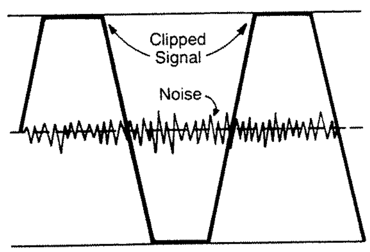
For a line-level active device, like a limiter, the same rule applies. If the limiter has a rated minimum load impedance of 600-ohms, you can connect the output of the limiter to the input of any device whose *input impedance* is 600-ohms or higher. (The input impedance of most active devices is considerably higher than 600-ohms.)

Some professional power amplifiers, on the other hand, have *input* impedances of 5000-ohms or lower. Connecting a hi-fi type tuner, with a 10,000-ohm minimum load impedance to the professional power amplifier, with its 5000-ohm input impedance would reduce the output level from the tuner and might also cause an increase in distortion.

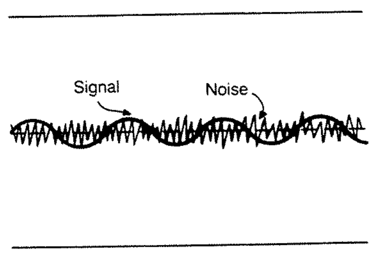
Impedance and Cable Length

One more aspect of impedance watching involves the effect of cable length on the frequency response of high-impedance microphones. *A too-long cable on a high-impedance microphone will cause a loss in high-frequency response*, that is, the sound from the microphone will be dull and voices will lack intelligibility. This results from the interaction between the capacitance in the cable and the high impedance of the microphone which form a low-pass filter (a low-pass filter passes only low frequencies which means that it attenuates high frequencies). The lower impedance of a low-impedance microphone also interacts with the capacitance of the cable but the effect is noticable only at very high frequencies (out of the audio range). A good rule of thumb is to avoid cables longer than 15 feet with a high-impedance microphone (some high-

Level Watching



If the signal level is too high, clipping distortion may occur.



If the signal level is too low, it may be "buried" in the noise.

impedance microphones will tolerate cable lengths up to about 25 feet). A low-impedance microphone, on the other hand, will perform properly with cables as long as 100 feet or more.

This same cable length consideration applies to line-level devices. Thus, cables connected to the Effect Outputs of your 4000 Mixer should be limited to 25 feet or less.

(Signal) Level Compatibility

Achieving level compatibility between devices means two things: avoiding too-high levels, which cause clipping distortion and avoiding too-low levels, which allow electronic noises (usually hiss).

There are three basic classifications of level in professional sound devices:

1) **Low level devices** (like microphones and pickups).

2) **Line level devices** (like limiters and graphic equalizers).

3) **High-level devices** (the output from a power amplifier).

The first rule of level compatibility, then, is to avoid connecting devices from different classifications *unless they are specifically designed for each other.*

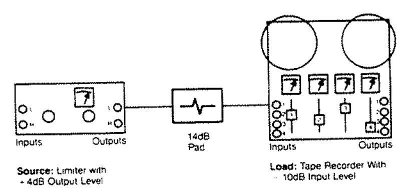
For example, you wouldn't connect a microphone directly to a power amplifier because the output of the microphone is too low. This connection wouldn't damage anything but you would get very low sound level and the noise from the power amplifier might be almost as high as the sound level.

As another example, you wouldn't connect the output from a power amplifier to the input of a limiter. The power amplifier output level is far too high for the input of the limiter. Thus, you would almost certainly experience severe clipping distortion (you might even damage the limiter).

Many devices, however, have an input that is compatible with one level and an output that is compatible with the next higher level. For example, the Input Channels of your Fender 4000 Mixer are compatible with low-level devices like microphones and your 4000 Mixer has both line-level and low and high-level line outputs. A power amplifier, like the Fender 2244, as another example, has a line-level input and a high-level output. Thus, you could connect the output of a limiter or graphic equalizer directly to the input of the Fender 2244. (You must always consider impedance, too, but the input impedance of the Fender 2244 is high enough to be impedance-compatible with the output of almost any professional line-level device.)

The situation is complicated somewhat by variations in the level of devices in a given catagory. For example, a condenser microphone, like the Fender P-1, has a higher output than a dynamic microphone, like the Fender D-1. Let us design a hypothetical mixer with inputs that are fully compatible with dynamic microphones like the D-1. If you plug the P-1 into the input of this mixer, the mixer's input circuits may be overloaded (clipping distortion). If, on the other hand, you design the mixer for the higher output level of a condenser microphone and then use it with a lower-level dynamic microphone, you may find that the output level from the mixer is too low and that there is an excessive amount of electronic noise (hiss).

Using a Pad



One solution to this problem is to design the mixer for the lower level microphone and provide a "pad" for the higher level microphone. A pad is a device to lower the level of a microphone or other low-level or line-level device (see "Pads and Transformers"). With your 4000 Mixer, you will never need a pad because the Trim control on each Input Channel allows you to adjust the gain of the Input Channel for whatever microphone you may be using. That way, you achieve both of our level-compatibility goals: avoiding clipping and avoiding electronic hiss noise.

The same kind of level-compatibility problems show up in line-level devices. Some line-level devices, mostly special effects devices, are designed for input and output levels as low as -20dB. Others, including the so-called "semi-pro" tape machines, are designed for input and output levels of -10dB. Most professional line-level equipment, however, is designed for input and output levels of +4dB.

The process of achieving compatibility with these line-level devices is similar to the process for low-level devices (like the microphones discussed earlier). Whenever possible, connect the output of a -20dB device to the input of another -20dB device (the same applies to -10dB devices and +4dB devices).

If this isn't possible, and the *source* device has a *higher* output level than the *load* device, use a pad to attenuate the level of the source device. For example, if the source is a +4dB limiter and the load is the input to a -10dB tape machine, you need a 14dB pad to achieve level compatibility. Without the pad, you risk clipping distortion. Just turning down the output of the source device probably won't solve the problem, either. This may result in that other level compatibility problem, electronic hiss noise. (For pad information, see "Pads and Transformers" or refer to "Sound System Engineering" by Don and Carolyn Davis or to "The Audio Cyclopedia" by Howard M. Tremaine, both published by Howard W. Sams.)

If the source device has a lower output level than the load device, you *could* place a line-level preamplifier between them to give you the required amount of gain. Or, you may wish to simply "give it a try." The worst that can happen here is additional hiss noise and it may be tolerable in many cases.

Fortunately, your Fender 4000 Mixer is equipped with a variety of line-level inputs and outputs. For example, the Effect Return jack(s) and the Aux 1 and 2 In jacks are planned for a -20dB device (many reverberation and other effects devices operate at this level). The Direct In jacks, on the other hand, are planned for +4dB input levels and the Insertion Out/Insertion In jacks are planned for -10dB levels.

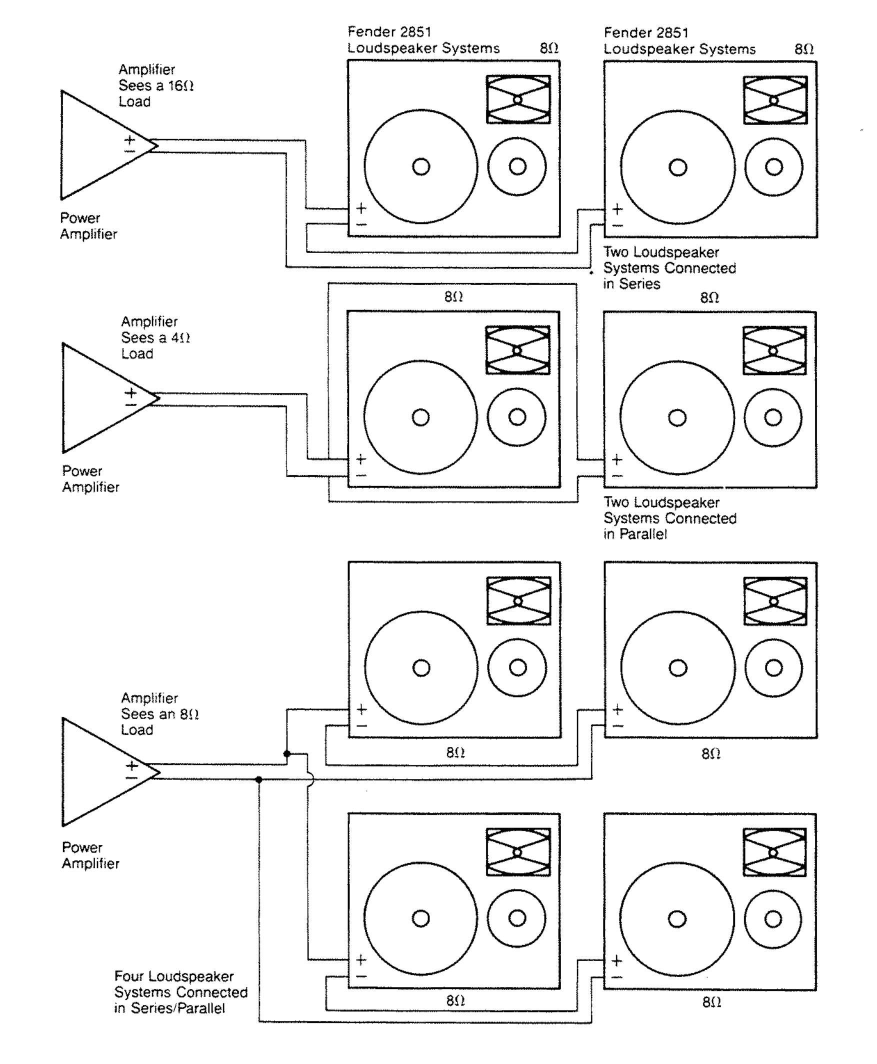
By adjusting the Trim control, the Hi-Z inputs on your 4000 Mixer can be used for line-level devices (of *any* level up to +4dB) as well as Hi-Z microphones.

Calculating Series and Parallel Impedances

For the most part, the only thing we ever connect in series or in parallel in audio is loudspeakers. In rare cases, we may connect two microphones in parallel by using a "y-cable" but this may degrade the performance of the microphones (so don't do it unless you have to). Neither is it a good idea to connect microphones in series (the adapter cables are difficult to make anyway).

Loudspeakers, on the other hand, are often connected in parallel and sometimes connected in series. Connecting loudspeakers in series will degrade their performance somewhat (it lowers the "damping factor" and thus degrades transient response) but connecting them in parallel will cause no degradation at all *provided the power amplifier is not overloaded.*

Series, Parallel and Series/Parallel Connections



The only way you can know whether or not the power amp is overloaded, of course, is to calculate the *total* impedance connected to it. Here's how:

Series Impedance Calculations

Series impedances (see diagram) are easy. You just add them up. Thus, two, 4-ohm loudspeakers connected in series result in a *total* impedance of 8-ohms. Three, 16-ohm loudspeakers connected in series result in a *total* impedance of 48-ohms.

Parallel Impedance Calculations

Parallel impedances (see diagram) combine according to this formula:

Formula 1: Total Impedance =

$$\frac{1}{(1/Z_1 + 1/Z_2 + 1/Z_3 + \ldots + 1/Z_n)}$$

Where Z1 is the first impedance, and Zn is the "nth" or last impedance.

This formula works on any group of parallel impedances, even if they are different values. Fortunately, it's a lot easier if there are only two impedances. Then the formula reduces to this next one:

Formula 2: Total Impedance (Two Impedances in Parallel) =

$$(Z_1 {*} Z_2) / Z_1 + Z_2)$$

This formula, like the previous one, works even if the two impedances are of different values. The most common situation, in audio, however, is even easier. When we parallel two or more loudspeakers *which all have the same impedance*, the *total* impedance is just the impedance of one loudspeaker divided by the number of loudspeakers in parallel. In other words, paralleling two, 8-ohm loudspeakers results in a 4-ohm total impedance. Paralleling two, 16-ohm loudspeakers results in an 8-ohm impedance. Paralleling three, 8-ohm loudspeakers results in a 2.67-ohm total impedance (8 divided by 3).

Series/Parallel Calculations

Occasionally, we connect a set of loudspeakers in "series/parallel" (see diagram). To get the total impedance in the first series/parallel diagram, you calculate the impedance of each series group (just add them) and then combine those groups in parallel (by using one of the parallel formulas). To get the total impedance in the second series/parallel diagram, you calculate the impedance of each parallel group and then combine these groups in series. In each case, you break down the connection into groups, calculate the impedance of each group and then treat each group as if it were a single impedance.

The most common use of series/paralleling in audio is to take a group of 4 loudspeakers (all having the same impedance) and connect them in series/parallel. The resulting impedance is exactly the same as a single loudspeaker. Four, 8-ohm loudspeakers in series/parallel, for example, results in a total impedance of 8-ohms.

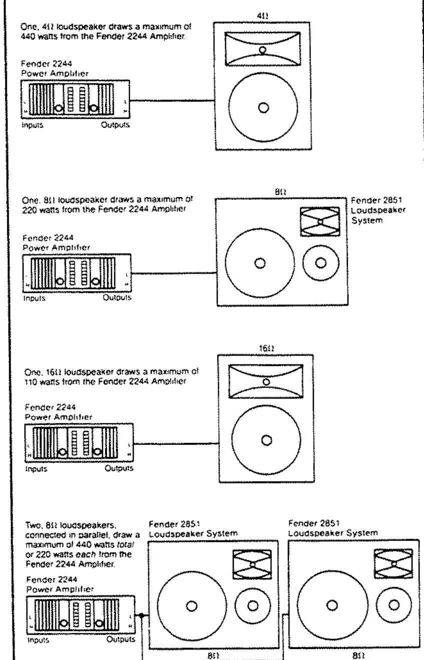
Impedance and Power Transfer

It's important to understand what happens to the power output of an amplifier when you connect different impedances to it. You have to know the *rated power* output of the amplifier and its *rated load impedance*. That rated load impedance, of course, will often be the amplifier's *minimum* acceptable load impedance.

In addition, you should know the loudspeaker's true *minimum* impedance as well as its "rated" or "nominal" impedance. Normally, you will use the loudspeaker's "nominal" impedance to make "impedance watching" calculations like those described in the next paragraph. A loudspeaker's *minimum* impedance, however, can fall significantly below its *nominal* impedance and a loudspeaker with an extremely low minimum impedance could overload your power amplifier.

The Fender 2224 and 2244 power amplifiers are designed to accept impedances as low as 2-ohms

Power Transfers (Typical of any Power Amplifier)



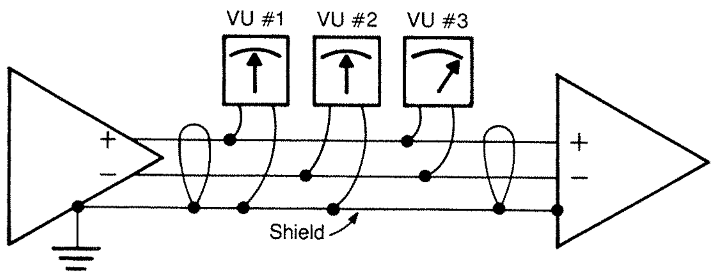
because of the very low minimum impedance of some loudspeakers. Many 8-ohm loudspeakers (8-ohms is the "rated" or "nominal" impedance), for example, have *minimum* impedances of 6-ohms or even as low as 5-ohms. Two of these loudspeakers in parallel would have a minimum impedance of 2.5-ohms which would still be within the safe limits for your 4000 Mixer.

The easiest way to describe the change in power output with different load impedances is to take an example. Let's use the Fender 2244 Power Amplifier which is rated at 440 watts per channel into a 4-ohm load. (The 2244 has a *minimum* load impedance of 2-ohms even though its 440 watt power rating is at 4-ohms.) 440-watts into 4-ohms means exactly that. If you connect a 4-ohm loudspeaker to one channel of the 2244, the amplifier will produce as much as 440 watts into that loudspeaker. If you connect two, 8-ohm loudspeakers (in parallel) to one channel of the 2244, the 2244 will, again, produce as much as 440 watts into the resulting total impedance of 4-ohms. *Each, 8-ohm, loudspeaker in this example, will receive exactly one-half of the total power, or a maximum of 220 watts.*

If you now connect a single, 8-ohm, loudspeaker to one channel of the 2244, that loudspeaker will still only receive a maximum of about 220 watts. (The actual power will be slightly higher.) If you connect a single, 16-ohm, loudspeaker, it will receive a maximum of about 110 watts. In other words, doubling the load impedance *halves* the power output of a power amplifier. Conversely, halving the load impedance *doubles* the amplifier's power output. Remembering this simple relationship can help you make sure that a loudspeaker and power amplifier will be compatible in terms of impedance and power levels.

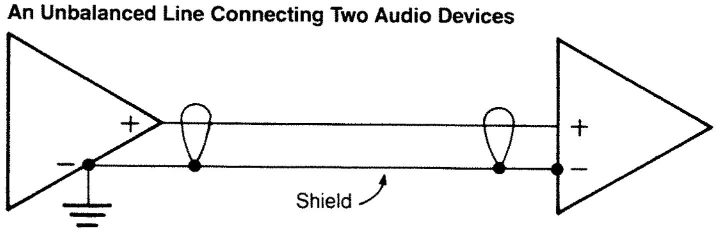
Balanced, Floating and Unbalanced Lines

A Balanced Line Connecting Two Audio Devices



On a true "balanced" line, VU #1 and VU #2 will read exactly ½ the value of VU #3. On a "floating" line, the readings of VU #1 and VU #2 are undetermined. Both balanced and floating lines help reject noise pickup.

An Unbalanced Line Connecting Two Audio Devices



Understanding Balanced and Unbalanced Lines

(The term "line" refers to a cable or connection between two pieces of audio equipment.) Every audio signal requires at least two wires. In an unbalanced line, the shield (outer conductor) is also one of the audio signal wires. Thus, an unbalanced line needs only the shield and one additional wire (a total of two wires). In a balanced line, the shield does not carry audio signal. Thus, a balanced line requires two additional wires to carry the audio signal (for a total of three wires).

In a true balanced line, the audio signal level is "balanced" between the two audio wires and the shield. Thus, two VU meters, each connected between one of the audio wires and the shield, would display the same reading (½ the total audio signal level). The primary advantage of a balanced line is that it is much less likely to pick up external electronic noises (hum, buzzing, static, radio stations) than an unbalanced line.

This reduced noise pickup is very important for low-level devices like microphones. Thus, the Lo-Z inputs on your Fender 4000 are all balanced (transformer-less) to allow you to use balanced microphones (all the Fender microphones are balanced) and the Program and Monitor Outputs on your Fender 4000 are all balanced to allow you to use balanced, line-level devices.

Most transformer-coupled "balanced" devices are actually *floating*. For example, the Program and Monitor Outputs on your 4000 Mixer are transformer-coupled, "floating" outputs. On a floating line, connecting two VU meters between each of the two wires and ground would show undetermined results — each wire "floats" at an undetermined voltage from ground. In most cases, floating lines provide the same advantages as balanced lines. In other words, the difference, for our purposes is academic and the two terms may be treated as equivalent.

Connectors and Cabling

As simple a subject as this may seem, *faulty connectors and cabling are the source of a majority of sound system problems.* Well-made cabling, of the proper type, with the right connectors for the job, on the other hand, will keep your system operating at maximum efficiency with a minimum of noise pickup. Here are some tips on how to do it right.

Some General Notes on Cable

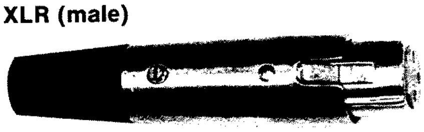
A "cable" is a group of two or more wires, usually in a single outer (insulating) sheath, and designed for a particular function.

Cables for portable audio systems should always be made from stranded, not solid, wire. Solid wire cables will break after the repeated flexing of portable usage. Shields should be braided wire, not foil, for the same reason.

Some General Notes on Connectors

There are only a few types of connectors in general use in professional audio. The most common of these are:

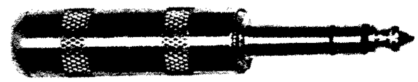
XLR (male)



XLR (female)

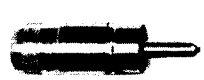
1) "XLR" Type Connectors The term "XLR" was first used by the Cannon Company but has almost become a "generic" label for these high-quality audio connectors, now made not only by Cannon but also by Switchcraft, Neutrik, ADC and others. XLRs are the connector of choice for any balanced low-level or line-level audio signal.

¼" T/S Phone (Tip/Sleeve)



¼" T/R/S Phone (Tip/Ring/Sleeve)

2) ¼" Phone Plugs The term "phone" comes from the telephone company who used a type of phone plug in their early, non-automated, switchboards. Recording studio patch bays are close relatives of these telephone switchboards and, again, use some type of phone plug. The most common type of phone plug used in pro audio has a ¼" diameter shank and comes in two-wire (known as "Tip/Sleeve" or "T/S") and three-wire (known as "Tip/Ring/Sleeve" or "T/R/S") versions. ¼" phone plugs are commonly used for instrument amplifiers, hi-Z microphones and are the type used on your 4000 Mixer. Beware when you purchase a blister-pack phone plug, however, because smaller diameter varieties exist (and won't work in most audio equipment). Smaller varities of phone plugs, like those used on portable hi-fi equipment, are seldom used in pro audio. Unlike XLRs, which are almost invariably high-quality, the quality of commercially available phone plugs can vary widely. Your best bet is to purchase a well-known brand name at a reputable *audio* store (like your Fender dealer).



"RCA" Phono

3) "RCA" Type Phono Plugs Note the term *phono*, not *phone* indicating that these plugs got their start on phonographs (assumably those manufactured by the RCA company). Phono plugs, or "RCAs," are used primarily on hi-fi equipment but you may need to use them to, say, adapt a hi-fi tuner to an input on your Fender Mixer. Phono plugs, however, are fragile and would not make good general purpose pro audio connectors.

Cable and Connectors for Microphones and Other Low-Level Devices

Lo-Z balanced microphones (most professional microphones, including the Fender series microphones are in this catagory) use shielded, two-wire cable and XLR type connectors. Hi-Z (unbalanced) microphones usually use a ¼" phone plug connector. Microphone cable should have a flexible, tough outer sheath, a braided shield and stranded inner wires.

Although the XLR type connector is an industry standard for lo-Z balanced microphones, unfortunately, the wiring of these connectors is not completely standardized. While pin 1 on the connector is almost always connected to the cable shield, some manufacturers use pin 2 as "high" or "+" and other manufacturers use pin 3 as "high" or "+" (with the remaining pin "low" or "-"). This means that if you use two microphones, from different manufacturers, with different "+" pins, the two microphones will be "out-of-phase" with each other and that can cause undesirable effects like "comb filtering" when the microphones are very near each other and both picking up the same source (see "What Do We Mean By Phasing and Polarity"). About your only defense against this problem is to make sure you *know* which is the "+" pin on any microphone you use (and on any mixer you use!) and try not to use both types in the same system. Your Fender dealer may also be able to help you resolve this problem with a special type of adapter known as a "polarity reversal" or "phase-reversal" adapter.

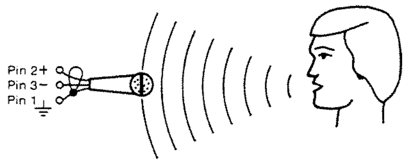
What Do We Mean by Phasing and Polarity?

Polarity is an easier concept, so let's start there. Every electrical signal has a "polarity." A transistor radio battery has a "+" and a "-" terminal. Put it in a radio with its "+" terminal connected to the radio's "+" terminal and the battery's "polarity" is said to be "normal." Turn the battery around so that it's "+" terminal is connected to the radio's "-" terminal and the battery's polarity is said to be "reversed."

The polarity of a microphone is a bit more complex, but similar in concept. A sound wave in the air consists of alternate layers of compressed and rarefied (uncompressed) air. The compressed layers are defined as "positive pressure"; the rarefied layers are defined as "negative pressure." When a positive pressure wave hits the diaphragm of the microphone, it produces a positive voltage on one of the pins of the microphone connector (relative to one of the other pins). That pin is the "+" pin of the microphone connector (the other pin is the "-" pin). Thus, if pin 2 of the microphone connector is specified as the "+" connector, you know that a positive pressure wave striking the diaphragm of the microphone will produce a positive voltage on pin 2 of the connector (with pin 3, the remaining pin, used as the reference or "-" pin, and pin 1 the shield). If you plug this microphone into a mixer which also uses pin 2 as its "+" pin, the polarity will be "normal." If the mixer uses pin 3 as its "+" pin, the polarity will be "reversed."

Reversed polarity between a microphone and a mixer is seldom a problem. Using two microphones with different polarity standards can, on the other hand, be a real problem, at least when the microphones are close together and both picking up the same source. The reason is that the same positive pressure wave, striking the diaphragms of both microphones, will cause a *positive* voltage on pin 2 of one microphone and a *negative* voltage on pin 2 of the other microphone! When these two microphones are mixed together, inside a mixer, the positive voltage from one microphone partially or completely *cancels* the negative voltage from the other microphone and you end up with bad sound or no sound at all!

Microphone Polarity



A positive pressure wave hitting the diaphragm of a Fender Microphone will produce a positive voltage on Pin 2 with respect to Pin 3.

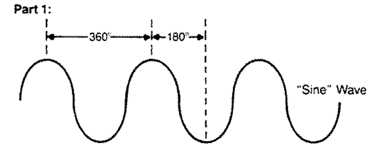
You can experience this effect by taking two microphones of the same model and brand and using a polarity reversal adapter with one of them (see "Adapters"). Plug each microphone into an Input Channel on your 4000 Mixer and set the controls on both Input Channels the same. Start, however, with one fader all the way down and adjust the other fader for a comfortable listening level. Now, holding the two microphones very close together, talk into both of them and bring up the second fader. As the level of the second fader approaches the level of the first fader, the sound level goes *down*, not up!

You can check the polarity of unknown microphones in a similar manner. Select a standard microphone (one whose polarity is known) and check all the others against this standard just like you did in the above exercise (omit the polarity reversal adapter). If the unknown microphone is out-of-polarity with the standard microphone, the level will go down as you bring up the second fader.

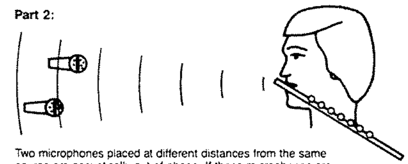
You can use a polarity reversal adapter with an out-of-polarity microphone to bring it back to "standard" polarity. Or, it's okay to use an out-of-polarity microphone without the polarity reversal adapter. Just don't use two microphones with opposite polarity to mic the same (or a nearby) instrument or voice.

Phase

Part 1:

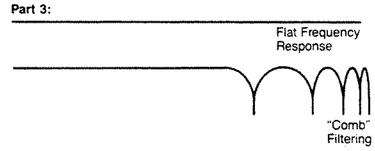


Part 2:



Two microphones placed at different distances from the same source are acoustically out-of-phase. If these microphones are mixed together (in any mixer) they will produce an effect known as comb filtering (see below).

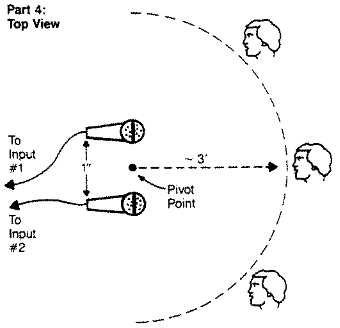
Part 3:



"Phase" and "polarity" are related, but different, concepts even though the terms are often used inter-changably. When you hear the term "out-of-phase," for example, that probably means "reversed polarity" (reversed polarity is a more technically accurate description of the problem discussed in the previous paragraph).

"Phase" is a way of measuring, in *degrees*, the distance between two points in a sound wave, or of two points in the corresponding electrical signal. In the first part of the diagram, you can see that the phase difference between one positive portion of this sound wave and the next positive portion of the sound wave is always 360 degrees. The wave-form shown is called a "sine wave" and is typical only of very pure tone instruments like a flute. Most sound waves are much more complex. Yet, for a discussion of phase, this waveform is useful because of its simplicity. If the distance between one positive portion of the wave and the

Phase



To hear the effects of acoustic phase cancellation, connect two mics as shown, bring their faders up to the same level and have a friend talk into them while walking the path shown.

next positive portion of the wave is always 360 degrees, then the distance between a positive portion and the adjacent negative portion of the wave must be 180 degrees. If you reverse the polarity of a microphone, that causes the positive electrical signal from the microphone to become negative and vice-versa. Thus, a polarity reversal is very similar to a 180 degree phase shift (moving from the positive portion of the wave to the negative portion of the wave). This is the origin of the use of "out-of-phase" or "180 degrees out-of-phase" to mean "polarity reversal."

A true "phase shift," however, can be anywhere from 0 degrees to 360 degrees (or even a large multiple of 360 degrees). In the second part of the diagram you see a flute player and the sound coming from the flute. Two microphones are placed at different distances from the flute. Because these two microphones are at different points in the sound wave, we can say that they are truly "out-of-phase" with each other *even though their polarity is the same*.

This kind of out-of-phase condition causes the problem known as "comb filtering." Comb filtering causes a flat frequency response to look like the third part of the diagram. If you want to hear this kind of problem, take your two microphones again and connect them to two Input Channels of your 4000 Mixer but this time don't use the polarity reversal adapter. Place the microphones a foot or so apart on two stands, bring the faders on both Input Channels up to the same point, and have a friend talk into both microphones at the same time from a position like that shown in the fourth part of the diagram. Now, have your friend keep talking but walk around the pivot point shown while facing the microphones. The sound quality will change dramatically as your friend moves around the microphones. This effect is usually called "phasing" (accurately!) and the special effects devices called "phasers" duplicate this effect electronically. In sound systems, however, the phasing effect is undesirable and the best way to prevent it is to use a single microphone for each source (for each singer or group of singers and for each instrument). For individual singers or instruments, ask the performer to stay as close as possible to the microphone. This allows you to turn down the fader on that microphone which minimizes the pickup from other sources (and therefore minimizes phasing problems from those other sources).

This same problem happens between two loudspeakers and you can experience it by playing a constant source at equal level through two loudspeakers separated by a few feet. Use the inter-station noise from a tuner (or a "pink noise" source if available). Or use a single microphone (in another room) picking up a sustained guitar or piano chord. Walk between the two speakers (at some distance from them) and listen to the results. The radio station or pink noise source will sound like a jet plane taking off (with similar effects on the guitar or piano).

It's almost impossible to completely eliminate this problem in a real sound system, but you can minimize it by using each loudspeaker to cover a specific area of the room. In other words, don't overlap the coverage any more than you have to. It's those overlap areas where the phasing problems occur. Fortunately, unless you have critical listeners in your audience, and you are reinforcing sustained chords, and the listeners are moving from point to point, the audible effects of phasing are not great. Still, for the best sound quality, everywhere in a room, it's a good idea to try to minimize these effects, before a performance begins.

Microphone "Snake" Cables

A "snake" cable is actually a group of microphone or line-level cables all in one outer sheath. These cables use foil shields to reduce their overall diameter to a reasonable size. Because of the fragility of the foil shields in a snake cable (and because of the high cost per foot) you must take extra care in their handling. Avoid sharp bends in these cables. Also avoid placing heavy objects on a snake cable or rolling heavy carts across them. Snake cables can be a money saver and time saver when you are setting up a large, multi-microphone system. Ask your Fender dealer for help in selecting a snake for your system.

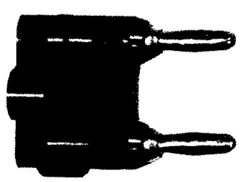
Cable and Connectors for Line-Level Devices

Line level devices normally use the same type of cable and connectors as microphones and other low-level devices. That is, balanced line-level devices normally use XLR type connectors and unbalanced line-level devices normally use ¼" phone plug connectors. Some balanced line-level devices use three-conductor, ¼" Tip/Ring/Sleeve (T/R/S) connectors.

Polarity is, again, unfortunately, not standardized on balanced line level devices using XLR connectors. Either pin 2 or pin 3 may be the "+" pin (pin 1 will almost always be the shield). Thus, you should check the polarity of any unfamiliar device you may be using.

Cable and Connectors for Loudspeakers

Speaker cable carries *much* higher levels of electrical power than either microphone or line-level cable. For this reason, speaker cables use larger guage wire. Typical speaker cable uses anywhere from #18 guage wire to as large as #10 wire. #18 guage wire is suitable only for low-level loudspeakers (like the hi-fi speakers in your den). #16 guage wire is suitable for short runs (less than 25 feet) of low to medium level pro audio loudspeakers. #14 guage wire is suitable for most pro audio work unless loudspeaker runs are longer than about 75 feet. In that case, #12 guage wire should be used. For very long runs of high-power speaker cable, use #10 (or even #8) wire. A better way to handle long speaker cable runs, however, is to move the power amplifier closer to the loudspeakers and run line-level signals over the long distance.



Dual Banana

Loudspeaker connectors are another story. The most common loudspeaker connector in pro audio is the ¼" phone plug. Unless you use very high-quality phone plugs, however, they are actually not very suitable for the high-current use they get in pro audio. Thus, ¼" phone plugs are only suitable for low and medium level loudspeakers (perhaps up to 200 watts or so per loudspeaker). For higher power loudspeakers, a higher current connector, like a "dual banana" connector is a good choice. XLR connectors are sometimes used for loudspeaker connectors but their current capacity is limited, too and they should not be used for higher power capacity systems.

Adapters

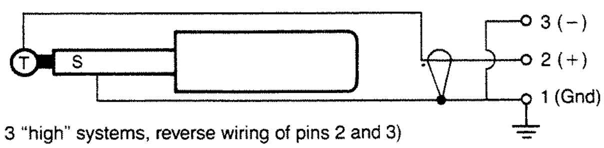
If all audio devices used the same connector, we wouldn't need adapters. Suffice it to say that we do need them — often! Take care in using the adapters shown here. They will, in most cases, allow you to *connect* one type of device to another. They do *not* help you maintain impedance and level compatibility! In some cases, you may need a pad or transformer (or even a preamplifier) along with an adapter in order to be able to connect two audio devices together. Consult your Fender dealer if you need help with these connections.

"RCA" Phono to ¼" T/S Phone

¼" T/R/S Phone to XLR (male)

Adaptor Wiring Diagrams

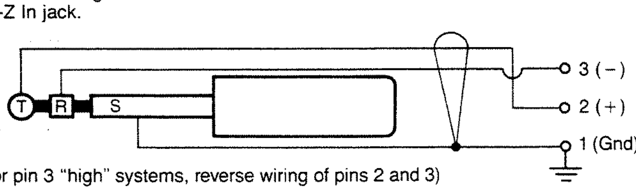
T/S Phone Plug to XLR Connector (for Pin 2 "high" systems)



(for pin 3 "high" systems, reverse wiring of pins 2 and 3)

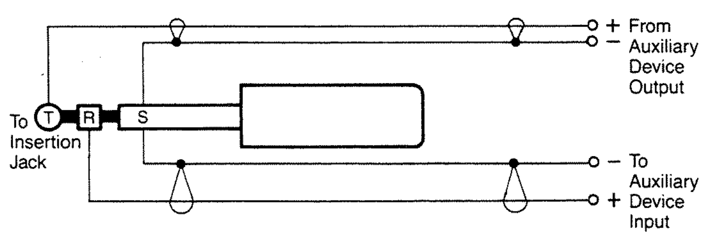
T/R/S Phone Plug to XLR Connector (for Pin 2 "high" systems)

Use this wiring to connect a balanced, line-level device to the Hi-Z In jack.

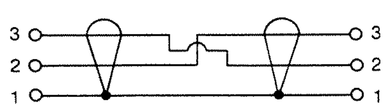


(for pin 3 "high" systems, reverse wiring of pins 2 and 3)

Wiring for 4200 Mixer Insertion Connector



XLR Polarity Reversal Adapter
Use Switchcraft Part # S3FM



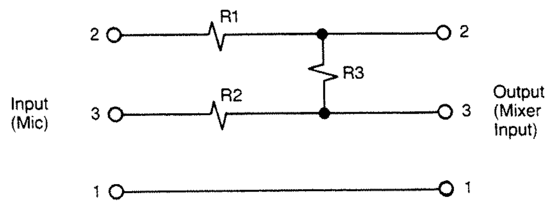
Pads and Transformers

A "pad" is a resistor circuit that reduces the output level from a source device to make it "level compatible" with a load device. For example, a pad could be used to connect the output of a +4dB limiter to the -10dB Insertion In jack on your 4000 Mixer.

The pads shown here are of two types, balanced and unbalanced. The balanced pads are meant for balanced microphones or for low-source-impedance balanced line-level devices. In most cases, however, you will not need a microphone pad with your 4000 Mixer since proper adjustment of the Trim control achieves the necessary level compatibility. The unbalanced pads are meant for low source-impedance, line-level devices and should not be used with high-impedance microphones.

The balanced pads may be constructed inside a Switchcraft Model S3FM using ¼-watt or ½-watt resistors. The unbalanced pads may be constructed in a small metal parts box or, using ¼-watt resistors and a lot of care, they may be assembled inside a ¼" phone plug (make sure to mark the cable that has such a pad/plug attached).

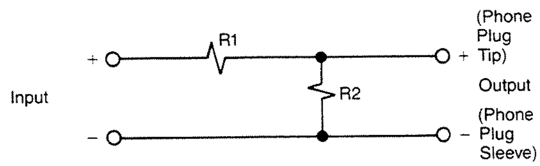
Pads



A "U" Pad for use with Lo-Z microphones (or low-impedance, balanced, line-level devices). Construct with ¼-watt resistors in Switchcraft Part #S3FM.

dB Attenuation	R1	R2	R3
10 dB	1500Ω	1500Ω	1500Ω
16 dB	1500Ω	1500Ω	560Ω
20 dB	1500Ω	1500Ω	330Ω

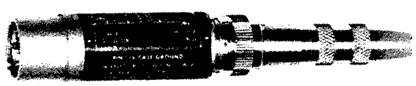
(Values assume 4200 Mixer input impedance of 8k Ω. However, pads will work well with most Lo-Z mics and mixers.)



An "L" Pad for use with unbalanced, line-level outputs. Use 16 dB pad between Pre Amp Out and Power Amp In jacks to allow "0" VU Position to be "nominal" (see "The VU Meters"). Construct in a metal "project box". Use ¼-watt or ½-watt resistors.

dB Attenuation	R1	R2
10 dB	10kΩ	5600Ω
16 dB	10kΩ	2200Ω
20 dB	10kΩ	1000Ω

(Values assume 4200 Mixer Power Amplifier Input Impedance of 16k-Ω.)



Hi-Z Mic to Lo-Z Mic In-Line Transformer

Transformers are devices that can sometimes be used to connect devices with unlike impedances and levels. For example, a "Hi-Z to Lo-Z" microphone transformer converts the high (voltage) level and high-impedance of a high impedance microphone to the low (voltage) level and low-impedance of a low-impedance microphone. Other transformers can convert high-impedance, high-line-level devices to low-impedance, low-line-level devices.

Because a transformer is not an "active" device, however, it cannot *amplify*. Thus, when you convert from high-impedance to low-impedance, for example, you also convert from high-level to low-level. A transformer cannot convert impedance without also converting level (usually in the direction we don't want). Transformers are also level-sensitive. That is, a microphone hi-z to lo-z transformer cannot be used for line-level impedance conversion (it would distort). Neither can a line-level transformer be used for microphone-level conversions (it would also distort, although in a different manner). Thus, when selecting transformers, you must define your needs in terms of both the impedance ratio desired and the level of the devices that will be connected to the transformer.

One valuable use of a microphone hi-z to lo-z transformer is to convert a high-impedance microphone to low-impedance to allow it to be used with longer cable lengths. A high-impedance to low-impedance line-level transformer could be used to allow a high-impedance, line-level device to be used with longer cable lengths, too.

For more specific information on pads and transformers, consult your Fender dealer or see: "Sound System Engineering" by Don and Carolyn Davis or "The Audio Cyclopedia" by Howard M. Tremaine; both published by Howard W. Sams.

Grounding and Shielding

Caution

In any audio system installation, governmental and insurance underwriters' electrical codes must be observed. These codes are based on safety, and may vary in different localities; in all cases, local codes take precedence over any suggestions contained in this manual. CBS Fender Musical Instruments shall not be liable for incidental or consequential damages, including injury to persons or property, resulting from improper, unsafe or illegal installation of a Fender 4000 Series Mixer or of any related equipment; neither shall CBS Fender Musical Instruments be liable for any such damages arising from defects or damage resulting from accident, neglect, misuse, modification, mistreatment, tampering or any act of nature.

Note

The AC Power discussions in this section apply to the U.S.A. only. The general discussions of grounding and shielding, however, should be applicable audio systems used in any location. Always obey local fire and electrical safety regulations wherever you are using your audio system.

Why We Must Consider Grounding and Shielding

There are two primary reasons for careful grounding and shielding in an audio system. The first reason is safety. A poorly grounded system, especially outdoors, may be a shock hazard. The second reason is to reduce pickup of external noise. That external noise expresses itself in the form of hums and buzzes and other noises including radio station pickup. The following discussions of electrical safety and grounding assume you are using your 4000 Mixer in the USA. The general concepts, however, should apply well to safety considerations anywhere in the world.

Grounding for Safety

The third (round) prong on the AC cable of your 4000 Mixer is the AC safety ground. It is connected to the metal chassis of your Mixer. When you plug this cable into a properly wired AC receptacle, the chassis of your Mixer is connected to the AC ground through the third prong of the AC receptacle.

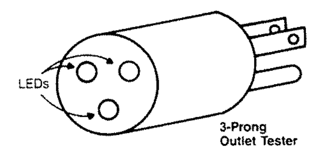
This is the ideal situation from a safety viewpoint. Under these conditions, there is almost no combination of events that could cause a shock hazard from your 4000 Mixer *by itself*. Unfortunately (from a safety viewpoint, that is), your Mixer will never be used by itself; there are always other pieces of equipment involved and most of the time, these are also AC powered. In addition, you may be forced to deal with one or both of the following: 1) older audio equipment with 2-wire AC plugs and "ground" or "hum" switches; 2) Older, 2-wire AC receptacles or improperly wired 3-wire AC receptacles.

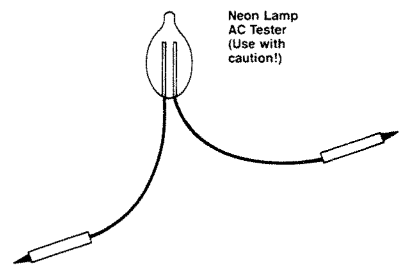
Dealing With Improperly Wired AC Receptacles

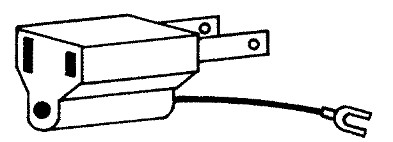
No matter whether you consider yourself a technician or not, there are two items you ought to have with you every time you set up your system in a new facility. One of these is a three-prong outlet tester, the second is a neon lamp AC voltage tester. You should be able to buy these inexpensive items at most hardware or electrical stores (try a lighting store if you can't find them elsewhere).

The three-prong outlet tester will tell you if the outlet is properly wired. An improperly wired outlet may have its two AC wires reversed ("polarity reversal") or it may have a disconnected ground. *Any* fault in the wiring of the AC receptacle is potentially hazardous and thus, *the best, and perhaps only safe way to deal with an improperly wired AC receptacle is to simply refuse to use it until it has been repaired.*

AC Outlet Testers and Adapters







Two-Prong to Three-Prong AC Adapter (wire on two-prong end ***must*** be connected to grounded screw).

Dealing With Two-Wire AC Receptacles

The problem with two-wire AC receptacles is they don't have that important third ground prong. Thus, to use one of these two-wire receptacles you have to "adapt" it to the three-wire AC plug on your 4000 Mixer with a two-wire to three-wire AC adapter. *Properly used*, these adapters maintain a safe ground for your Mixer as well as a three-wire receptacle.

To make this two-wire adapter work properly, you *must* connect the loose wire on the two-wire end to a *grounded* screw on the two-wire AC receptacle. How do you know whether or not this screw is grounded? Easy! First, connect the loose wire on the adapter to the screw on the two-wire receptacle; then plug the two-wire adapter into the two-wire receptacle. Now, plug your three-wire AC outlet tester into the adapter. If the screw is grounded, your AC outlet tester will tell you. (Most three-wire AC outlet testers either have a "good" light or else they don't light at all on a good receptacle.) If the screw is not grounded, the outlet tester will so indicate. In this case, you must connect the loose wire from the adapter to some other grounded screw in order to maintain a safe ground for your Mixer.

If the outlet tester shows a good ground but reversed polarity on your two-wire to three-wire adapter, simply reverse the adapter in the receptacle.

Dealing With Older Two-Wire Audio Equipment

Some newer equipment may come with a two-wire AC cable. This newer equipment may be as safe as if it had a three-wire AC cable. A good example of such a piece of (non-audio) equipment with a two-wire AC cable is one of the so-called "double-insulated" power tools (drills, saws, etc). One way to judge the safety of a piece of audio equipment with a two-wire AC cable is to look for a "UL" (Underwriter's Laboratories) sticker. Other reliable safety agencies are the City of Los Angeles and the CSA (a Canadian safety organization). Seals from any of these three organizations are reliable indicators of the fire and shock safety of a piece of equipment.

It's the older, two-wire audio equipment, however that can be potentially hazardous. The details of how a shock hazard can develop are complex, but dealing with this problem is straight-forward. The shock hazard, if there is one, will probably develop between the chassis of an older, two-wire device like a guitar amplifier and the chassis of a microphone.

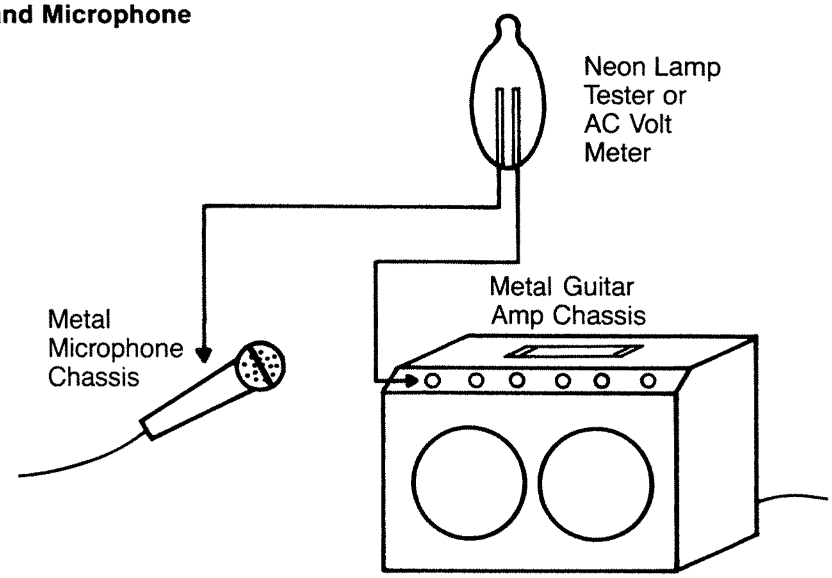
The chassis of the microphone is connected to the chassis of your 4000 Mixer through the shield of the connecting cable. Thus, if your 4000 Mixer is properly grounded, the chassis of the microphone is properly grounded, too, and neither the microphone or the Mixer will present any safety hazard. The guitar amp (or other two-wire equipment), however, is, potentially, *not* properly grounded. That means that a hazardous AC voltage could be present on the chassis of the guitar amp *or on the strings of a guitar* which are connected to the chassis of the amplifier through the shield of the guitar cord.

How do you discover this type of hazard? Easy! Use the neon lamp AC tester. Place one lead on the chassis of the guitar amp and the other lead on the chassis of the microphone. (Don't touch both the chassis of the microphone and the chassis of the guitar amp at the same time with your hands.) If the lamp lights, you've got problems! You may be able to solve the problem by reversing the guitar amp's AC plug in the AC receptacle (pull it out, twist it ½ turn and put it back in). Also, try reversing the position of the "hum" switch if the guitar amplifier has one. If the problem doesn't go away, try plugging the guitar amp into a different AC receptacle (on the same building AC circuit as your Mixer, if possible).

One problem with this approach is that the neon tester will only show the presence of voltages greater than about 90 volts AC. While the majority of hazardous voltages are 90 volts or greater, some may be below this voltage. You can test for these with a low-cost volt-ohm meter which you can purchase at most electronic hobby shops (your Fender dealer may carry these). The ohms function on this meter will be useful in checking out suspected faulty audio cables, too.

Another thing you can do with your volt-ohm meter is to actually measure the AC receptacle voltage. Especially in an unfamiliar facility, this is an excellent suggestion. Voltages that are too high or too low for your equipment could cause improper operation, or even damage your equipment and too-high voltages could also pose a shock hazard. Most audio equipment will work fine on an AC outlet with voltages as low as about 105 volts AC and as high

Testing for Potential Shock Hazard Between Guitar Amp and Microphone



Caution: Do not touch the microphone and guitar amp with your hands until you are certain no shock hazard exists!

as about 125 volts AC. Check the specifications for the equipment you are using.

Grounding for Safety Outdoors

The most common safety problems outdoors are improperly wired portable AC cables and wet ground or stages (and, of course, rain). Check your wiring carefully, the same as you would indoors. Consider canceling a performance if rain begins. If you must perform on wet ground or in the rain, the best way to avoid shock hazards to the performers is to use wireless microphones and wireless guitar transmitters. These same outdoor problems, of course, can develop indoors on a damp floor, so watch out!

Grounding to Reduce External Noise Pickup

One myth about grounding is that you *must* ground your equipment to avoid noise pickup. Anyone who owns a portable cassette machine knows that that simply isn't true. The *primary* reason we ground our audio equipment is for *safety*. An important secondary reason is that, with AC powered equipment, under some conditions, proper grounding *can* help reduce external noise pickup.

The third reason that we must pay attention to grounding is that, while proper grounding won't always reduce external noise pickup, *poor* grounding can unquestionably *increase* external noise pickup!

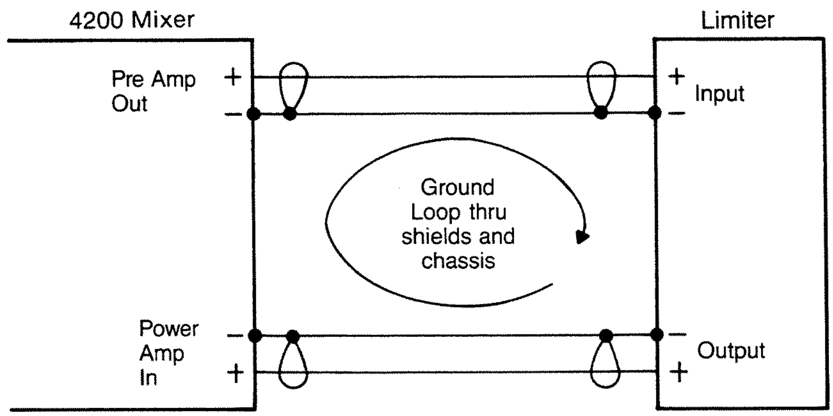
Poor grounding practice usually results in "ground loops" and avoiding these ground loops is the second most important part of proper grounding (the first most important part, of course, is maintaining the safety ground).

A couple of examples will help explain what a ground loop is and how to avoid them. In Example 1, the loop is between two audio cables that connect a limiter to a 4000 Mixer. This is an example of an unavoidable ground loop. The best way to deal with this type of ground loop is to physically place the two audio cables as close together as possible (lace or tape them together if your setup will allow it). This reduces the *area* enclosed by the loop which will significantly reduce the pickup of external noise. This same situation could result if you were connecting the Left and Right Program Out jacks from your 4000 Mixer to the left and right inputs of an unbalanced input power amplifier. Again, the best way to reduce the pickup of external noise from this unavoidable ground loop is to run the two cables very close to each other.

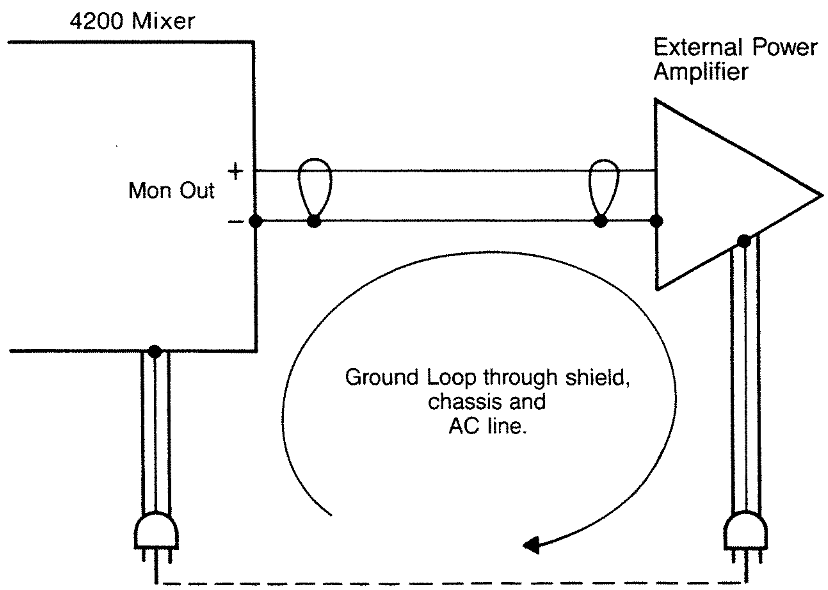
In the second example, the ground loop occurs between a 4000 Mixer and an external monitor power amplifier. Even though there is only one audio cable connecting the two devices, a second ground connection, through the

Ground Loops

Example 1: Unavoidable Ground Loop (keep cables close together).



Example 2: Ground Loop Through AC Line.



AC cables of the devices, makes the return connection and forms a ground loop. The only way to break this ground loop is to lift the AC ground on the power amplifier with a two-wire to three-wire AC adapter (leaving the loose wire on the adapter unconnected). Because this practice is in conflict with the AC safety ground, here are two rules to minimize the safety conflict: 1) Don't lift the safety ground on any piece of equipment unless it demonstrably reduces noise pickup. 2) Never defeat the AC safety ground on your 4000 Mixer by using a two-wire to three-wire AC adapter in this manner. The reason for this second rule is that your Mixer chassis is connected to the chassis of all your microphones and *only by properly grounding the Mixer can you be assured of a safe ground on your microphones!* Always maintain at least this one ground for safety!

It's worth noting that, by using balanced connections between two pieces of audio equipment, you can "lift" (disconnect) the shield at the "sending" end of the audio cable to interrupt the type of ground loop discussed in Example 2. Since, in a balanced line, the shield does not carry audio signal, you can disconnect the shield at one end without interrupting the audio signal (and without disrupting the effectiveness of the shield). Unfortunately, this is not a very practical solution to the problem in a *portable* audio system because it would require special cables which have the shield disconnected on one end.

Using Proper Shielding to Reduce Noise Pickup

Proper *grounding* helps prevent pickup of noise that is transmitted *magnetically*. Magnetically transmitted noise most often comes from motors or, more commonly in audio, from large AC power transformers (either building transformers or the power transformers in a power amplifier or other piece of audio equipment). Proper *shielding*, on the other hand, helps prevent pickup of noise that is transmitted *capacitively*. Capacitively transmitted noise may be in the form of radio waves from a radio station or CB radio or it may be in the form of "static" from certain types of motors or from lighting dimmers. (Noise from lighting dimmers may also come through the AC lines, as discussed below.)

Fortunately, proper shielding is pretty easy. Just make sure that you are using high-quality shielded cables on all microphones and on all line-level equipment. Some very low cost audio cables including guitar cables have poor quality shields. Watch for these potential sources of noise pickup.

It is seldom necessary to use shielded cable for your loudspeakers since they operate at very high level. The noise picked up by a loudspeaker cable is actually as high level as the noise picked up by a microphone cable but, because the loudspeaker operates at a much higher level than the microphone, the *signal to noise ratio* is vastly better and the noise is seldom a problem.

Reducing Noise Pickup From AC Lines

Some types of noise, notably noise from lighting dimmers, gets into your audio equipment from the AC power cable. There are two ways to reduce this problem. 1) Install filters *on the dimmer circuits* (filters at your audio equipment won't help as much and probably will cost a lot more). 2) Make sure the dimmer circuits are properly loaded. That is, if the dimmers are rated for 1500-watt loads, make sure they have 1500-watts worth of lighting connected to them. (Or add a suitable "dummy load" to simulate a full rated load on the dimmer.) The reason for this is that the noise filters (if there are any) will only work properly when the dimmer is loaded properly (this is an example of impedance "matching"). 3) Make sure the lighting circuits are properly grounded (improper grounding can increase noise levels at the source as well as at your audio equipment). 4) Use a different AC circuit (you know its a different circuit if it uses a different house fuse or circuit breaker).

Finding the Source of a Noise Problem

This can be the hardest part. A "buzzing" noise in your system may be attributable to a set of lighting dimmers in the house, but *you must find out how the noise gets transmitted from those dimmers into your system before you can cure the problem*. Is the noise transmitted magnetically? (If so, eliminating ground loops in your system should help). Is it transmitted capacitively? (If so, look for poor quality shields or faulty connectors.) Is it transmitted through the AC power lines? (Install filters at the source of the problem or move your audio equipment to a different AC circuit.) In the end, you may have to try all of these methods to solve a given problem but if your system is carefully grounded and properly shielded in the first place, you'll be less likely to experience a noise pickup problem. Here then are some final tips on noise reduction.

More Tips on Reducing Noise Pickup

1) Rack mount your equipment. Rack mounting, especially when the rack mount rails are made of metal, connects together the chassis of all your equipment into a unitized shield. Perhaps more important, rack mounting allows you to use shorter connecting cables and to keep them closer together. When rack mounting large power amplifiers, however, do not place sensitive, low-level equipment right next to them in the rack. The power transformer in a large power amplifier can produce a large alternating magnetic field that can "induce" hum in low-level equipment.

2) Keep your cables short. Rack mounting can help here. So can simple "neatness."

3) Keep cables of the same type close together. By "the same type," we mean cables that carry the same signal level (like line level signals). Especially when they form an unavoidable ground loop, like the one in Example 1, keeping your cables close together will help reduce noise pickup.

4) **Keep cables of different types as far apart as possible.** That means keep your microphone cables away from loudspeaker cables. And keep all audio cables away from the AC power cables. On long cable runs, keep line-level cables and microphone cables separated. It's a common, but risky, procedure to run microphones through a "snake" (a multi-microphone cable) to a mixer and then run the outputs from the mixer back to the power amplifier through the same snake. This mixing of levels, in a long cable run (greater than about 25 feet could be a problem) can cause a form of electronic feedback that could cause harmful oscillations in your mixer.

5) **Keep your wiring "neat."** Carefully made cables, of the proper length (not too long) and carefully laid out on a stage or in an installation, are probably the best way of all to reduce external noise pickup.

Dealing With Feedback, Hum, Hiss and Other Noises

Feedback

Feedback is the "howling" sound caused by bringing a microphone too close to a loudspeaker. As simple as it seems, however, feedback is actually a complex phenomena and solving feedback problems involves working with a number of variables.

Equalizers are commonly used to control feedback. They aren't necessarily the best way, as discussed below. When you use an equalizer, like the Graphic Equalizers on your 4000 Mixer, to help control feedback you pay a price. That price is an uneven frequency response (and the resulting un-natural sound quality) caused by using the Graphic Equalizer to control feedback (rather than to make the system sound good). This price may be acceptable on a monitor system, however since the performers, not the audience, are the only ones who hear it.

If you are using an external graphic equalizer on the Monitor 1 or 2 Output, you can pre-set it for minimum feedback by starting out with all the graphic equalizer controls at their mid position and then *carefully* turning up the Monitor (1 or 2) fader until you get just the beginning of feedback. We urge care in this operation because high-level, sustained feedback can damage loudspeakers (and ears!). Then, using your musician's ears, or a real-time analyzer (or even a frequency counter) determine the approximate frequency of the feedback and pull down the graphic equalizer control nearest the feedback frequency. Pull it down just enough to quiet the feedback and then increase the Monitor fader a little more. Chances are, after you do this two or three times, a different feedback frequency will appear. Work on two or three of these feedback frequencies at the most. Trying to cure more than that will result in a very un-natural sound quality (because of the settings of the graphic equalizer) and also will result in diminishing returns.

A "narrow-band" equalizer or "notch filter" may work better at feedback reduction than the relatively wide-band filters in your 4000 Mixer's Graphic Equalizers (which, unlike narrow-band types, were designed primarily to enhance the audio sound quality). You will, however, always reach a point of diminishing returns in this process and remember that *almost any change in the stage setup can change the entire feedback situation in your system!*

Acoustic Solutions

Feedback is an acoustic problem. Thus, the primary way to deal with feedback is to trace its source and attempt to stop it by acoustic means (this may be as simple as altering your stage setup). First, find out which loudspeaker and which microphone are causing the feedback. You can do this by simply turning off all but one microphone and all but one loudspeaker and then trying the next microphone and so on. When you find out which microphone and loudspeaker are the culprits, try moving one or the other (or several if it seems like several mics are involved). Sometimes a small movement can cure a feedback since feedback can be caused by reflections from floors, walls, or even table-tops as well as direct sound from a loudspeaker to a microphone. You may also try a different microphone. As discussed in the section entitled "Choosing and Using Microphones," a cardoid microphone may help lower your system's feedback potential. Teaching the performers to work the mic close to their mouths or instruments can help immeasurably in controlling feedback because this technique allows you to reduce fader levels (this also reduces "bleed" from one instrument to more than one microphone). Only after you have tried all of these cures should you resort to using an equalizer to control feedback.

Hum

Hum and buzzing may be caused by internal problems in a piece of audio equipment or they may be caused by external noise sources. If the problem is internal to a piece of audio equipment the solution is simple: get it fixed. If the problem is pickup of external noise, the solution may not be so simple. Read the sections entitled "Impedance and Level Watching," "Grounding and Shielding" and "Cable and Connectors" for some suggestions.

Hiss

Hiss is random electronic noise that is generated by every piece of audio equipment. A certain amount of electronic noise is in-escapable in any piece of audio equipment. In a high-quality audio device like your 4000 Mixer, this hiss level is extremely low. Some hiss may be generated, however, when we connect two pieces of audio equipment together (see "Impedance and Level Watching").

You actually have a great deal of control over this process. Simply adjusting the Trim control properly for each Input Channel will help a lot. That's because, properly adjusted, the Trim control boosts the level of an incoming signal as high as possible without causing clipping of the Input Channel. This higher level signal helps bury the hiss noise. Another way you can reduce hiss noise is to simply plug devices into the appropriate input. For example, if you plugged a hi-Z microphone into the Aux In jack, and turned up the Aux In Program control far enough, you could probably get an audible sound. However, you would also get a lot of hiss because the hi-Z microphone just doesn't have enough output level to work well in the Aux In jack. Plug the hi-Z microphone back into its proper input, the Hi-Z jack on the Input Channel, and things will work right again.

One common source of hiss is a tape machine, in which case the best solution is to use a tape noise reduction device. Another source of hiss is a tuner, in which case you may get an improvement from a better tuner antenna.

Other Noises

One, unfortunately very common, source of noise is faulty audio cables or connectors. A bad cable can make a loudspeaker sound like it is "blown" or it can make a microphone sound like a kazoo. In fact, bad cables are so common that you should probably suspect them when just about any problem occurs in your system. If you suspect a bad cable, try shaking it to see if the problem gets worse (or better). Also see the section entitled "Troubleshooting."

Troubleshooting a Sound System

Repairing a sound system may require the skills of a trained technician. *Troubleshooting*, that is, *finding the problem* is something almost anyone can do if they:
1) Know the block diagram of their system. 2) Understand what each component in the system is *supposed* to do. 3) Know where to look for common trouble spots.

Know Your Block Diagram

A sound system block diagram tells you how the various components in the system are connected to each other and what happens to a signal as it flows through the system. Reading a block diagram is relatively easy. See the section entitled "Understanding Block Diagrams" for a review. Because the block diagram shows the way the sound system operates, it is extremely useful in the troubleshooting process.

Know What Each Component is *Supposed* To Do

As obvious as it may sound, you can't tell whether a component is working properly or not unless you know what it's supposed to do in the first place! Thus, it's a good idea to keep instruction manuals on all components handy. Some "repairs" are as simple as repositioning a control knob or throwing a switch that someone has inadvertently changed.

Know Common Trouble Spots

Cables and connectors are by far the most common sources of problems in audio systems. This is the best reason to keep lots of spares, especially of cables that are moved around a lot, like microphone cables.

Other common trouble spots are fuses and circuit breakers, switches and controls that are in the wrong positions and problems with house AC power.

Logical Troubleshooting

The process of troubleshooting involves logical thought and methodical tracking down of a problem by elimination.

Logical thought processes come into play when a problem first occurs. If a single microphone goes suddenly dead, your logic tells you that the power amplifier probably isn't at fault. If, on the other hand, your whole system is suddenly quiet, the power amplifier might be at fault, but it's not likely that all of your microphones have failed at once.

A methodical elimination process can track down the source of most problems very quickly. The idea is to find out what *component* (microphone, cable, mixer, amplifier, loudspeaker) is causing the problem and to replace or repair it. During a performance, of course, replacing a faulty component is the most likely cure since a repair might take up too much time.

Your mixer is a good place to begin the trouble-shooting process because it has the controls for the entire system. If you hear a noise in the system, for example, look at your VU Meters, and Signal and Peak LEDs. This alone, may tell you that the noise problem is coming from one microphone. Pull down the fader for that Input Channel. If the noise goes away, check out the microphone, or more likely, the microphone cable.

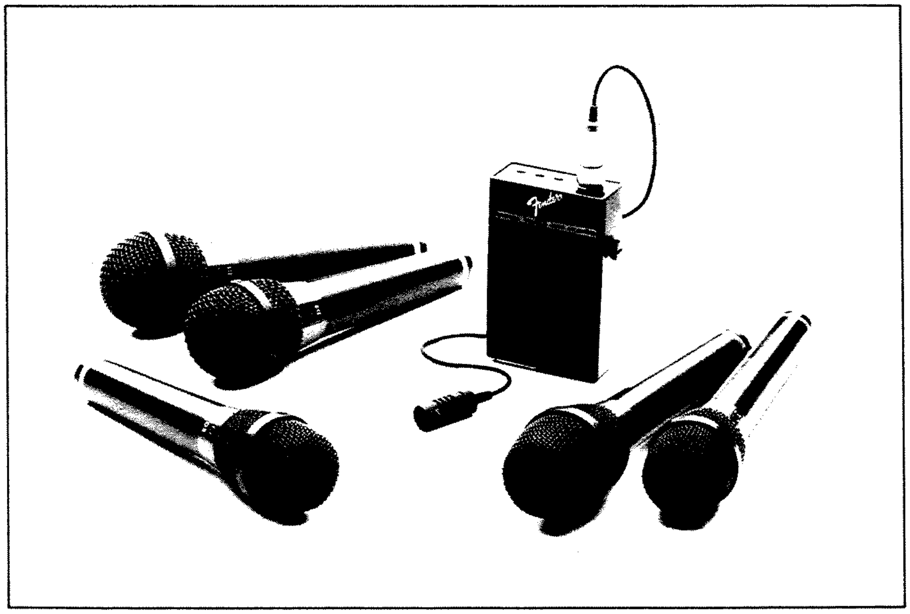
If your entire system suddenly goes dead, again, check out your VU Meters and LEDs. If they are still active, then you know your system is working at least through the Mixer. Thus some component further along in the system must be the culprit. Think through your block diagram at this point to find the next suspect component. (One "component" that's often a problem is the house AC power!)

When possible, patch around suspect components. For example, a limiter can be completely removed from the system and the system will still operate. Thus, if you suspect a limiter, use a patch cable to bypass the limiter (or, if it is connected to the

Insertion Out, Insertion In jacks, just remove it from those jacks). If the bypass operation causes the system to begin operating again, the limiter was at fault.

When the suspect component is necessary to the operation of the system, try to replace it with some other equivalent component. If you suspect a bad loudspeaker, for example, try switching your left and right loudspeakers or using a monitor loudspeaker temporarily in place of a main system loudspeaker. If you suspect your mixer, try running a tape machine directly into your power amplifier to make sure that that portion of the system is still working.

"Repairs" may be as easy as replacing a bad cable or patching around a bad limiter or removing a faulty effects device from the system. But, before you can repair a system, you must find the problem and that is what troubleshooting is all about.



Fender Professional Microphones

Choosing and Using Microphones

Types of Microphones

There are three primary ways to classify microphones: by impedance, by element type and by directional pickup pattern.

Pro audio microphones are either "low-impedance (lo-Z)" or "high-impedance (hi-Z)." A low-impedance microphone will have a "source" impedance (see "Impedance and Level Watching") of anywhere from about 50-ohms to as high as 600-ohms but 150-ohms to 250-ohms is most common. A high-impedance microphone will have a source impedance ranging anywhere from 1000-ohms to as high as 10,000-ohms (10k-ohms) or greater. Because of the problem of high-frequency loss with hi-Z microphones using long cables, most pro audio microphones are lo-Z. In addition, because of the superior noise rejection of a balanced line, most pro audio microphones are balanced.

There are many different ways to convert sound energy to electrical energy and the portion of a microphone that accomplishes this task is called the "element." For pro audio, most microphones use either a "dynamic" or a "condenser" element. Dynamic microphones use a moving coil, attached to the diaphragm and immersed in a magnetic field, to convert sound to electrical energy. Condenser microphones use a pair of electrically polarized plates, one moving, one fixed, to convert sound into electrical energy. Condenser microphones require some kind of external power source, either a battery or "phantom power" from a mixer (your 4000 Mixer supplies phantom power).

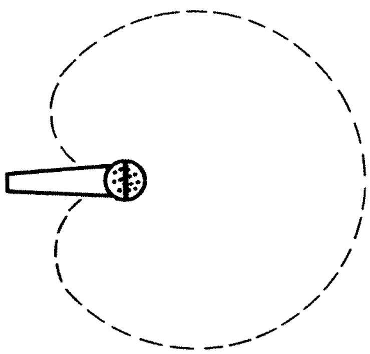
The directional pickup patterns of microphones fall into two general catagories: omni-directional and directional. Ideally the frequency response of an omni-directional microphone is very uniform at any angle away from the front of the microphone. The pickup patterns of directional microphones vary with their design. The most common pro audio directional microphone has a "cardioid" pickup pattern (somewhat like a heart shape rotated into 3 dimensions). Omni-directional and directional microphones are available with either dynamic or condenser elements and in both lo-Z and hi-Z impedances.

Choosing Microphones for Pro Audio

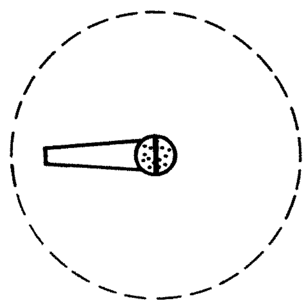
Dynamic microphones are noted for their ruggedness, their reliability and their ability to handle very high sound pressure levels without distortion. Condenser microphones are somewhat less rugged and may distort when presented with extremely high SPL levels. Newer condenser microphones are, however, are more rugged and will handle higher SPL levels than earlier models. Thus, your choice can be based on pickup pattern and subjective sound quality.

The cardioid pickup pattern is valuable in pro audio because a cardioid microphone rejects sound that comes from behind the microphone (like audience noise). Because of this rear-rejection, a cardioid microphone is less prone to feedback (howling)

Microphone Pickup Patterns



Cardioid pickup pattern microphones attenuate sound from the rear.



Omni-directional pickup pattern microphones pick up sound uniformly from all directions.

caused by stage monitor loudspeakers feeding back into the rear of stage microphones. Cardioid microphones, on the other hand, may have non-uniform sound quality as you move off-axis. That is, if you talk into the side of a cardioid microphone, the sound level will decrease, and the sound quality will change. This is undesirable, especially for microphones that need to pick up more than one source (like a mic used for a pair of vocalists, or a choir mic in a house of worship). This effect varies greatly among brands and models of microphones, so don't automatically reject a cardioid mic for a multi-source application, just check it out carefully.

The sound quality of a cardioid pickup microphone may also vary with distance away from the microphone. This is known as "proximity effect" and it results in more bass response (a warmer sound) as a singer moves closer to the microphone. In some cases, this can be a distinct advantage to a singer who knows how to use this effect to their advantage. For someone who doesn't understand microphones, however, proximity effect could cause unwanted "boominess" in the sound. (Proximity effect is also discussed in the section entitled "The Input Channel EQ Controls.")

Another aspect of a microphone's sound quality is known as "presence." A microphone with a lot of "presence" probably has a slight boost in the mid to high frequencies to improve the clarity of the sound. Presence can enhance the sound quality of many vocalists and can increase intelligibility in speech-only systems.

Omni-directional microphones have much more uniform sound quality as you move around the mic than do cardioid microphones. They may, however, be more prone to feedback, and cannot reject noises coming from the rear of the microphone. One way to overcome these problems is to "work" the mic very closely. That is, the performer should sing or play their instrument very close to the microphone at all times. That way, the mixer operator can reduce the level of the fader on that Input Channel and that helps reduce feedback and noise pickup. The sound quality of an omni-directional microphone also does not vary appreciably as you move away from the microphone. This can be an advantage. Again, however, the best way to judge a microphone's sound quality is to try it out yourself.

Two excellent references for more information on microphones are "The Microphone Handbook" by John M. Eargle, published by ELAR Publishing Company of Plainview and "Microphones," a short text available from Gotham Audio Corp, 741 Washington St., New York, NY 10014.

Using Direct Boxes and Instrument Pickups

A direct box is a device that splits the signal from an electric guitar, or any instrument with a pickup, and sends it to both the normal instrument amplifier and a mixer. The direct box usually includes a transformer, or, in an "active" direct box, a preamplifier, to convert the high-impedance, unbalanced instrument pickup to a balanced low-impedance signal. Since this signal is about the same level as a low-impedance microphone, you can plug the output from most direct boxes into the Lo-Z Input on your 4000 Mixer. (The signal sent to the instrument amplifier is unchanged from the normal signal.)

Other instrument pickups include their own preamplifiers. The output of these preamplifiers may be low line-level which means that you can connect them to the Hi-Z Inputs on your 4000 Mixer.

Direct boxes, instrument pickups and their associated preamplifiers are sometimes called "microphone substitution devices" because they are sources whose output level is nearly the same as a microphone.

Other Sources

Most other sources, from tape machines to tuners, can be connected to either the Hi-Z Inputs or the Aux or Direct Inputs on your 4000 Mixer. Phonograph turntables, of course, require an "RIAA" phono preamplifier. Several manufactures make small, separate phono preamplifiers which usually have low line-level outputs. Or, you can use the "preamp outputs," "tape outputs" or "auxiliary outputs" on your hi-fi preamplifer, receiver or integrated amplifier.

Using Special Effects Devices

Special effects devices may be connected to either the Effects mix or the Input Channel Insertion jack. These connections are discussed earlier in this manual. The most common effects include reverberation, instrument effects (fuzz, phasing, flanging, etc.), delay lines, expanders, and the various noise sources used in live theatre.

If the device is an instrumental special effect, it may be used primarily on one instrument. In that case, you would use it through the Insertion jack on one of the Input Channels of your 4000 Mixer (or connect it directly to the instrument itself). If the device is meant to enhance a voice, then you would, again, probably use it on a single Input Channel via the Insertion jack.

Some effects, however, like reverberation, will probably be used on all instruments and voices. Use these through the Effects mix.

Using Limiters and Compressors

Actually, limiters and compressors are two versions of the same device. In fact, many such devices are called "compressor/limiters." An "expander" is a similar device. All three devices monitor the signal level and change it (like an automatic volume control) in some pre-determined way.

A compressor reduces the level of high-level signals and increases the level of low-level signals. In other words, it reduces the "dynamic range" of the signal. Compressors are used by background music suppliers to keep the level of their music nearly constant. This allows the music in a department store, for example, to always be loud enough to hear (above crowd noise) but never so loud as to be annoying. You could use a compressor for the same purpose in mixing a quiet group for a hotel lounge. Compressors are also useful for tape recording. The dynamic range of live music must be reduced to fit the dynamic range capabilities of a tape recorder, and a compressor can be used for this purpose. Special noise reduction devices, like those made by Dolby and dbx and others are probably a better choice for this purpose, however.

A limiter reduces the level of high-level signals but does not affect low-level signals. While compressors are operating most of the time, a limiter only operates above a fixed "threshold." That is, the limiter begins to reduce the signal level only when it exceeds some preset level. Limiters are used by radio stations to avoid over-driving their transmitters. Limiters are used extensively in pro audio to keep the audio signal from overdriving a power amplifier (overdriving a power amplifier can cause clipping distortion and can even cause damage to the power amplifier and loudspeakers). External limiters are probably the best way of protecting your external power amplifiers and loudspeakers from damage and are an excellent way to help you avoid clipping distortion.

An expander actually increases the level of high-level signals and reduces the level of low-level signals. Thus, an expander *increases* the dynamic range of a signal. Expanders in pro audio are used primarily for special effects. An expander, used improperly, could present a danger to your system since it could increase high-level peaks to the point of clipping.

Equalization

What Do We Mean By "Equalization?"

The term equalization originally meant "to equalize the frequency response of a sound system to match a room." The term, equalization, however, now applies to just about any process that changes the frequency response of a signal. The Input Channel Equalization controls, for example, would probably be called "tone controls" on a hi-fi product. In pro audio, however, they are called "Equalization" controls.

A "graphic" equalizer is so called because the position of its sliders form a curve, like a *graph* of the frequency response.

Using Equalizers

As we discussed in "The Input Channel Equalization Controls," the Input Channel Equalization controls are used to change the tonal character of an individual voice or instrument. The Program and Monitor Graphic Equalizers (or an external graphic equalizer) are used to affect the frequency response of an entire mix to compensate for room acoustics, for example.

Elaborate test equipment, including "pink noise generators" and "real-time analyzers" is available to aid in the process of room equalization. The instruction manuals that come with a real-time analyzer usually explain the process of room equalization or you can purchase one of several books on the subject including "Sound System Engineering" by Don and Carolyn Davis, published by Howard W. Sams.

So-called "narrow-band" equalizers or "notch filters" are sometimes used to help stop feedback (howling) in a system or the ringing that comes just before feedback. Using equalizers to help control feedback is covered in more detail in the section entitled "Dealing with Feedback, Hum, Hiss and other Noises."

An equalizer, of any type, is a powerful tool. Equalization can, indeed, help compensate for undesirable room acoustics. An equalizer can, within limits, be used to compensate for poor loudspeaker frequency response. Graphic equalizers or notch filters can be used, again within limits, to control feedback. And, of course, tone controls like the Input Channel Equalization controls on your 4000 Mixer, can be used to enhance an individual instrument or voice.

(So-Called) "Room Equalization"

At one time, it was thought that an equalizer could actually reduce reverberation in a room (it can't). Equalizers were also thought of as the answer to controlling feedback (they can help but are not a cure-all). We now know that room reverberation can only be affected by acoustic treatment and that feedback has complex causes that are not all related to system frequency response.

You can, however, use the Program Graphic Equalizers on your 4000 Mixer to *help* compensate for poor room acoustics. For example, most highly reverberant rooms have their worst reverberation in the lower frequencies. Reducing the level of the lower frequencies on your Graphic Equalizer may help the system sound less "boomy." Try to avoid high SPL levels in a reverberant room, too. In many rooms, lowering the overall SPL will help reduce the apparent reverberation.

At the opposite end of the reverberation problem, some hotel lounges have so much carpet, acoustic tile and padded furniture that they sound extremely "dead." Adding a little high-frequency boost with your 4000 Mixer's Graphic Equalizer can bring some life back into the sound (try an external reverberation device, too!).

Equalizers and Loudspeakers

Most non-biamplified loudspeakers have some amount of equalization designed into their crossover networks. The purpose of this equalization is to help smooth their frequency response. These loudspeakers seldom require additional equalization to improve their frequency response. The same applies to those loudspeakers which come with an external, active equalizer meant to be installed between a mixer and power amplifier. The equalization in that active equalizer is all that you should need.

Adding additional equalization to these loudspeakers to, for example, increase their bass response, may work very well at low power levels. At higher power levels, however, this kind of additional equalization may result in amplifier clipping and even loudspeaker damage. Thus, if your loudspeaker system always seems to need additional low-frequency (boost) equalization, consider adding a subwoofer. If your loudspeaker system always seems to need additional high-frequency (boost) equalization, consider adding a super-tweeter instead.

Biamplified or triamplified (etc) systems, designed from separate components, may need some equalization to smooth out their frequency response. In most cases, what will be needed is *reduction* of some frequency bands, however. If more than 3 or 4 dB of boost seems necessary your loudspeaker system may need additional (or different) components.

Choosing and Using Loudspeakers

Types of Loudspeakers

There are two basic ways to purchase a loudspeaker system: as a pre-packaged system and as a component system.

Pre-packaged systems are usually designed and built in a single enclosure by a manufacturer. Most pre-packaged systems are designed for low to medium SPL applications. In groups, however, they may be useful in medium to high SPL applications. Some pre-packaged systems are designed primarily for portable usage and come with handles and corners and a protective finish. Others are designed primarily for permanently installed systems and come in furniture finish or neutrally finished wood enclosures. Because they are manufactured on an assembly line, pre-packaged systems are usually a better value per dollar spent. However, a pre-packaged system cannot offer the versatility of a component system.

Component systems are constructed from individual woofers, midrange loudspeakers and tweeters and may be assembled by a dealer or by a knowlegeable end user. A suitable component system can be assembled for any permanent or portable application. Because they are custom-assembled, a component system may cost more than an equivalent pre-packaged system (unless you do much of the work yourself). Component systems, however, can be custom-designed to fit the exact requirements of your application.

Two-Way and Three-Way and So On

A few loudspeaker systems are "one-way," that is, they use only one type of loudspeaker to cover the full range. Column speakers are often "one-way"; so are some of the loudspeakers which use an external, active equalizer. Most loudspeaker systems, however, are two-way, three-way or multi-way. That is, they use two, three or more different *types* of components to cover the audio frequency range.

Two way systems are common in permanent installations and speech-only systems. Some two-way systems are designed for low-to medium SPL entertainment (music and voice) applications. Three-way and multi-way systems, however, are more common for medium and high SPL entertainment systems and four-way and even five-way systems are used for some high SPL systems.

Woofers and Tweeters

Most woofers (low-frequency loudspeakers) are cone type loudspeakers. 12" diameter or even 10" diameter loudspeakers may be used as woofers in small systems. 15" diameter and 18" diameter loudspeakers are almost always used in larger systems. A small diameter woofer may be able to produce very low frequencies quite well but a larger diameter woofer will, in most cases, be able to produce those same low frequencies *at higher SPL levels*. In trade, the smaller woofers usually have better midrange response which makes them a good choice for two-way systems (which have no separate midrange component).

Woofers may be installed in simple, sealed wooden enclosures (often called "infinite baffle" enclosures). A "vented" or "ported" or "bass reflex" enclosure has a hole or tube in the front baffle which can improve the low-frequency response of the woofer (compared to an infinite baffle enclosure).

Some woofers are "horn-loaded," that is, they are placed behind a low-frequency horn which is usually a part of the enclosure. Horn-loading can increase efficiency and provide a measure of control over the woofer's dispersion. In order for a horn to work at low frequencies, however, it must be very large and the horns that are used with most woofers actually work well only in the midrange. Thus, horn-loaded woofers are most often seen in two-way systems where the woofer covers at least part of the midrange frequencies. Horn loading is often combined with a vented enclosure. In this case, the horn section aids the woofer's performance in the midrange, and the vented enclosure aids the woofer's performance in the low frequencies. This type of system is sometimes called a "vented horn."

Tweeters (high-frequency loudspeakers) come in several varieties. Hi-fi and low-level pro audio loudspeaker systems sometimes use small cone loudspeakers or dome-type loudspeakers for tweeters. Almost all medium to high level pro audio loudspeaker systems, however, use some form of "compression driver" and horn to cover the high frequencies. A "compression driver" is a device that works much like a cone loudspeaker, that is, it has a magnet, voice-coil and a "cone" or, more likely, a "dome." A compression driver, however, has a device known as a "phase plug" between the dome and the output of the driver and it has a "compression chamber" behind the dome, from whence comes its name. Compression drivers are usually very efficient. Typically, a compression driver will produce from 4 to 10 times as much sound per electrical watt as a cone type loudspeaker. It would be very difficult, however, to design a compression driver to work well at low frequencies, so compression drivers are used as midrange and high-frequency components.

A compression driver *must* be connected to a horn, or in some cases, a "lens" (which performs much the same function as a horn). The horn "loads" the driver in much the same way that an enclosure "loads" a woofer. This horn-loading makes possible the efficiency of the compression driver in the same way that horn-loading a woofer can improve its efficiency. The horn also helps control the dispersion of the sound.

There are several types of horns. "Exponential" horns come in two varieties: "straight" and "radial." Straight exponential horns are low cost, small in size (for a given frequency range) and usually have a pleasant sound quality but may become very "beamy" (narrow dispersion pattern) at high frequencies. Radial exponential horns are slightly higher in cost, about the same size (for a given frequency range) as straight exponential horns, and usually have a pleasant sound quality. The dispersion pattern of a radial horn *in the horizontal plane* is usually fairly consistant over a wide frequency range. The dispersion in the vertical plane, however, usually narrows at the high frequencies in a similar manner to a straight horn. Radial horns, by the way, are often called "sectoral" horns. Although it is sometimes mis-used, the term "sectoral" means that the horn is made from a sector of a sphere. "Sectoral" has nothing to do with "sections" in the horn. Straight horns are sometimes used in packaged loudspeaker systems but radials are a more common choice. Because a well-designed straight or radial horn usually has a very pleasing sound quality, they are often used in entertainment (music plus voice) oriented loudspeaker systems.

Multicell horns are actually made from groups of narrow-coverage-angle straight exponential horns. Multicells were an early, and reasonably successful, attempt to overcome the high-frequency beaming problem of straight exponential horns. Multicells are primarily used in voice-only systems and, because of their good dispersion control, they are often used in highly reverberant rooms. (Good dispersion control means you can point the sound at the audience and do a good job of keeping it away from the walls and ceiling.)

So-called "constant directivity" horns are a relatively new development. Now offered by several manufacturers, constant directivity horns have very good dispersion control, in both the horizontal and vertical planes, over a wide frequency band. At least some constant directivity horns have very good sound quality. Unfortunately, for a given frequency range, *true* constant directivity horns tend to be somewhat larger than other horn types.

In a two-way system, the woofer and tweeter share the midrange frequencies. In a three-way or multi-way system, a separate component is used to cover the midrange. That component may be a compression horn and driver or it may be a cone-type loudspeaker, depending on the frequency range it must cover and its SPL output.

Some systems include "subwoofers" or "super-tweeters" or both. A subwoofer is designed to extend the low-frequency response of a loudspeaker system or to improve the SPL capabilities of a system in the lower frequencies. Subwoofers are usually 15" or 18" loudspeakers in vented enclosures although some are also horn loaded. Super-tweeters are designed to extend the high-frequency response of a loudspeaker system. Sometimes, a compression driver and horn are used as a super-tweeter. "Ring-radiators," "piezo-electric" tweeters and other devices are also used as super-tweeters.

Choosing Loudspeakers

Besides the obvious question of budget, here are a few other things to consider in making a choice of loudspeakers:

1) **Power Handling** The loudspeaker system must be able to handle the full power output of your power amplifier (200 watts per channel

for your 3000 Mixer) for an extended period of time over the full rated frequency range of the loudspeaker.

2) **Frequency Range and Response** The loudspeaker's response should be smooth over its intended operating range. If your system will be used primarily for voice, you can choose a loudspeaker system whose low-frequency response is limited to as high as 70 or 80 Hz. If you must reinforce an entire musical group, especially a popular musical group, the system's low-frequency response should extend down to about 40 Hz.

3) **Sensitivity** This is a measure of the loudspeaker's efficiency. It tells you how many dB SPL the loudspeaker will produce at a given distance from the loudspeaker when the input power is a certain number of watts. High sensitivity is an advantage because it not only increases maximum SPL output capabilities, it also improves headroom. Remember that a decrease of only 3dB in sensitivity means double the amplifier power needed to maintain the same SPL!

4) **Coverage Pattern** In a pre-packaged system, you will usually get a "short-throw" coverage pattern (about 90 degrees horizontal by 40 degrees vertical). In a component system, you can choose several mid and/or high-frequency horns with different coverage patterns so that you can have "long-throw," "medium-throw" and "short-throw" devices. Long-throw horns are usually 40 degrees horizontal by 20 degrees vertical and are usually only needed in large concert systems and permanently installed systems. Medium throw horns are usually 60 degrees horizontal by 40 degrees vertical and are valuable in many portable as well as permanent systems to reach farther back in an audience. Short throw horns are usually 90 degrees horizontal by 40 degrees vertical and are used to reach the front of an audience or may be used to cover an entire audience in a small portable system.

5) **Sound Quality** Only your ears can tell you the answer to the all-important question: "How does it *sound*?" It's an entirely subjective evaluation and that means that your own personal tastes play an important part. That's the way it should be, of course. In a sense, the sound system is your "instrument," and it should sound like you want it to sound. Yet your goal is not to alter but to *reinforce* and, to some extent, to enhance the sound of a performance. Your subjective evaluation of the sound quality of a loudspeaker system, then, should be based on how well you believe that loudspeaker system will *accurately* reinforce your performance.

You must be familiar with the way your performance sounds *without* reinforcement to judge the accuracy of a loudspeaker system. In addition, you should do your listening tests with live sources if at all possible. Recorded music of any kind, especially if played from a cassette machine or tuner, hides many defects in a loudspeaker system. Live music, or even a simple live microphone test, because of its increased dynamic range and transients, reveals the true nature of a loudspeaker system. If, on the other hand, you're buying a set of loudspeakers for disco use, by all means, evaluate them with recorded music!

Using Loudspeakers

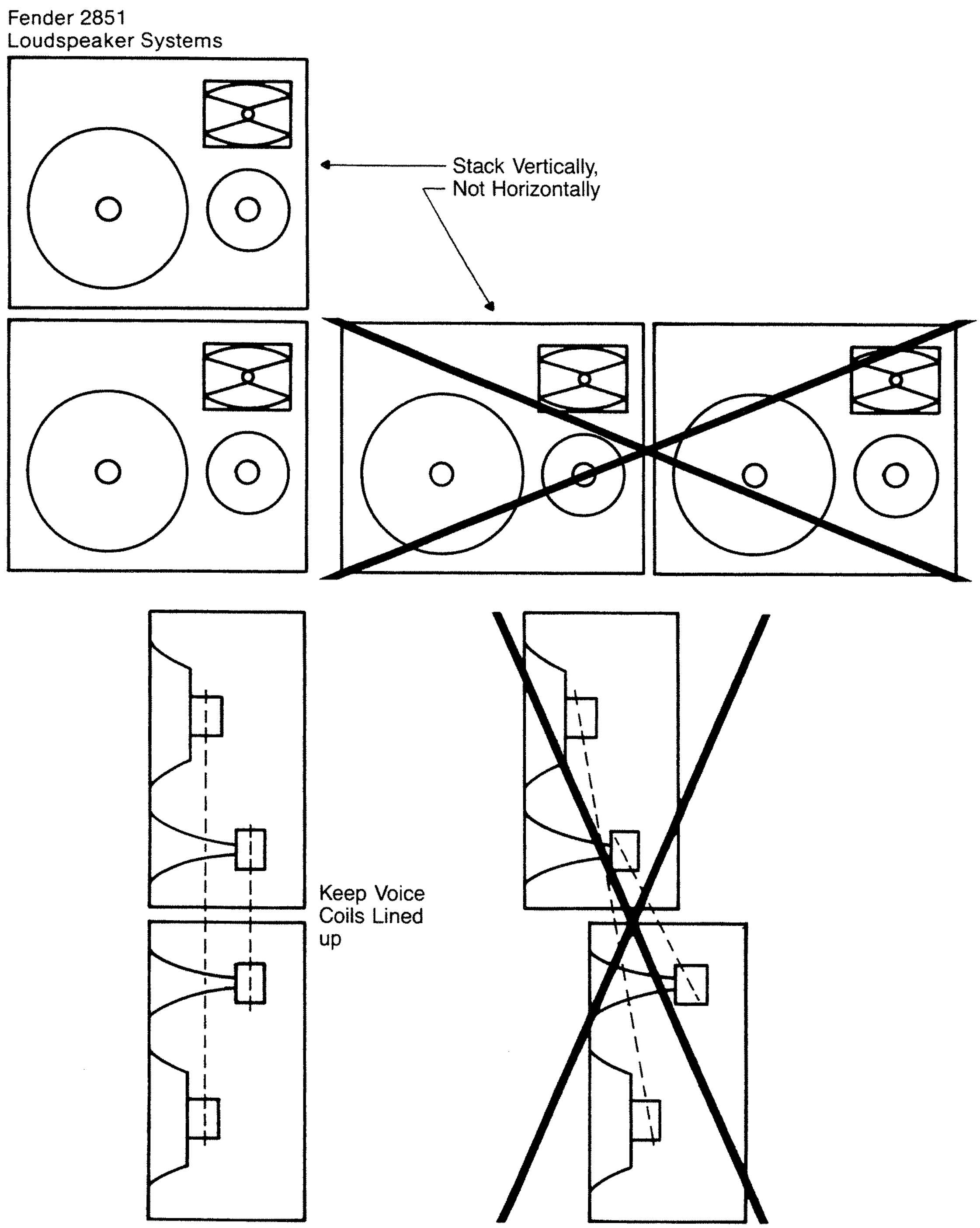
The "one on each side of the stage" system can work quite well for a portable system. In permanent installations, however, except in very low-ceiling rooms, a single loudspeaker "cluster" usually works better than a pair of widely separated loudspeakers. The reason is the "phasing" problem discussed in the section on microphones. Here are some additional tips on loudspeaker usage:

1) **Keep Voice Coils in Line** When stacking loudspeakers, try to keep their voice coils lined up in a vertical line. This will help minimize those "phasing" problems.

2) **Stack Vertically not Horizontally** Whenever possible, stack two loudspeakers on top of each other, not side by side. Again, this helps minimize phasing problems.

3) **Keep High-Frequency Components Together** When stacking two of the same kind of packaged loudspeaker system, turn the top one upside down so that the horns are close together. This can improve the "throw" of the stack in the high-frequencies.

Stacking Loudspeakers

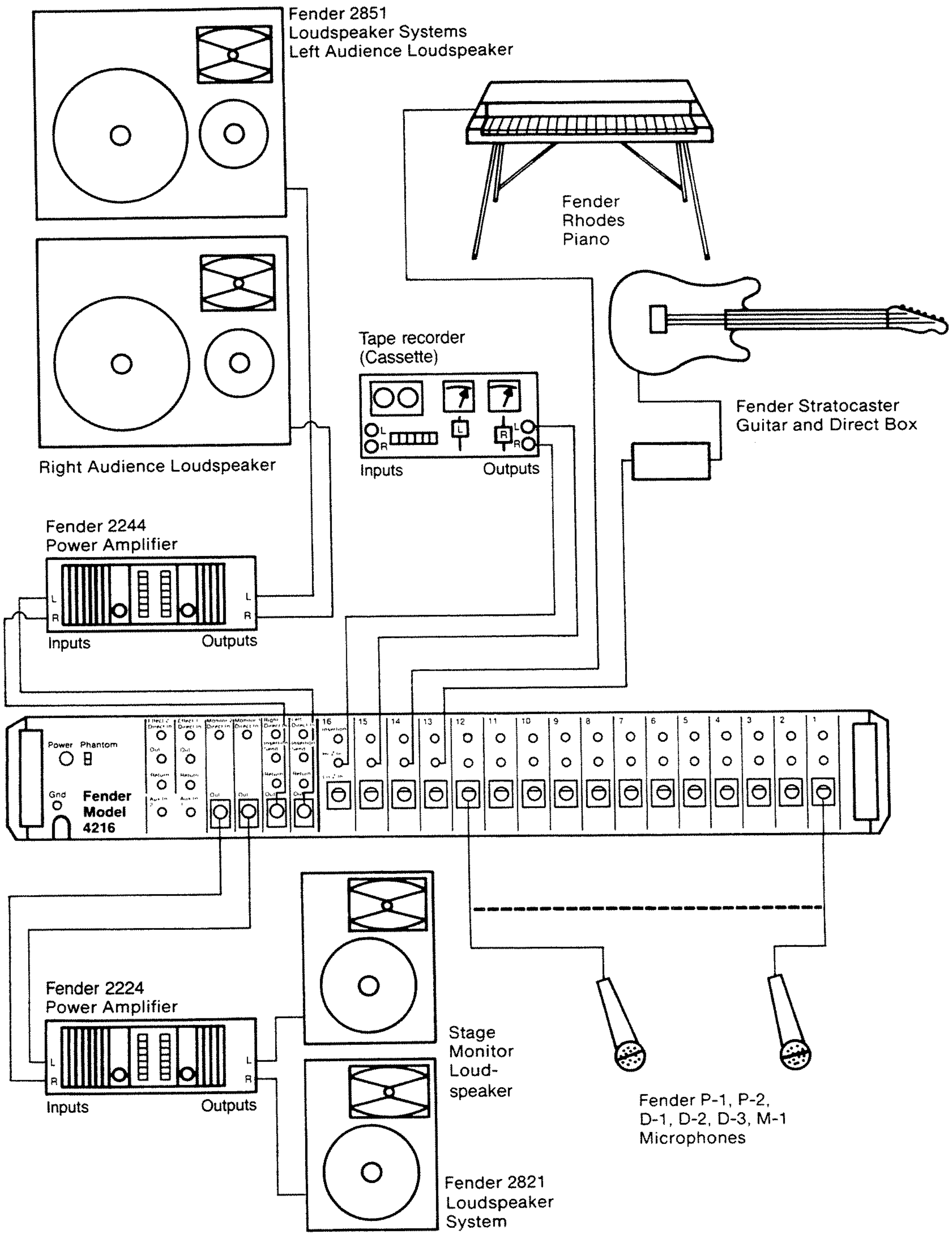


Section IV: Examples of Some Systems Using the Fender 4000 Series Mixers

A Portable Entertainment System

This simple system shows how the Fender 4000 Series Mixers can be the heart of a cost-effective yet versatile pro sound system. This system would be ideal for working through the exercises in Section 1 of this manual. It may also be all you need for a small club system when monitors aren't needed.

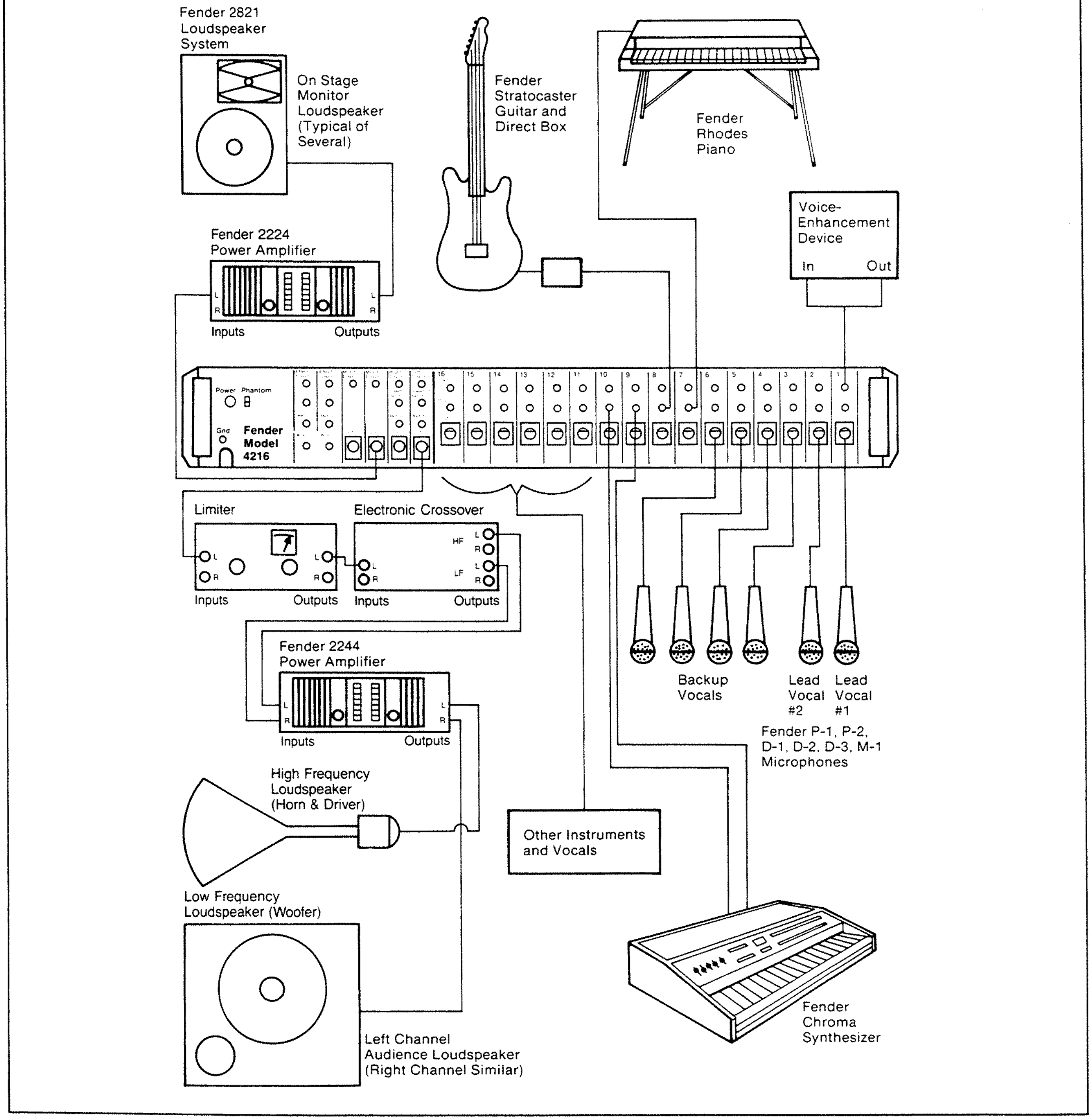
A Portable Entertainment System



A Larger Portable Entertainment System

In this system, a Fender 2224 and 2244 power amplifier are used to power a set of biamplified loudspeaker systems and a second 2224 powers a set of on-stage (foldback) monitors. In addition, we show a stereo limiter used on the Program mixes and an external voice-enhancement device used on one microphone.

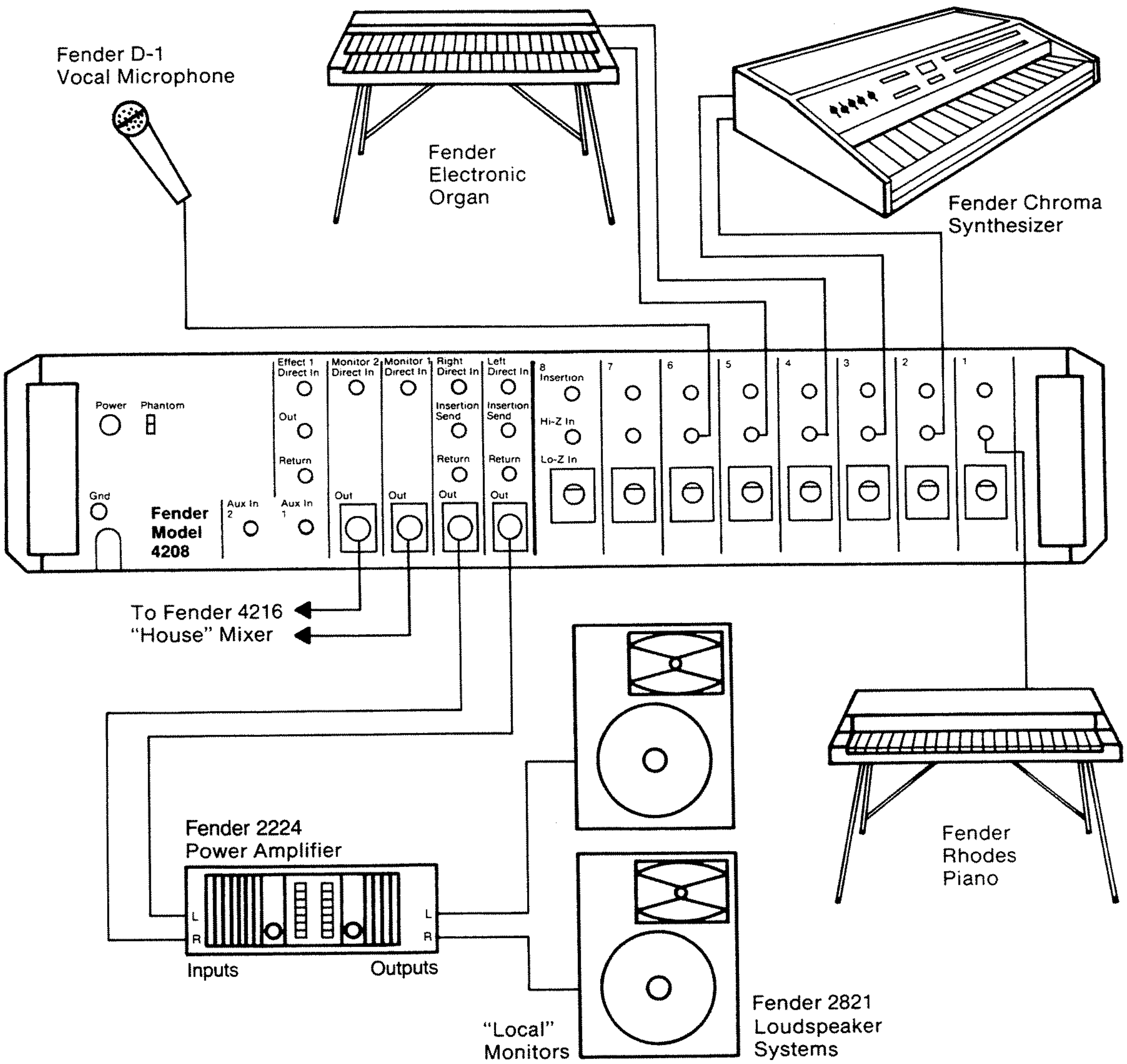
A Large Portable Entertainment System



An Instrument Mixing System

A Fender 4000 Series Mixer makes a great keyboard mixer. The Hi-Z inputs can be adapted (using the Trim controls) to the outputs of just about any keyboard and a Fender 2224 power amplifier can power "local" keyboard monitor loudspeakers. The Program Out jacks in this system are used to feed the main house mixer while the lower-level Insertion Out jacks are used to feed a semi-pro tape machine.

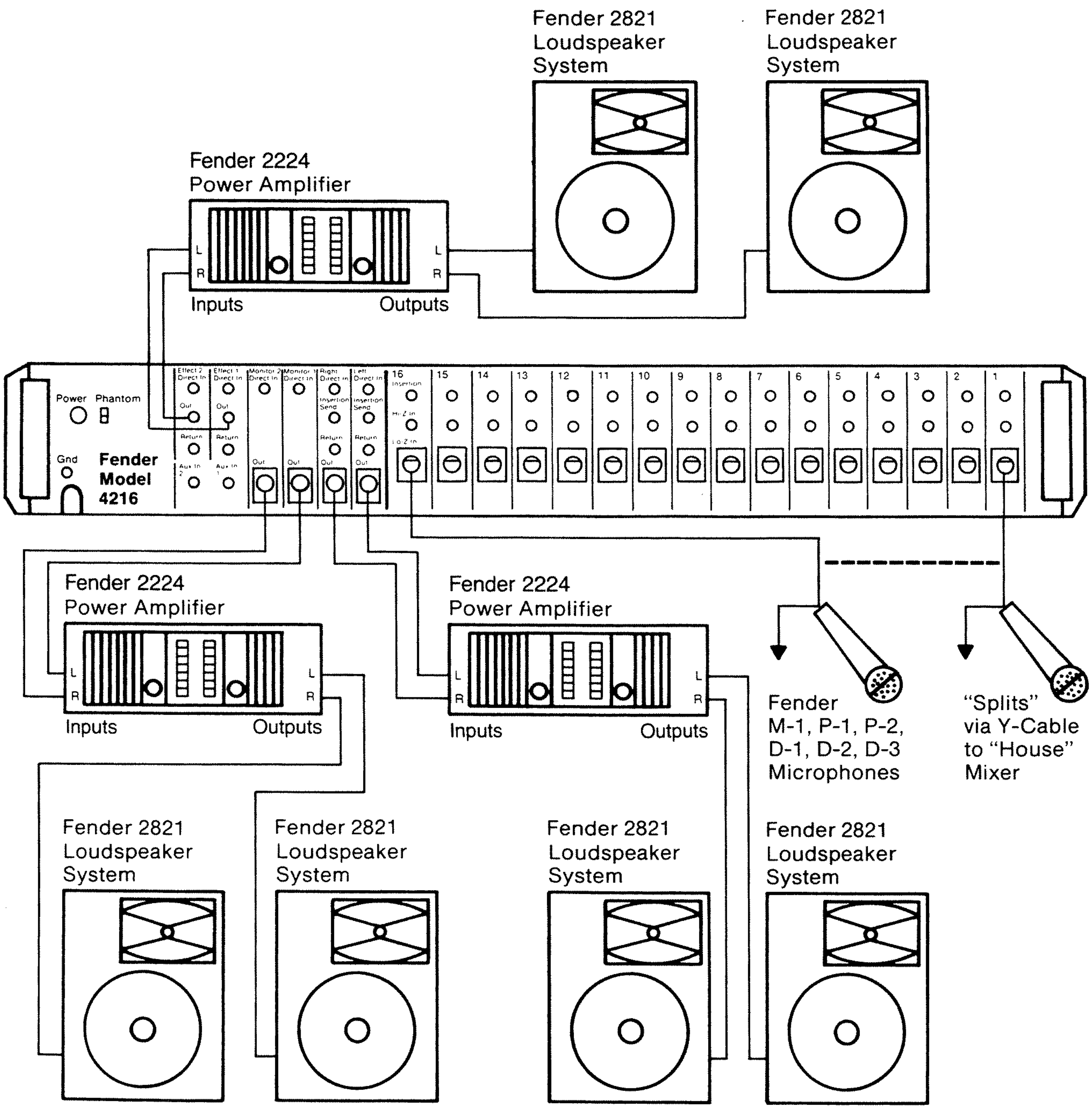
An Instrument Mixing System



A Monitor Mixing System

Larger portable sound systems often have a separate mixer just for the monitors. Your 4000 Mixer can be a versatile monitor mixer because it has as many as six separate mix buses, for six separate monitor mixes: Program Left and Right, Monitor 1 and 2 and Effect 1 and 2 (five mix buses on the 4208 and 4212 which have a single Effect bus). In addition, your 4000 Mixer has two Graphic Equalizers (on the Program Outputs) to help control feedback.

A Monitor Mixing System

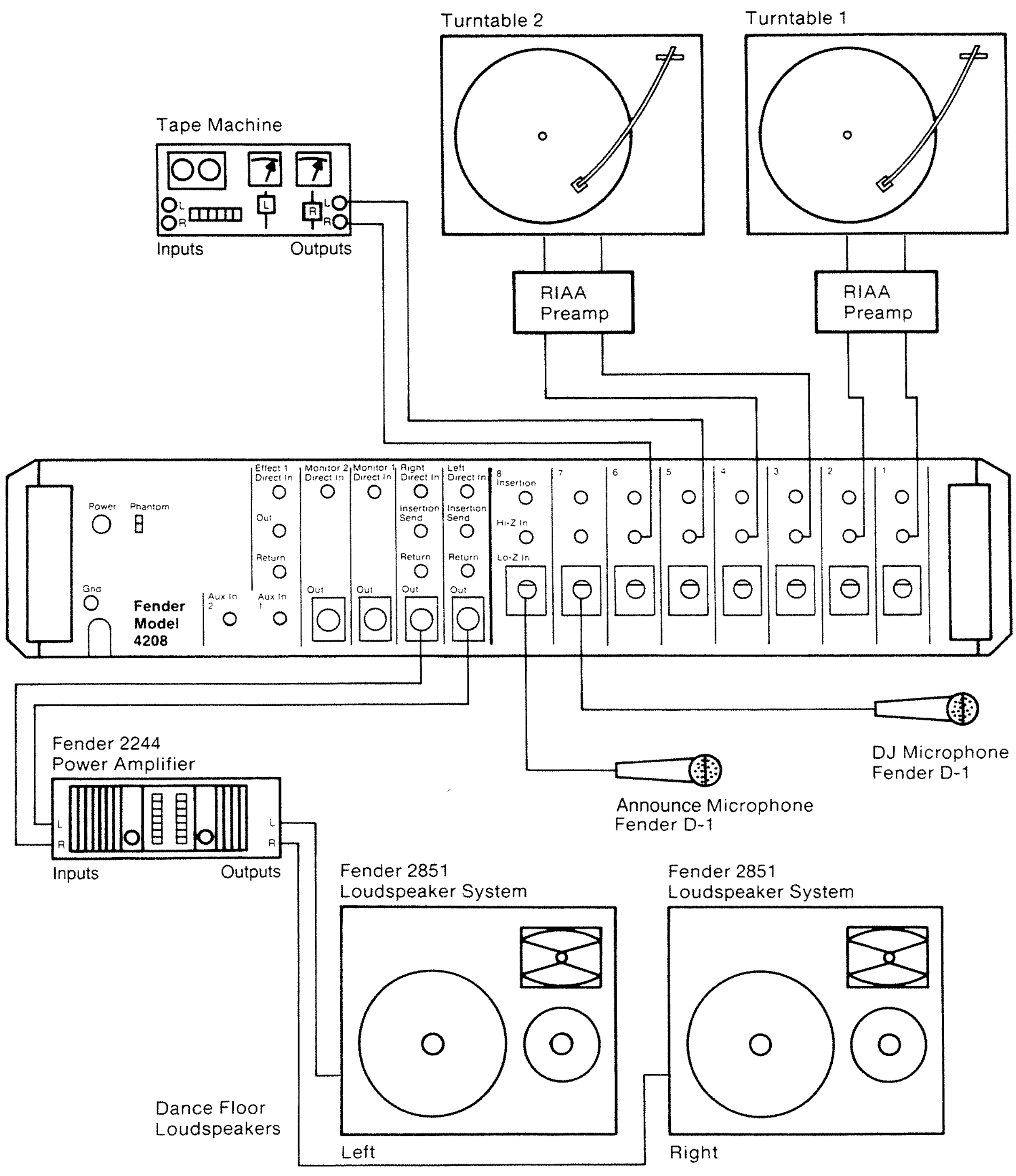


A Music Playback System

With external phono preamplifiers, a Fender 4000 mixer becomes a disco mixer. To operate a 4000 Mixer as a stereo disco mixer, pan one Input Channel fully left and the next Input Channel fully right and use these two Input Channels for the left and right outputs of a phono preamp. Then, bring these two channels up and down together to fade one turntable "in" (and use a different pair of Input Channels to fade the other turntable "out"). Use the Phones system to "listen-in" on what's going on in your system, and, if you have a 4216, you can use the Cue system to "cue up" one turntable while another is still playing!

Use your 4000 Mixer's Graphic Equalizers to enhance the sound quality of a particular record and use the VU Meters and the Peak LEDs to help maximize sound output while avoiding distortion.

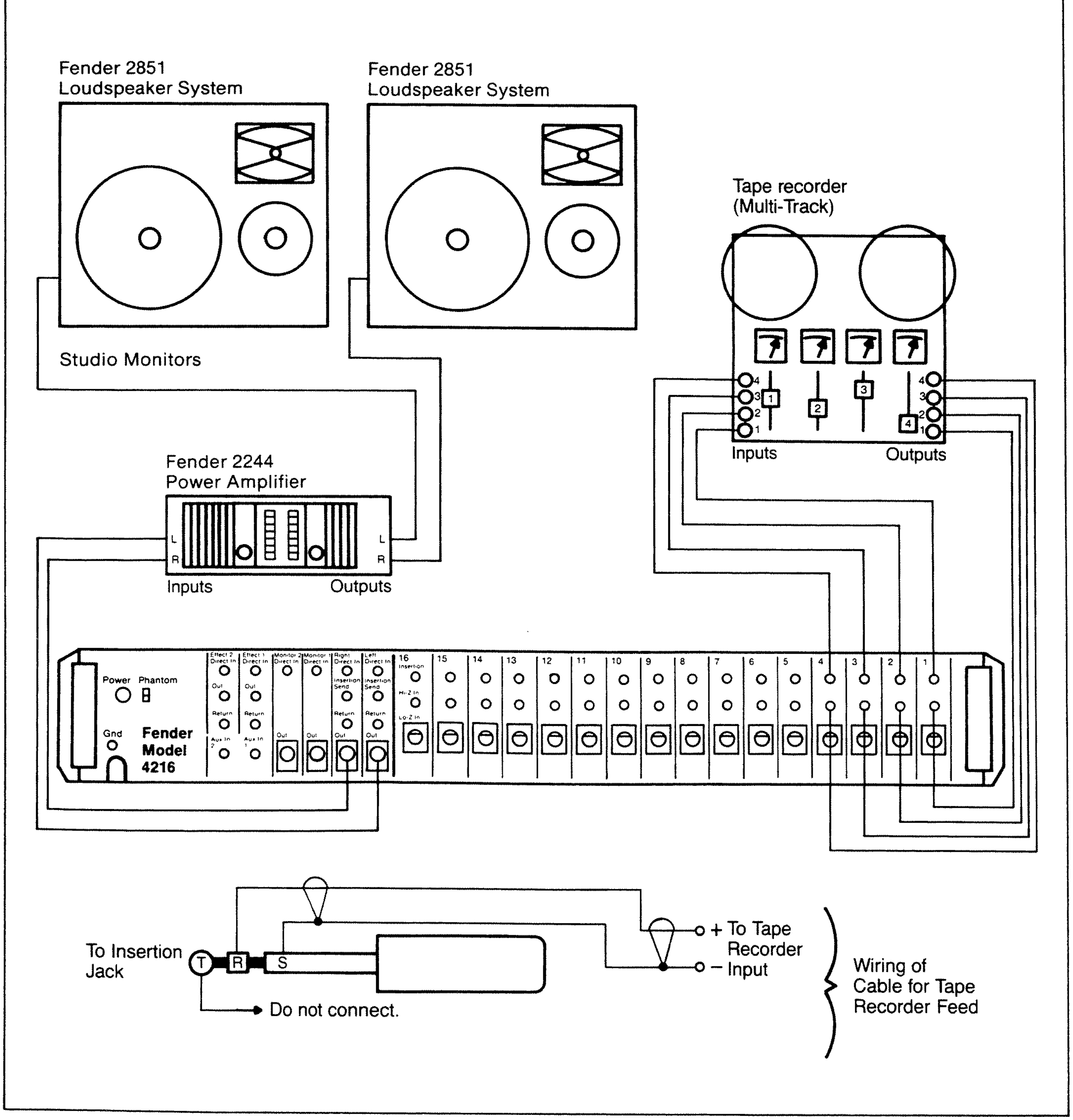
A Music Playback System



A Recording System

A Fender 4000 Mixer can be used for both recording and mixdown. During recording, use the tape machine's input controls to set levels for the recording and use the 4000 Mixer's controls (and the Phones system) to monitor the recording in process. This setup records unequalized "dry" tracks. During mixdown, you can use the Input Channel Equalization controls and the Program Equalizers as well as an external reverberation (or other effect device to enhance the mixdown recording.

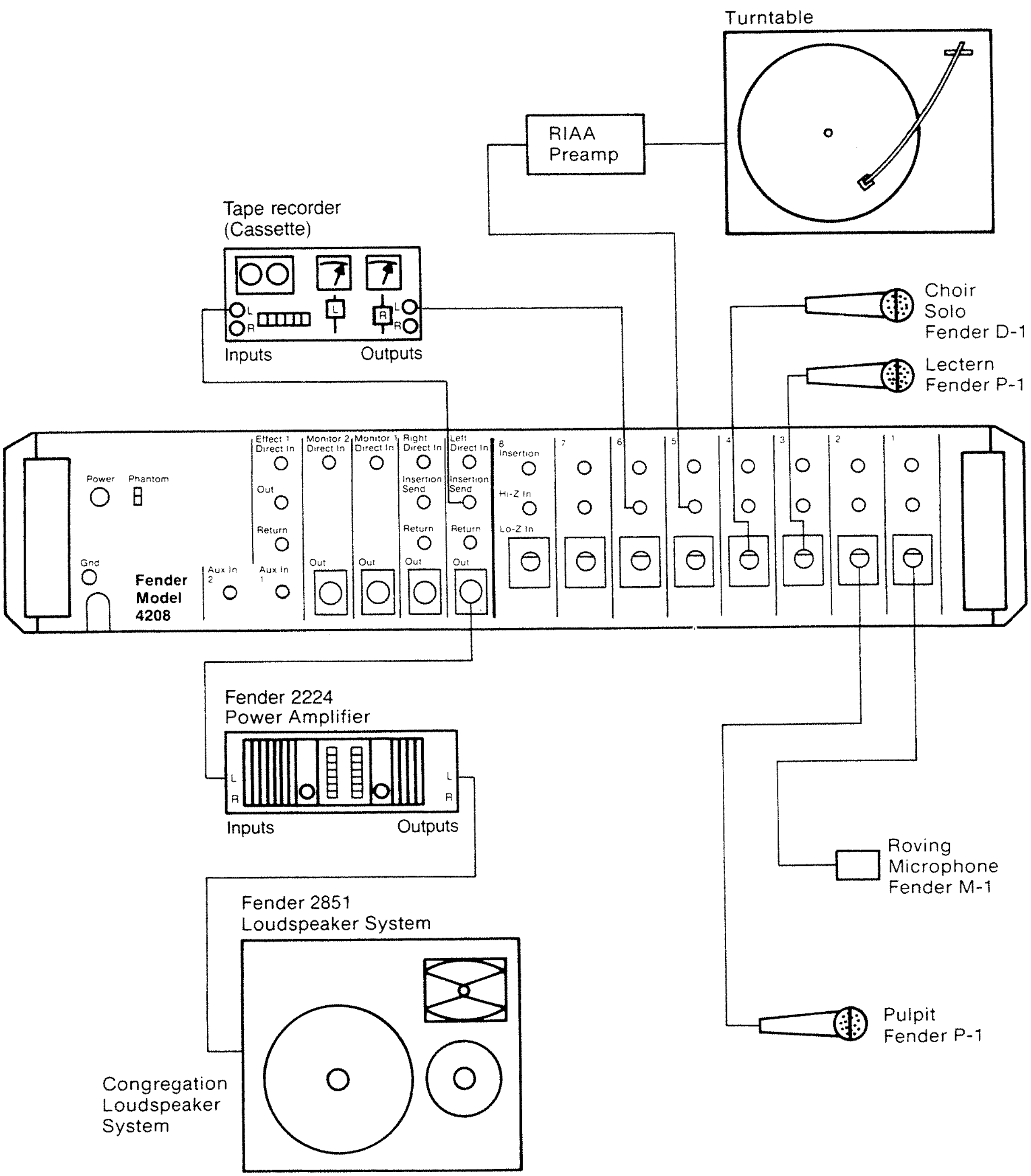
A Recording System



A Fixed Sound Reinforcement System

In this example, a Fender 4208 Mixer is used to provide cost effective yet versatile mixing and equalization capabilities in a small house of worship sound system. A Fender 2224 power amplifier completes the system. The second channel of the 2224 could be used for choir monitors or it could feed ceiling loudspeakers in classrooms, or "privacy" or "mother's" rooms.

A Fixed Sound Reinforcement System



For More Information

For more information on your Fender product, or for warranty or service information, please contact your Fender dealer or write Fender at the following address:

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Customer Service
1300 East Valencia Drive
Fullerton, California 92631

Fender Musical Instruments

1300 East Valencia Drive
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