SECTION 5. GAIN STRUCTURE AND LEVELS

5.1 STANDARD OPERATING LEVELS

There are a number of different "standard" operating levels in audio circuitry. It is often awkward to refer to a specific level (i.e., +4 dBu) when one merely wishes to describe a general sensitivity range. For this reason, most audio engineers think of operating levels in three general categories:

A. MIC LEVEL OR LOW LEVEL

This range extends from no signal up to about -20 dBu (77.5 mV), or -20 dBm (77.5 mV across 600 ohms = 10 millionths of a watt). It includes the outputs of microphones, guitar pickups, phono cartridges, and tape heads, prior to any form of amplification (i.e., before any mic, phono, or tape preamps). While some mics can put out more level in the presence of very loud sounds, and a hard-picked guitar can go 20 dB above this level (to 0 dBu or higher), this remains the nominal, average range.

B. LINE LEVEL OR MEDIUM LEVEL

This range extends from -20 dBu or -20 dBm to +30 dBu (24.5 V) or +30 dBm (24.5 V across 600 ohms = 1 watt). It includes electronic keyboard (synthesizer) outputs, preamp and console outputs, and most of the inputs and outputs of typical signal processing equipment such as limiters, compressors, time delays, reverbs, tape decks, and equalizers. In other words, it covers the output levels of nearly all equipment except power amplifiers. Nominal line level (the average level) of a great deal of equipment will be -10 dBu/dBm (245 millivolts), +4 dBu/dBm (1.23 V) or +8 dBu/dBm (1.95 V).

C. SPEAKER LEVEL AND HIGH LEVEL

This covers all levels at or above +30 dBu (24.5V) or +30 dBm (24.5 V across 600 ohms = 1 watt). These levels include power amplifier speaker outputs, AC power lines, and DC control cables carrying more than 24 volts.

NOTE: A piece of consumer sound equipment ("hi-fi") may operate at considerably lower nominal (average) line levels than +4 dBu. This is typically around -16 dBu (123 mV) to -10 dBu (245 mV) into 10,000 ohms or higher loads. Peak output levels in such equipment may not go above +4 dBu (1.23 V). The output current available here would be inadequate to drive a 600-ohm terminated circuit, and even if the professional equipment has a higher impedance input, the output voltage of the hi-fi equipment may still be inadequate. The typical result is too-low levels and too-high distortion. This can damage loudspeakers (due to the high frequency energy content of the clipped waveform), and it can damage the hi-fi equipment (due to overloading of its output circuitry). There are exceptions, but one should be very careful to check the specifications when using consumer sound equipment in a professional application.

Let's discuss these levels in the context of a sound system. The lowest power levels in a typical sound system are present at the output of microphones or phono cartridges. Normal speech at about one meter from the "average" dynamic microphone produces a power output from the microphone of about one trillionth of a watt. Phono cartridges playing an average program selection produce as much as a thousand times this output – averaging a few billionths of a watt. These signals are very weak, and engineers know that they cannot be "run around" a chassis or down a long cable without extreme susceptibility to noise and frequency response errors. This is why microphone and phono preamps are used to boost these very low signal levels to an intermediate range called "line level." Line levels are between 10 millionths of a watt and 250 thousandths of a watt (1/4 watt). These levels are related to the "dBm" unit of measurement as follows:

-20 dBm	=	10 microwatts	=	0.00001 watts
0 dBm	=	1 milliwatt	=	0.001 watts
+4 dBm	=	2.5 milliwatts	=	0.0025 watts
+24 dBm	=	250 milliwatts	=	0.025 watts
+30 dBm	=	1000 milliwatts	=	1.0 watts
+40 dBm	=		=	10.0 watts
+50 dBm	=		=	100.0 watts

While some console and preamp outputs can drive lower impedances, primarily for driving headphones, typical line levels (measured in milliwatts) cannot drive speakers to usable levels. Not only is the power insufficient for more than "whisper" levels, the console circuits are designed to operate into loads of 600 ohms to 50,000 ohms; they cannot deliver even their few milliwatts of rated power to a typical 8-ohm speaker without being overloaded. A power amplifier must be used to boost the power output of the console so it is capable of driving low impedance speaker loads and delivering the required tens or hundreds of watts of power.

5.2 Dynamic Range and Headroom

5.2.1 What Is Dynamic Range?

Every sound system has an inherent noise floor, which is the residual electronic noise in the system equipment (and/or the acoustic noise in the local environment). The dynamic range of a system is equal to the difference between the peak output level of the system and the noise floor.

5.2.2 The Relationship Between Sound Levels and Signal Levels

A concert with sound levels ranging from 30 dB SPL (near silence) to 120 dB SPL (threshold of pain) has a 90 dB dynamic range. The electrical signal level in the sound system (given in dBu) is proportional to the original sound pressure level (in dB SPL) at the microphone. Thus, when the program sound levels reach 120 dB SPL, the maximum line levels (at the console's output) may reach +24 dBu (12.3 volts), and maximum power output levels from a given amplifier may peak at 250 watts. Similarly, when the sound level falls to 30 dB SPL, the minimum line level falls to -66 dBu (0.388 millivolts) and power amplifier output level falls to 250 nanowatts (250 billionths of a watt).

The program, now converted to electrical rather than acoustic signals, still has a dynamic range of 90 dB: +24 dBu - (-66 dBu) = 90 dB. This dB SPL to dBu or dBm correspondence is maintained throughout the sound system, from the original source at the microphone, through the electrical portion of the sound system, to the speaker system output. A similar relationship exists for any type of sound reinforcement, recording studio, or broadcast system.

Note: Refer to Figure 5-1 (next page) while reading the following disucssions of headroom and dynamic range.

5.2.3 A Discussion Of Headroom

The average line level in the typical commercial sound system just described is +4 dBu (1.23 volts), corresponding to an average sound level of 100 dB SPL. This average level is usually called the "nominal" program level. The difference between the nominal and the highest (peak) levels in a program is the headroom. In the above example, the headroom is 20 dB. Why is this so? Subtract the nominal from the maximum and see: 120 dB SPL - 100 dB SPL = 20 dB. The headroom is always expressed in just plain "dB" since it merely describes a ratio, not an absolute level; "20 dB" is the headroom, not "20 dB SPL". Similarly, the console output's electrical headroom is 20 dB, as calculated here: +24 dBu - (+4 dBu) = 20 dB. Again, "20 dB" is the headroom, not "20 dBu". Provided the 250-watt rated power amplifier is operated just below its clipping level at maximum peaks of 250 watts, and at nominal levels of 2.5 watts, then it also operates with 20 dB of headroom (20 dB above nominal = 100 times the power).

5.2.4 What Happens When The Program Source Has Wider Dynamics Than The Sound Equipment?

If another mixing console were equipped with a noisier input circuit and a less capable output amplifier than the previous example, it might have an electronic noise floor of -56 dBu (1.23 millivolts), and a peak output level of +18 dBu (6.16 volts). The dynamic range of this system would only be 74 dB. Assuming the original program still has an acoustic dynamic range of 90 dB, it is apparent that 16 dB of the program will be "lost" in the sound system. How is it lost? There may be extreme clipping of program peaks, where the output does not rise higher in response to higher input levels. Quiet passages, corresponding to the lowest signal levels, may be buried in the noise. Typically, portions of that 16 dB difference in dynamic range between the sound system capability and the sound field at the microphone will be lost in both ways. A system with +24 dBu output capability and a -66 dBu or better noise floor, or +18 dBu output capability and -82 dBu noise floor, would be able to handle the full 90 dB dynamic range. Thus, for high quality sound reinforcement or music reproduction, it is necessary that the sound system be capable of low noise levels and high output capability.

In the special case of an analog audio tape recorder, where the dynamic range often is limited by the noise floor and distortion levels of the tape oxide rather than the electronics, there is a common method used to avoid program losses due to clipping and noise. Many professional and consumer tape machines are equipped with a noise reduction system, also known as a compander (as designed by firms like Dolby Laboratories, Inc.). A compander noise reduction system allows the original program dynamics to be maintained throughout the recording and playback process by compressing the program dynamic range before it goes onto the tape, and complementarily expanding the dynamic range as the program is retrieved from the tape. Compact (laser) discs, and digital audio tape recording, and the FM or vertical recording used in modern stereo VCR soundtracks are all additional methods of recording wide dynamic range programs which, in turn, demand playback systems with wide dynamic range.

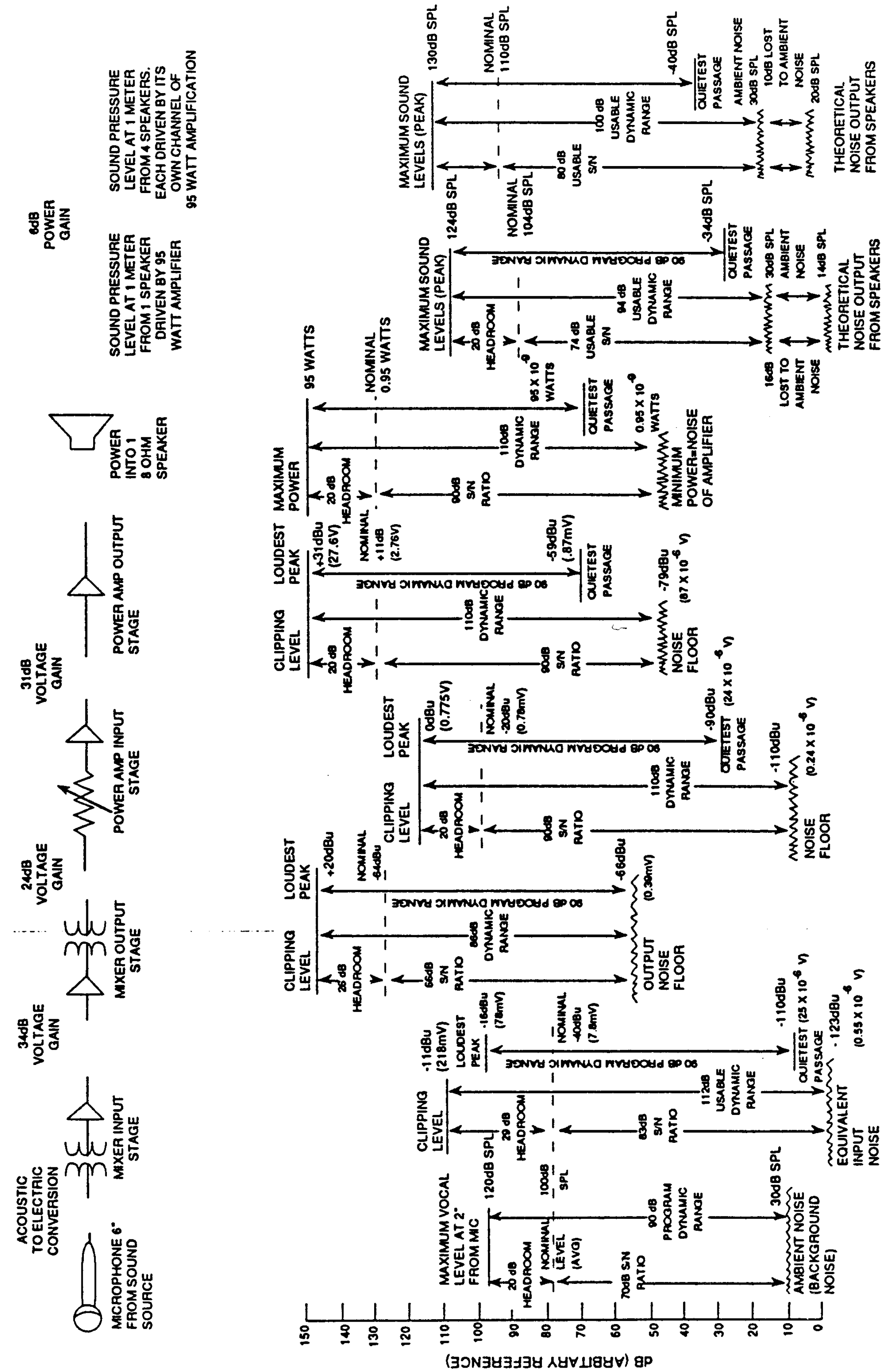


Figure 5-1. Dynamic Range and Headroom in Sound Systems

5.2.5 A General Approach To Setting Levels In a Sound System

Just because individual pieces of sound equipment are listed as having certain headroom or noise and maximum output capability, there is no assurance that the sound system assembled from these components will yield performance anywhere near as good as that of the least capable component. Volume control and fader settings throughout a sound system can dramatically affect that performance.

To provide the best overall system performance, level settings should be optimized for each component in the system. One popular approach is to begin by adjusting levels as close as possible to the signal source. In this case, the primary adjustments are made on the console input module. Set the input PAD and GAIN trim controls for the maximum level that will not produce clipping (i.e., avoid overdriving the input stage); this can be seen by examining the green "signal" and red "peak" LEDs, and in some cases it can be heard by listening for distortion while making PAD and GAIN adjustments. The next step is to set the level of the console input channel (the channel fader and/or the appropriate aux send control) so that it properly drives the mixing busses. You can refer to the VU meters to examine the bus levels.

If line amplifiers, electronic crossovers, equalizers or other signal processing devices are inserted in the signal chain, signal levels at the input of these units should be set so the dynamic range of each unit is optimized. In other words, set the input level at each device as high as possible without producing clipping, and, if an output level control is provided, also set it as high as possible without clipping the output – and without causing clipping in the input of the next device to which it is connected.

Check the operating manual of each piece of equipment to determine the specified nominal and maximum input levels. An accurate AC voltmeter is often helpful for verifying levels. As a rule, keep signal levels as high as possible throughout the system, up to the input of the power amplifier(s); at that point, reduce the program level, as required to achieve a given headroom value, using the amplifier's input attenuators. Input attenuators should be set so that maximum program levels from the source equipment won't drive the amplifiers to clipping (or at least, won't do it very often). This keeps overall system noise as low as possible.

5.2.6 How To Select a Headroom Value and Adjust Levels Accordingly

Recall that headroom is the amount of level available for peaks in the program that are above the average (nominal) signal level.

The choice of a headroom figure depends on the type of program material, the application, and the available budget for amplifiers and speakers. For a musical application where high fidelity is the ultimate consideration, 15 dB to 20 dB of headroom is desirable. For most sound reinforcement applications, especially with large numbers of amplifiers, economics play an important role, and a 10 dB headroom figure is usually adequate; in these applications, a limiter can help hold program peaks within the chosen headroom value, and thus avoid clipping problems. For the extreme situation (as in a political rally) where speeches and other program material must be heard over very high noise levels from the crowd, as well as noise from vehicular and air traffic, yet maximum levels must be restricted to avoid dangerously high sound pressure levels, a headroom figure of as low as 5 or 6 dB is not unusual. To achieve such a low headroom figure, an extreme amount of compression and limiting will be necessary; while the sound may be somewhat unnatural, the message will "cut through."

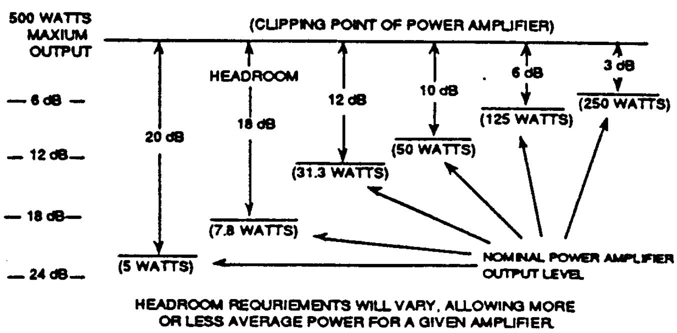


Figure 5-2. Headroom In Different Applications

Let's go through an actual setup procedure for a high quality, music reproduction system. First choose a headroom figure. For maximum fidelity when reproducing music, it is desirable to allow 20 dB of headroom above the average system output. While some extreme musical peaks exceed 20 dB, the 20 dB figure is adequate for most programs, and allowing for greater headroom can be very costly. A 20 dB headroom figure represents a peak level that is one hundred times as powerful as the average program level. This corresponds to an average 0 VU indication on the PM4000M meters (0 VU ≈ +4 dBu, which allows 20 dB headroom before the console reaches its maximum +24 dBu output level).

Remember that with a 20 dB headroom figure, a power amplifier as powerful as 500 watts will operate at an average 5 watts output power. In some systems such as studio monitoring, where fidelity and full dynamic range are of utmost importance, and where sensitive loudspeakers are used in relatively small rooms, this low average power may be adequate. In other situations, a 20 dB headroom figure is not necessary and too costly due to the number of amplifiers required.

After choosing a headroom figure, adjust the incoming and outgoing signal levels at the various devices in the system to achieve that figure. For a typical system, the adjustments for a 20 dB headroom figure would be made as follows:

1. Initially, set the attenuators on the power amp at maximum attenuation (usually maximum counterclockwise rotation). Feed a sine wave signal at 1000 Hz to the console input at an expected average input level (approximately -50 dBu (2.45 mV) for a microphone, +4 dBu (1.23 volts) for a line level signal. The exact voltage is not critical, and 1000 Hz is a standard reference frequency, but any frequency from 400 Hz to about 4 kHz may be used.
2. Set the input channel fader on the console at its marked "nominal" setting, and adjust the channel Gain so that the channel's LED meter read zero. The meter should be set to the Post-Fader mode (MTR PRE switch [20] disengaged. Be sure this channel is assigned to an output bus (i.e., one of the group busses or the stereo bus).
3. Set the master fader for the bus to which the channel is assigned so that the output level is 20 dB below the rated maximum output level for the console. Suppose, for example, the maximum rated output level is +24 dBu (12.3 volts); in that case, the output level should be adjusted to +4 dBu (1.23 volts), as indicated by a "zero" reading on the console's VU meter (0 VU corresponds to +4 dBu with a steady-state sine wave signal output per factory calibration).
4. If the rated maximum input level for the graphic equalizer to which the console output is connected is +24 dBu (12.3 volts), then no adjustment or padding of the input to the EQ is required. If the maximum input level is lower, for example +18 dBu, then there would be reduced headroom in the EQ unless its input is attenuated. Subtracting +4 dBu from +18 dBu leaves only 14 dB of headroom, so in order to maintain the desired 20 dB of headroom, 6 dB of attenuation must be dialed in at the EQ input, or a 6 dB resistive pad should be inserted between the console output and the equalizer input. The nominal signal level at the input to the equalizer should now be -2 dBu (616 mV), which can be checked with a voltmeter.
5. Assume that the maximum rated output level of the equalizer in this example is +18 dBu (6.16 volts). Adjust the master level control on the equalizer so that its output level is 20 dB below the rated maximum, or -2 dBu (616 mV). If the equalizer has no built-in VU meter, use an external voltmeter to confirm this level.

NOTE: If the graphic equalizer is placed in the console's group or stereo INSERT IN / OUT loop, the nominal sensitivity of the input is +4 dBu, which may seem to be 6 dB less sensitive than required for the necessary headroom. However, any boost applied with the EQ will raise the nominal level of the signal at the EQ output, so this may help preserve adequate headroom in the console. Remember, though, that applying boost with an equalizer can reduce headroom within the EQ itself, so you may want to turn down the EQ's output level to preserve the headroom.

6. Finally, starting with the attenuator(s) on the power amplifier at maximum attenuation (maximum counterclockwise rotation), slowly decrease the attenuation (raise the level), observing the amplifier's output level. When the POWER output is 1/100 of the maximum rated power (1/10 of the maximum output voltage), the amplifier has 20 dB headroom left before clipping. A 250 watt amplifier would operate at nominal 2.5 watts, or a 100 watt amplifier at 1 watt, on average level passages in order to allow 20 dB for the loud peaks.

To operate this system, use only the controls on the console, and avoid levels that consistently peak the console's VU meter above the "zero" mark on its scale, or that drive the amplifier above a safe power level for the speaker system. Any level adjustments in the other devices in the system will upset this established gain structure.

If, for a given amount of headroom, portions of the program appear to be "lost in the noise," the answer is not to turn up the levels since that will merely lead to clipping and distortion. Instead, it will be necessary to use either a compressor, or to manually "ride the gain" of those console faders that are required to raise the level when the signals are weak. This effectively reduces the required headroom of the signal, allowing the lower level portions of the program to be raised in level without exceeding the maximum level capability of the system. Compressors can be used in the INSERT IN/OUT loops of individual channels (say for a vocalist with widely varying levels), or at the group, aux or stereo master INSERT IN/OUT points or after the Matrix Outputs when the overall mix has too much dynamic range. Of course, another alternative is available: add more amplifiers and speakers so that the desired headroom can be obtained while raising the average power level.

5.3 Gain Overlap And Headroom

As explained previously, the PM4000M can deliver +24 dBu output level, a level which exceeds the input sensitivity of most other equipment. A power amplifier's sensitivity, for example, is that input level which drives the amplifier to maximum output (to the point of clipping). Hence, a power amplifier with a +4 dBu sensitivity rating will be driven 20 dB into clipping if driven with the full output capability of the PM4000M. It would appear, then, that the console has "too much" output capability, but this is not really true.

In fact, there are a number of real-world instances when the +24 dBu output drive is very desirable. For one thing, if the console's output is used to drive multiple power amplifiers in parallel, then the input signal strength available to each amplifier is diminished. Thus, the overlap becomes less of an excess and more of a necessity.

In other cases, the PM4000M may be driving a passive device such as a passive filter set, graphic equalizer or low-level crossover network. Such devices will attenuate some of the signal, often 6 dB or more. Here, the extra output capability of the console offsets the loss of the passive signal processor so that adequate signal can be delivered to the power amplifiers, tape machine inputs, etc.

Consider those instances where the PM4000M outputs are connected to a tape machine. Many professional tape machines are subject to tape saturation at input levels above +15 dBu. Why would one want +24 dBu output from a console? Well, it turns out that analog tape has what is considered a "soft" saturation characteristic, whereby the distortion is not terribly harsh in comparison to the clipping of the typical solid state line amplifier. If the mixing console were to clip at +18 dBu, for example, that clipping would overlay a very harsh distortion on the 3 dB of "soft" saturation on the tape. Because the PM4000M does not clip until its output reaches +24 dBu, there is less chance of applying harsh distortion to the tape. Today, however, there is another consideration: digital recording technology. Here, the available dynamic range of the digital tape recorders or direct-to-disk recorders is so great that all the headroom a console can provide is advantageous.

Section 6.
Optional Functions

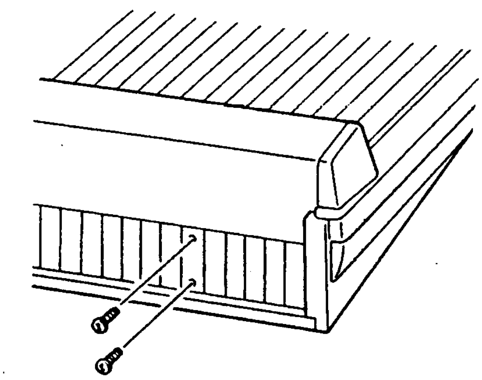
The PM4000M is factory wired to suit what Yamaha engineers believe to be the greatest number of applications. Yamaha recognizes, however, that there are certain functions which must be altered for certain specific applications. In designing the PM4000M, a number of optional functions have been built in, and can be selected by moving factory preset switches within certain modules.

WARNING: Underwriter's Laboratories (UL) requires that we inform you there are no user-serviceable parts inside the PM4000M. Only qualified service personnel should attempt to open the meter bridge, to remove a module, or to gain access to the inside of the console or power supply for any purpose. Lethal voltages are present inside the power supply, and the AC line cord and console umbilical cord should be disconnected prior to opening the console.

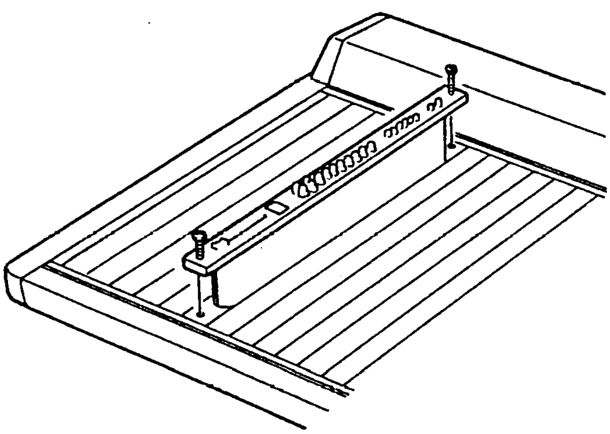
WARNING: We at Yamaha additionally caution you never to open the console and remove or install a module for the purpose of inspection, replacement or changing the preset switches unless the power has first been turned off. If a module is removed or installed with power on, the circuitry may be damaged. Unless you are a qualified service technician, do not plug in the AC cord while the interior of the power supply is exposed; dangerous voltages may exist within the chassis, and lethal shock is possible. Yamaha neither authorizes nor encourages unqualified personnel to service modules or console internal wiring. Damage to the console, the individual, and other equipment in the sound system can result from improper service or alterations, and any such work may void the warranty.

6.1 Removing and Installing A Module

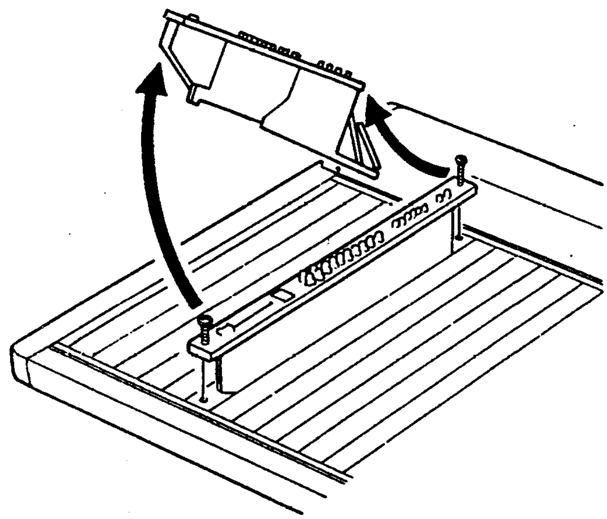
Figure 6-1. Removal of PM4000M Module



[6-1A]



[6-1B]



[6-1C]

1. Turn the Power OFF first, before removing or installing a module.
2. Loosen the screws at the top and bottom of the rear panel input/output strip corresponding to the module being removed. These screws are not retained so be sure to grasp them and set them aside for reinstallation of the module. [6-1A]
3. Loosen the retaining screws at the top and bottom of the module. These screws are retained in the module. [6-1B]
4. Lift up on the module's retaining screws (or you may also want to pull up gently on a control knob), and you will feel the two module connectors that join the connectors on the bottom of the console release. Then carefully lift the module out of the console. [6-1C]
5. Installation of a module should be done by reversing the order of this procedure. Work slowly to make sure that edge connectors mate properly.

NOTE: If you are moving a module to a different location in the mainframe, one which had housed no module or a different type of module, then you will have to also move the rear connector panel. Input modules may be placed anywhere in the frame, and you can exchange them freely. However, there should be no more than a total of 52 input channels per mainframe.

CAUTION: Do not install PM4000 Stereo Input modules in a PM4000M mainframe. We also do not recommend using PM4000 Mono Input Modules in the PM4000M mainframe since bus assignments will be incorrect (or at least, they will not make sense based on module bus identification).

6.2 Input Module Direct Out Jack: Pre-Fader/EQ or Post-Fader (switch) Pre-ON or Post-ON Switch (jumper)

A slide switch in each input module permits the Direct Out point to be altered. As shipped, the console is set so that the Direct Out point is derived ahead of the EQ and Fader (technically speaking, it comes before the VCA which is controlled by the fader). [PM4000 console users, please note that the PM4000M is factory configured with the opposite default of the PM4000 console Direct Out wiring.] If you wish the Direct Out to be Post-EQ and Fader, move the switch to the appropriate position, as illustrated (note 2 in triangle).

If you switch to Post-EQ, the "POST" direct out point comes factory set to a point ahead of the Channel ON switch, and is thus not affected by the Master Mute function. By changing an internal jumper, you can alter the Post-EQ/Fader Direct Out point to be Post-ON switch, too. The switch location is designated as item 5 (in the triangle) in the illustration below (Figure 6-2a). Also refer to figure 6-3 (next page) for the circuit location of this switch.

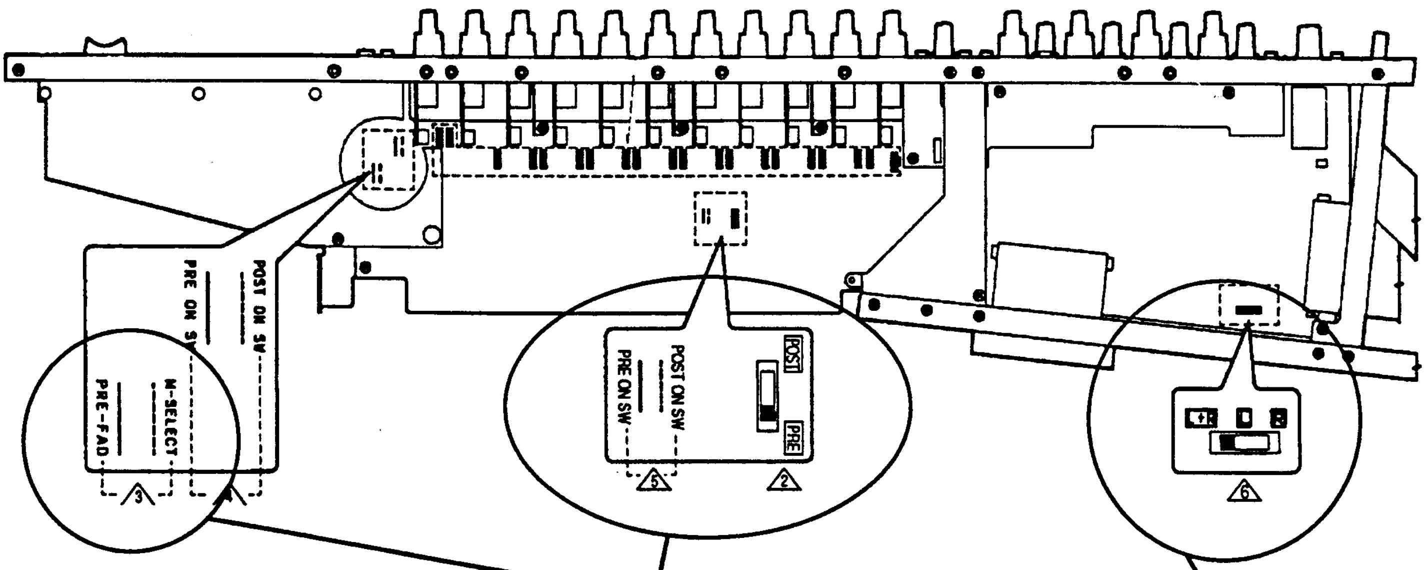


Figure 6-2b. Internal Jumper Positions for Cue Output Being Pre-Fader or Tracking the METER PRE Switch.

Figure 6-2a. Internal Switch Positions For Pre-Fader/EQ or Post-Fader/EQ Direct Out Point; Internal Jumper Positions for Direct Out "Post Fader/EQ" being Pre/Post Channel ON Switch.

Figure 6-2c. Internal Switch Positions for Cue Assigned to: L + R sides of the Cue Bus, L Only, or R Only.

6.3 Input Module Cue Outputs: Pre-Fader or to Follow the METER PRE Switch

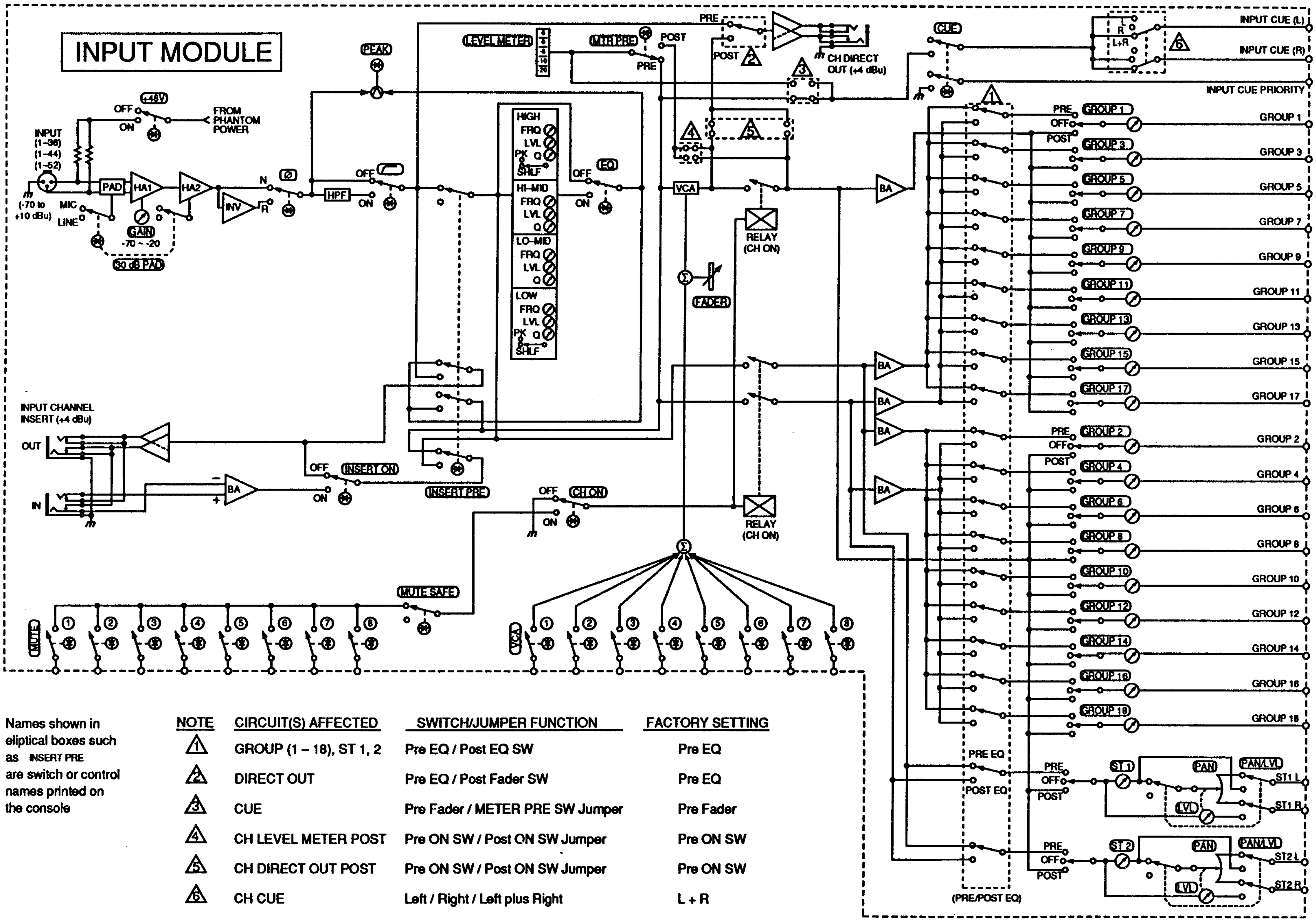
As shipped, the module's CUE outputs are derived from a point just before the fader (actually before the VCA which is controlled by the fader). However, an internal jumper can be reset so that the CUE outputs track the position of the front-panel METER PRE switch [17a]. In that case, the cue could be pre or post fader, depending on the position of METER PRE.

See Figure 6-2b above and Figure 6-3 (the block diagram). Look for note 3 in the triangles.

6.4 Input Module Cue Outputs: To L, R or L+R (switch)

Selection of Cue mode in the input module normally applies the same cue signal to both to the left side and right side of the stereo cue bus. However, a switch in each input module enables you to reassign the cue signal to either the Left or the Right side of the cue bus.

See Figure 6-2c above and Figure 6-3 (the block diagram). Look for note 6 in the triangles.



Names shown in eliptical boxes such as INSERT PRE are switch or control names printed on the console

NOTE	CIRCUIT(S) AFFECTED	SWITCH/JUMPER FUNCTION	FACTORY SETTING
1	GROUP (1 – 18), ST 1, 2	Pre EQ / Post EQ SW	Pre EQ
2	DIRECT OUT	Pre EQ / Post Fader SW	Pre EQ
3	CUE	Pre Fader / METER PRE SW Jumper	Pre Fader
4	CH LEVEL METER POST	Pre ON SW / Post ON SW Jumper	Pre ON SW
5	CH DIRECT OUT POST	Pre ON SW / Post ON SW Jumper	Pre ON SW
6	CH CUE	Left / Right / Left plus Right	L + R

Figure 6-3. Input Module Block with Location of Switches and Jumpers

6.5 Input Group & Stereo Bus Assigns: Pre Fader & EQ or Pre Fader/post EQ

Twenty slide switches in each input module permit each of the 18 Group bus assign controls and the two pair of stereo bus assign controls to be altered. As shipped, the console is wired so that if the front-panel aux PRE/OFF/POST switch is set to PRE position, the bus assign is derived ahead of the fader and equalizer (but after the high pass filter). This is useful for stage monitor work, for example, where the channel EQ may not be desirable for use in all output mixes, yet rumble-reducing filtering is desirable across all mixes derived from that input source. On the other hand, suppose that one or another Group mixes are used for pre-fader effects sends. In this case, it may be desirable to apply channel EQ to the sends. The POST position would provide EQ, but would also cause the channel fader to affect the send, which is not desirable. To solve the problem, the switch for that input's Group or Stereo mix assign can be reset so that the PRE position remains pre-fader, but is taken after the EQ. Refer to note 1 in the triangle in Figure 6-3 and 6-4a.

Figure 6-4a. Internal Switch Positions For Input Module Pre-EQ and Post-EQ Group and Stereo Mixing Bus Assignment. Slide the Switches toward the Front Panel to Select Post-EQ, or Toward the Rear of the Module for Pre-EQ.

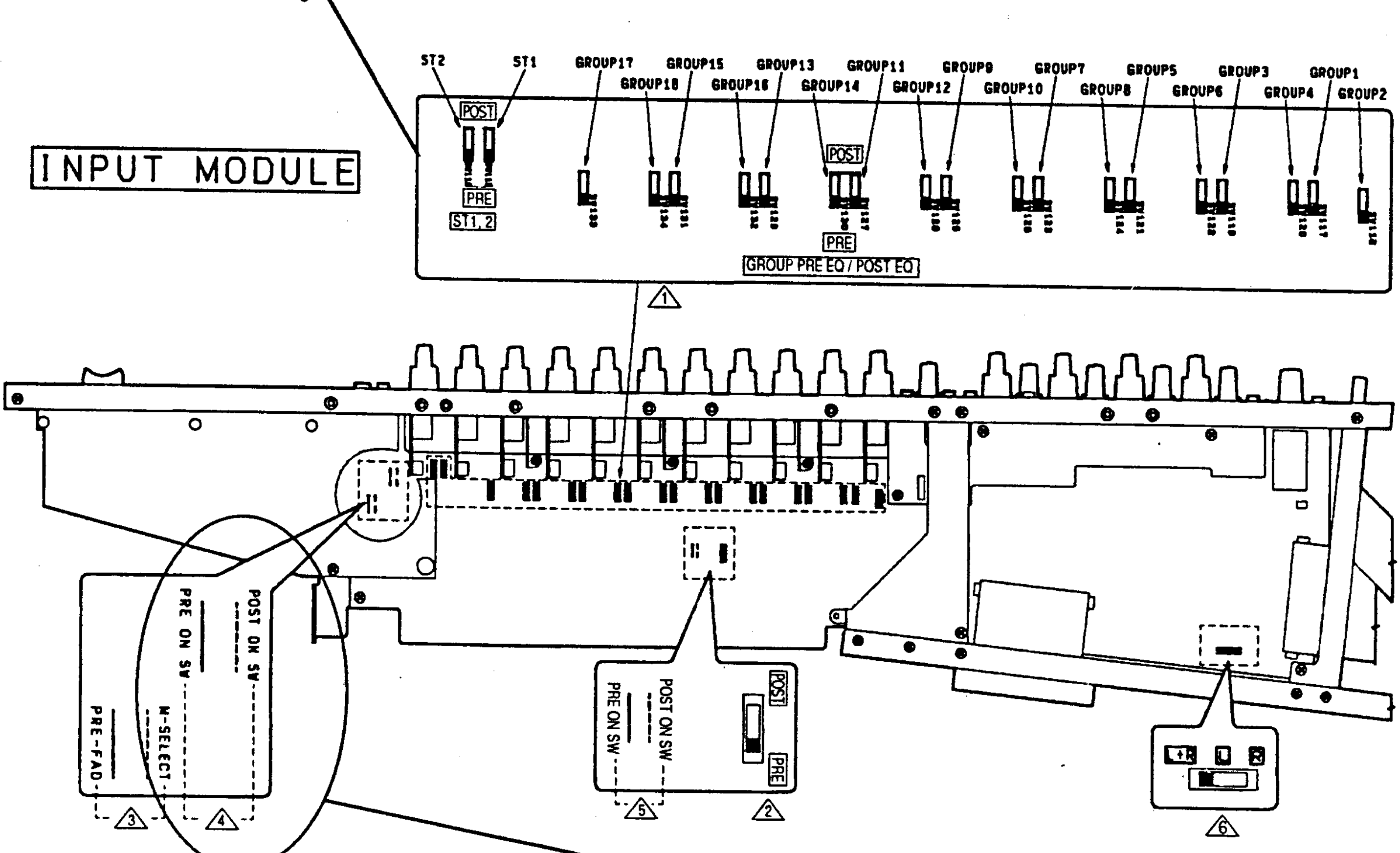


Figure 6-4b. Internal Jumper Positions For Input Module's Meter Post-Fader Source: Pre or Post Channel ON Switch.

6.6 Channel Level Meter "Post Fader" Signal Source: Pre or Post Channel ON switch (jumper)

You can select whether the channel's LED meter derives signal before (Pre) or after (Post) the channel fader by means of a front-panel METER PRE switch [17a]. If you select POST, the meter will be indicating the level after the fader (actually after the channel VCA) but before the channel ON switch. Thus, turning off the channel outputs will not affect the channel meter. If you want the meter to shut off when you turn off the channel, you can reset an internal jumper, which moves the METER PRE switch's POST source to a point after the channel VCA. Refer to note 4 in the triangle in Figure 6-3 and 6-4b.

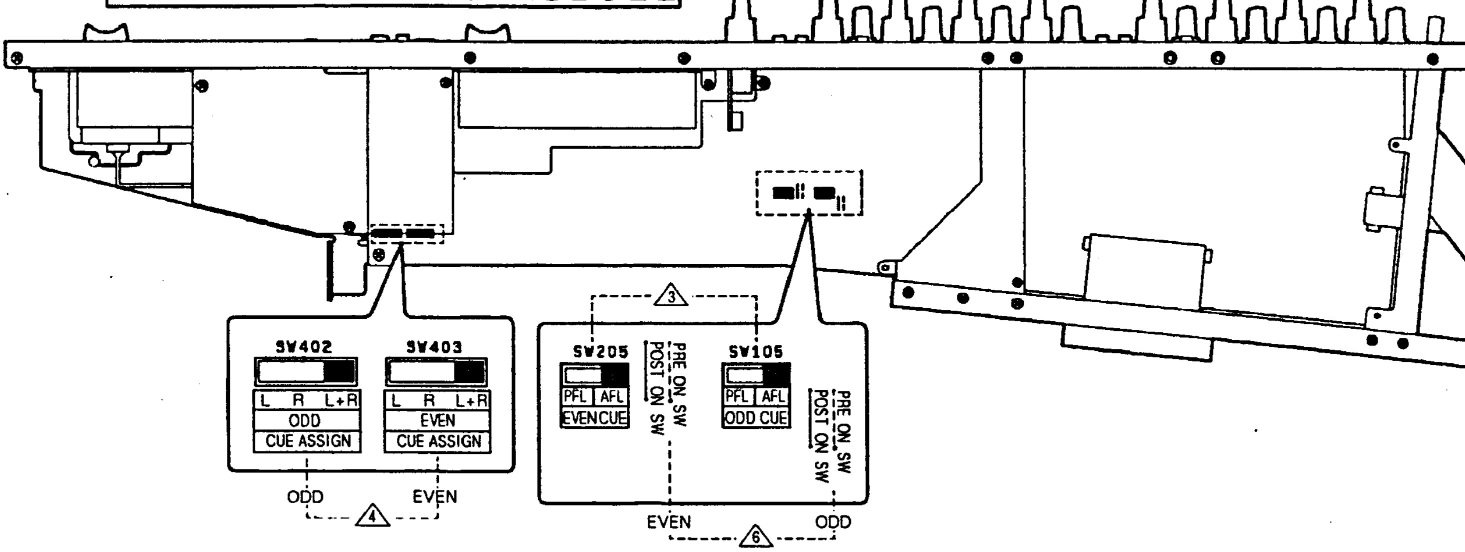
6.7 Master Modules: Cue PFL/AFL Preset Switches

Normally, when you select CUE on a Master module, the cue is derived after the bus output fader and ON/off switch (AFL=After Fader Listen). Thus, if you have pulled the fader down or turned off the bus output, you will not hear anything when you select this cue. This default mode is useful when you wish to hear the relative level of a Group mix as well as its content.

Sometimes, however, you may wish to hear the program on a bus without actually bringing up the bus output level (particularly if the bus is assgined to a stereo mix and you don't want to affect that mix's balance while you perform diagnostics or EQ touch-up – or you are previewing a mix for an upcoming theatrical cue but don't want to turn on the particular speaker feed just yet.

A pair of slide switches in each Master 1 through Master 9 module permits the module's cue pick-off points to be altered for the odd and even numbered busses controlled by that module. The internal switches in each of these modules can be repositioned so that the cue is derived before the bus fader and ON/off switch (PFL=Pre Fader Listen). See Figure 6-5a below and Figure 6-6 (the block diagram). Look for note 3 in the triangles.

GROUP MASTER1 MODULE



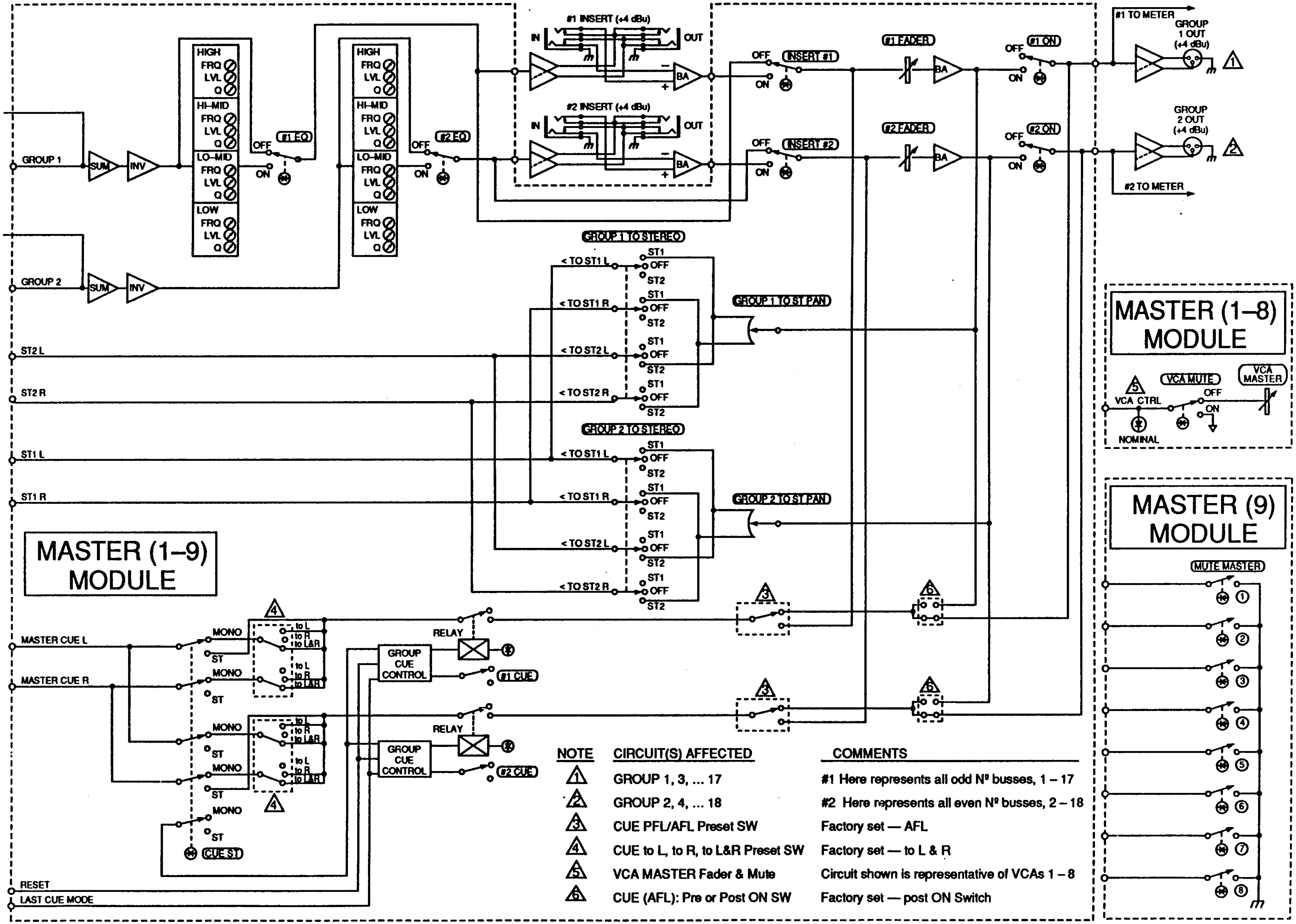
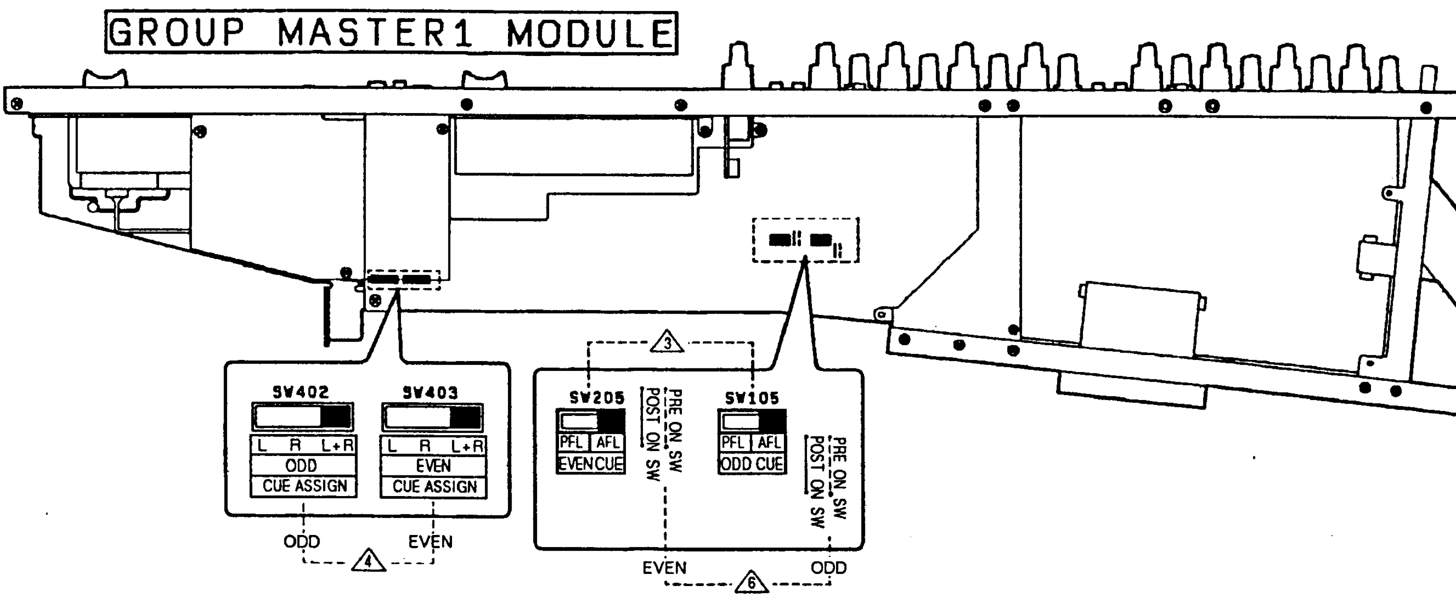


Figure 6-6. Group Master Module with Location of Switches and Jumpers.

6.9 Master Module (1–9): Cue to L, R or L&R Switches

Selection of Cue mode in the master modules normally applies odd-numbered Group Bus cues to the left side of the stereo cue bus and even-numbered Group Bus cues to the right side of the stereo cue bus. However, a pair of switches in each Master Module (1–9) enables you to reassign the cue signal to either the Left or the Right side of the cue bus.

If you utilize the pair of Group busses controlled by a given Master Module for discrete mono mixes, you may wish to use the factory preset (L + R). This will let you hear a dual-mono cue (that is, equal proportions of each Group bus assigned to both sides of the Cue bus). Alternately, you may want to reset the two switches to assign the two Groups' cue signals to just one side of the cue mix (left or right). If you are using this pair of Group busses to create a stereo mix, you may wish instead to reset the two switches inside the module to assign their respective cue feeds to the left and right sides of the cue mix. See Figure 6-7 below and Figure 6-7 (the block diagram). Look for note 4 in the triangles.



6.10 Stereo Master Module: Cue PFL/AFL Preset Switches

In Section 6.7 we discussed the function of Cue PFL/AFL switches as they affect the 18 Group Bus Feeds. The Stereo Master Module offers the same Pre/Post Fader Listen options for the Stereo 1 Bus and Stereo 2 Bus cues. The factory-shipped setting is AFL (After Fader Listen).

A pair of slide switches in the Stereo Master module permits the module's cue pick-off point to be changed to PFL (Pre Fader Listen) for the Stereo 1 or 2 bus. See Figure 6-8a below and Figure 6-9 (the module block diagram). Look for note 1 in the triangles.

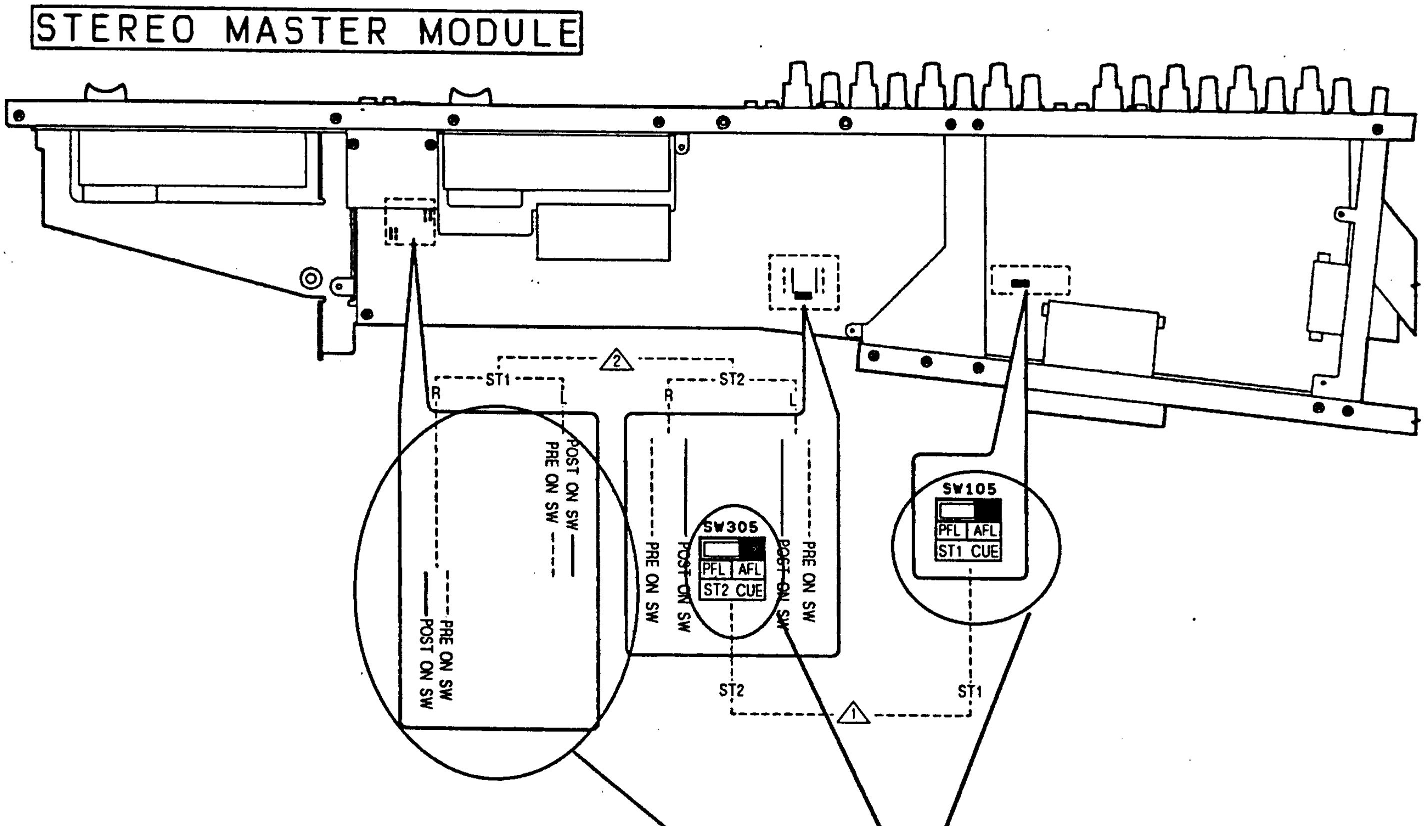


Figure 6-8b. Internal Jumper Positions For Pre- and Post- Stereo Bus ON switch Cue Feed. Only one of the two sets of L & R jumpers is circled here (for ST1). The other pair of jumpers to the right that straddle one of the PFL/AFL switches performs the same function for the ST2 bus.

Figure 6-8a. Internal Switch Positions for Pre- and Post- Stereo Master Fader Cue Feeds.

6.11 Stereo Master Module: Cue AFL Source Jumpers (Pre/Post Bus ON switch)

In Section 6.8 we discussed the function of the internal CUE Source jumpers when the internal PFL/AFL switch in in the default (AFL) position. The Stereo 1 and Stereo 2 busses work the same way, with the signal derived after the bus ON switch. If you cue the bus but have turned off the bus output, you will not hear anything. This default mode is useful when you wish to hear the relative level of a stereo mix as well as its content.

Sometimes, however, you may wish to hear the program on a bus at the set mix level, but without applying signal to the output. You can do this by resetting a jumper for that bus in this module. See Figure 6-8b above and Figure 6-9 (the block diagram). Look for note 2 in the triangles.

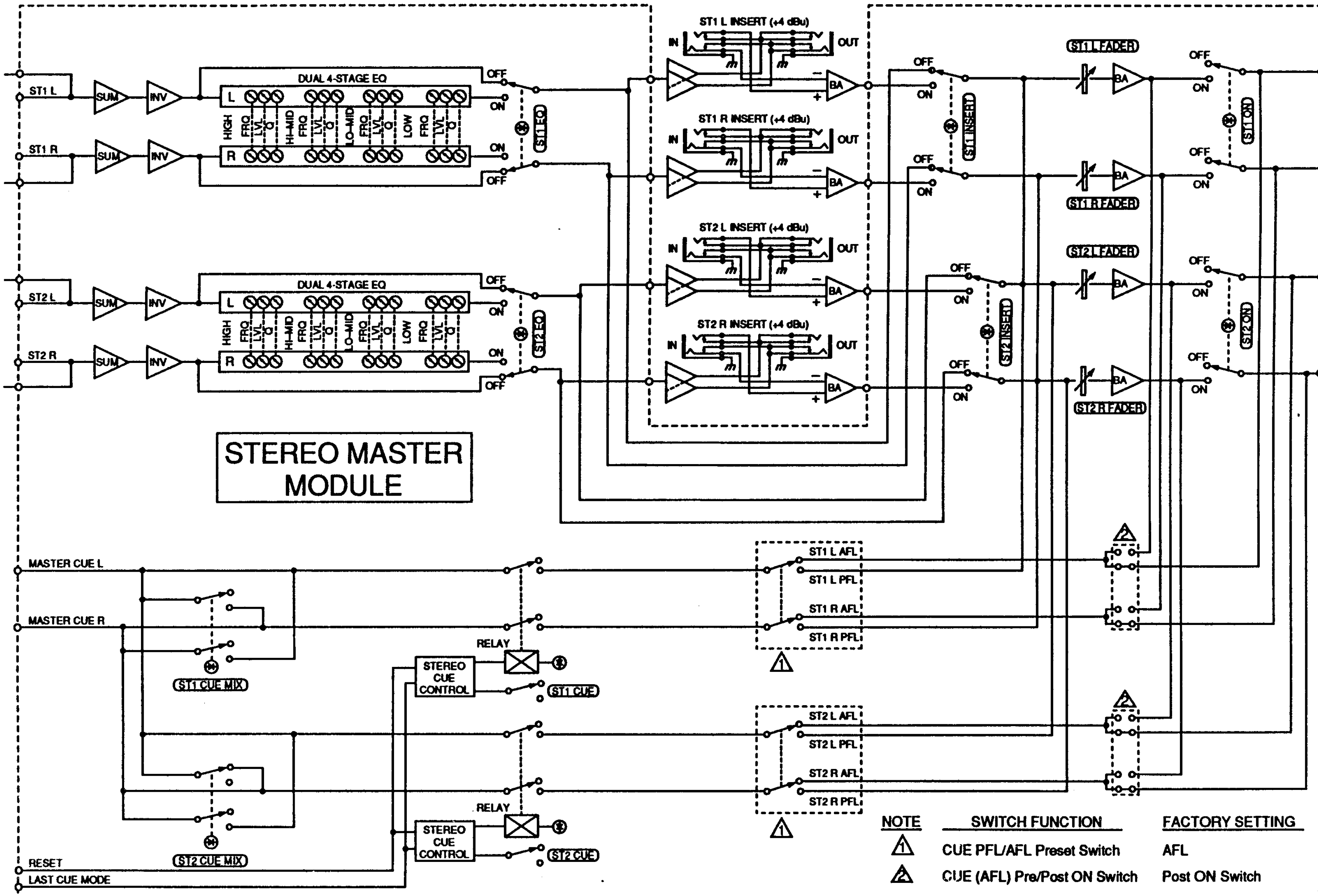


Figure 6-9. Stereo Master Module Block Diagram with Location of Switches and Jumpers.

6.12 Installation of Optional Input Transformers

The PM4000M standard input module is equipped with a balanced, differential input preamplifier for the XLR connector. That preamp, along with some circuitry for the resistive attenuation pads, is located on a small printed circuit board that "piggy back" mounts to the module's main circuit board. Refer to Figure 6-10A.

An optional transformer balancing option may be installed by a Yamaha PM4000M dealer or a qualified electronic service technician. The modification kit contains a replacement circuit board for the original differential preamplifier, and a separate input transformer. In order to install the kit, the following steps must be performed.

1. Shut off the power to the console.
2. Remove any input module(s) to be equipped with transformers.
3. Install the transformer into the included fitting with the nut as shown in Figure 6-10B.
4. Being careful with the wiring, unfasten Angle bracket H of the module by removing the two small flat-head screws and the two small bind screws.
5. From the inside of Angle H, insert the two small M3 screws provided with the transformer kit, and attach the transformer fitting (Figure 6-10C).
6. Restore Angle H to its original position.
7. Pass all the wiring through the slit in Angle R.
8. Solder the transformer wiring to the new input transformer board (Figure 6-10D).
9. Remove the present input circuit board (transformer area) and replace it with the new transformer board.
10. Reinstall the input module into the console mainframe. When doing so, be sure that the new transformer wiring does not protrude from the module. Damage could otherwise occur during module insertion or subsequent removal.

The above completes the procedure for intallation of an input transformer. Check the Fader and PAD signals to verify the installation.

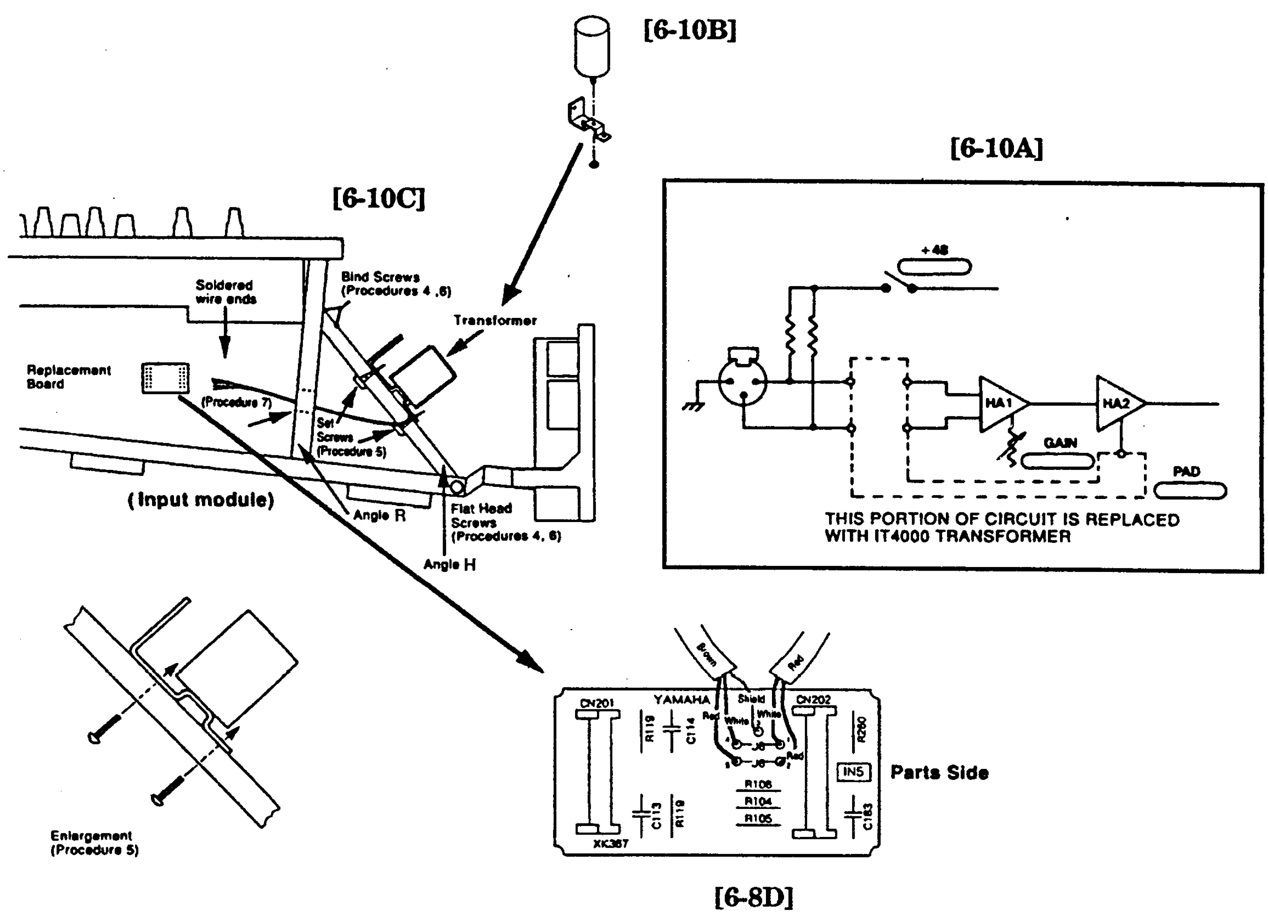


Figure 6-10. Optional Input Transformer Installation

6.13 Hints on Circuitry For Remote Control of the VCA Masters and Mute Groups

The VCA/MUTE CONTROL connector on the PM4000M rear panel is provided primarily so that two consoles may be linked, and just one console's VCA MASTER FADERS and/or MUTE MASTER switches will affect both consoles input channels. However, it is possible to create an independent controller so that these functions can be remoted from the console. One possible application would be to remotely adjust mix levels in the middle of a venue even though the console is located in a booth. Another possible application would be the creation of a limited automation system. Yamaha does not offer detailed instructions for this type of remote control. However, we do present here a schematic diagram of the VCA control fader circuit which, if constructed externally by a competent technician and interfaced via the VCA/MUTE CONTROL connector, can do the job.

Note that the nominal fader position delivers 0 VDC to the VCA, and the VCA operates at unity gain with that input. The control voltage scaling is approximately -20 dB per volt DC in the linear range of fader travel (above -50 dB on the fader scale). Thus, at maximum upward fader travel, a single fader will deliver about 0.5 volt negative, which drives the VCA to +10 dB of gain. If several VCA faders are set above nominal and assigned to a channel, the maximum negative voltage that will be applied to the VCA is -1.2 VDC (a DC limiter circuit prevents any more negative voltage from being passed and turns on the VCA MAX LED). This corresponds to +24 dB of gain. At minimum VCA fader setting, the output is +10 VDC, corresponding to over 100 dB of attenuation.

The VCA and MUTE connections are illustrated in Figure 2-13. In order to mute a group, ground the conductor corresponding to that group. The console's VCA MASTER/SLAVE and/or MUTE MASTER/SLAVE switch(es) must be set to the SLAVE position in order for the corresponding remote control to take effect on the designated busses and mute groups.

WARNING: Only qualified service technicians should attempt to construct and connect any circuit to interface with the PM4000M VCA/MUTE CONTROL connector. A circuit or wiring error could severely damage the console, and such damage is not covered under the terms of the PM4000M Warranty. Improper grounding could also create noise and/or safety hazards. This information is provided only to illustrate the extent of such a modification; the PM4000M Service Manual should be consulted before actually building any remote control device.

Refer to the parts list and the VCA control voltage curve on the following page.

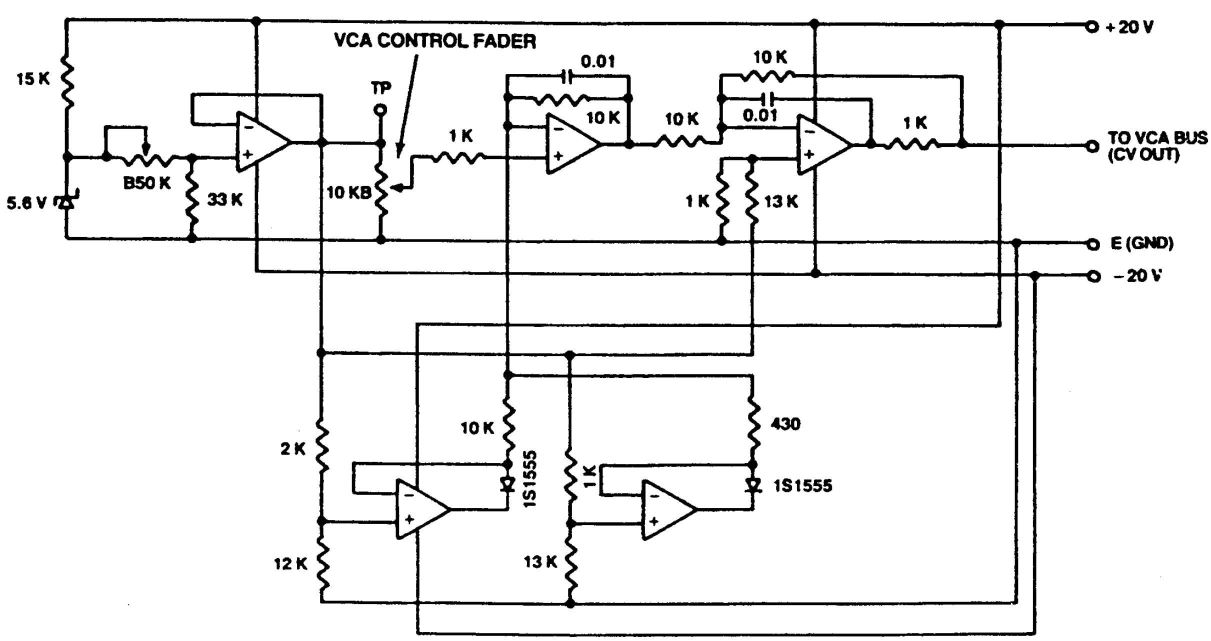
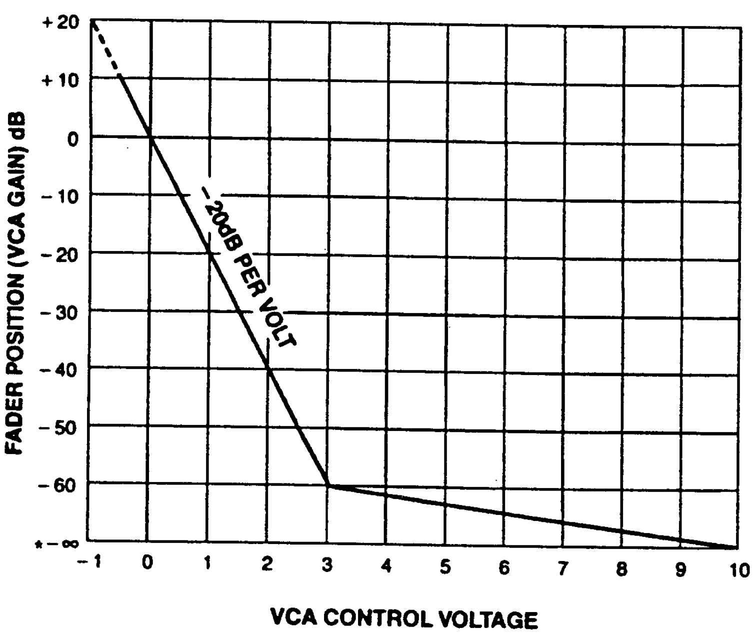


Figure 6-11. Suggested Circuit for Remote Control of a VCA Master Group

YAMAHA PART #	QUAN	SUFFIX LETTER	ITEM	VALUE OR TYPE
UA21410	2	K	MYLAR CAPACITOR	0.01 uF, 50 V
HU07543	1	F	METALIZED FILM RESISTOR	430 ohm, ¼ W
HU07610	4	F	METALIZED FILM RESISTOR	1 kohm, ¼ W
HU07620	1	F	METALIZED FILM RESISTOR	2 kohm, ¼ W
HU07710	4	F	METALIZED FILM RESISTOR	10 kohm, ¼ W
HU07712	1	F	METALIZED FILM RESISTOR	12 kohm, ¼ W
HU07713	2	F	METALIZED FILM RESISTOR	13 kohm, ¼ W
HK05715	1	J	CARBON RESISTOR	15 kohm, ¼ W
HK05733	1	J	CARBON RESISTOR	33 kohm, ¼ W
IG06920	3		IC AMP	MJM2041DD
HT56009	1	B	SEMI-FIXED VR (TRIMMER)	50 kohm
IF00004	2		DIODE	1S1555
IF00214	1		ZENER DIODE	RD5.6ED2
VA25610	1	B	SLIDER VR (FADER)	10 kohm

Table 6-1. Parts List for Making Remote VCA Control Circuit



CHANNEL ON relay opens when fader is at −∞ position.

Figure 6-12. VCA Control Voltage versus Fader Position

Section 7.
Operating Notes and Hints

This section is not meant to be comprehensive. Instead, it focuses on a few areas which we feel require special attention, or where a better understanding of the function can lead to far more utility or better sound quality from the PM4000M.

7.1 Console Gain Structure

In the GAIN STRUCTURE AND LEVELS section of this manual, we discuss some general considerations regarding levels and system setup. What of the proper gain structure within the PM4000M? How can the many faders and other level controls that affect a given signal all be adjusted for the optimum results? These are important questions to ponder, and we hope you will take some time to study the possibilities.

7.1.1 What Is The Proper Gain Structure?

Let's begin with the XLR channel input to the console. According to the INPUT CHARACTERISTICS chart in the SPECIFICATIONS section, the nominal input level ranges from -70 dBu (0.25 mV) to +10 dBu (2.4 V). These are the levels that will supply the ideal signal level throughout the module with the PAD set to 0 dB or -30 dB, the input GAIN control as required, fader set to its nominal position, and no VCA groups assigned. Actually, a wider range of levels can be accommodated if the fader is adjusted to other-than-nominal position; from -90 dBu (0.025 mV) minimum to +24 dBu (12.3V) maximum.

What is the correct gain structure? Simply stated, it is the level at which there remains adequate headroom so that peaks can be accommodated without clipping, while at the same time there is sufficient "distance" above the noise floor that noise does not become objectionable. If a signal is too high in level (too "hot") at a given point in the console, then peaks or, in the extreme, the entire signal, will be subject to distortion. If the signal is too low in level, there may be considerably more headroom and less risk of distortion, but the noise will be that much more noticeable, and quiet passages may be masked entirely by residual noise. The "ideal" level, then, where headroom and noise tradeoffs are optimum, is also known as the nominal level. There is no single value for the correct nominal level; it varies throughout the console. This is what the middle graph line in the GAIN STRUCTURE chart in Figure 3-xx depicts. The top graph line indicates the clipping point. The distance between these two lines, at any point along the horizontal signal flow scale, depicts the available headroom. It is important that wide headroom be available throughout a console, not just at the input and output; otherwise multiple signals applied to the busses may add together such that the mixed level approaches clipping, even though the individual feeds to the mix are within their acceptable nominal range. Sometimes a group or master fader can be adjusted to correct this condition, other times it cannot because the distortion is occurring in an amplifier ahead of the fader, and the only cure is to lower the signal levels applied to the bus. How can one know the best course of action when distortion, or excess noise, is encountered?

7.1.2 What Affects Gain Structure?

First, understand that signal levels can be increased by either increasing amplifier gain (including EQ boost), reducing the amount of attenuation, or adding multiple signals together. Similarly, signal levels can be reduced by either decreasing amplifier gain (including EQ cut), increasing the amount of attenuation (including filter roll-off), or splitting the signal to feed two or more circuits. With this in mind, it becomes clear that the mere act of feeding the "correct" nominal level signal into a console is no guarantee that it will remain at an acceptable level throughout the console.

7.1.3 Establishing The Correct Input Channel Settings

In the case of the PM4000M, the input channel meter LEDs [17B] make it relatively simple to obtain the correct gain structure at the input stage. Begin with the PAD set at maximum attenuation (-30 dB), the GAIN control centered, and apply the typical input signal to the channel input. If none of the meter LEDs are illuminated, or perhaps just the -20 LED, disengage the attenuation PAD switch to remove the 30 dB of attenuation. Adjust the GAIN control as required so that the red PEAK LED flashes on only occasionally, during the loudest program peaks, and the 0 LED flashes frequently or remains on. This establishes the correct channel sensitivity for the initial setup (you may wish to alter these values during an actual program mix, as explained in subsequent paragraphs).

NOTE: It is a good idea to set the Group Master and Stereo Master Faders at a very low level during the initial stages of setup. This will prevent uncomfortable or even dangerously loud signals from reaching the outputs while preliminary mix setup is established.

Given the correct GAIN and PAD settings, adjust the channel Fader to its nominal (0 dB) setting. This setting provides the best range of control, with some boost available if the signal must be raised in the mix, and plenty of resolution for fading the signal down in the mix.

Now the channel HP Filter and EQ can be set as desired. If a particular EQ setting causes the channel's PEAK LED to flash on more than occasionally, then the boost applied is raising the signal level too high. The solution is to either reduce the EQ boost setting in one or more bands, or to leave the EQ where you have it for the proper signal contour, and to instead reduce the signal level going into the equalizer. You must do this by adjusting the GAIN control (and, in some cases, also engaging the PAD); the Fader does not affect signal going into the EQ. Lower the GAIN only enough so that the PEAK LED does not flash on excessively.

The signal now may be assigned to any of the 18 group mixing busses or the two stereo busses. If an assign control is set to PRE-fader position, then the signal level applied to that bus will remain constant regardless of adjustments to the channel Fader, depending instead only on the bus assign control setting. In POST-fader position, the assign level will be determined by both the channel's bus assign control and the channel Fader.

This same procedure should now be followed for each input channel. Once this is done, the bus levels can be examined. Set the VU meter assign switches [73] to look at the Stereo Bus levels and, where applicable, to the GROUP OUT levels (you can see GROUP OUT levels all the time, with no switching, on the 44 and 52 input channel mainframes; only 36-channel mainframes not sold in the U.S.A may require switching to see a Group level). One bus at a time, monitor the group mix (use the headphones and the corresponding group CUE switch), and create a rough mix of all input channels which feed this group. Bring down the input Faders (or individual input channel-to-bus assign controls) for those sources which are too prominent in the mix; avoid raising input Faders to make other sources more prominent. Once this rough mix is established, raise the corresponding Group Master Fader to the nominal position (0 dB on the scale, NOMINAL LED illuminated); the rectangular LED at the nominal position will be illuminated when the VCA is at actual nominal position. If the signal level on any of these busses becomes too hot (red meter LED flashing on more than occasionally or VU meter pegged at the top of the scale), do not back off the Group Master Fader. Instead, pull down all the input channel Faders (or turn down all input channel-to-bus assign controls) which feed this Group by an equal amount. (If the channels also happen to be assigned to a given VCA Master, you can pull down that VCA Master, which, in turn, will reduce the signals applied to the group bus). This will leave the Group Master Fader at the desired nominal position, will preserve the desired balance between input channels, and will keep the bus level from being too hot. Finally, release the Group CUE switch.

This same procedure applies to setting the Stereo 1 and Stereo 2 Master levels where input channels are assigned directly to the stereo bus. However, you will have to adjust the Group Master levels on any groups which are assigned to the stereo bus(ses).

7.1.4 Establishing The Correct Group Master Settings

Follow the same procedure for each of the other Group Masters. Once all Group Masters are calibrated in this manner, the Stereo mix can be similarly calibrated. Any Group outputs which are to be applied to the stereo mix should be so assigned. Any input channels which are to be applied directly to the stereo mix should be so assigned. Monitor each stereo mix by engaging the respective Stereo 1 or 2 CUE switch, and adjust the various stereo PAN pots as desired.

If you're not sure about the stereo position of a given group mix in the stereo perspective, you can temporarily cue that group by pressing its CUE switch. Assuming the bus' master is set (via an internal switch on the master module) for discrete feed to the left or right side of the stereo cue mix, you will hear only that source in the left or right side of the stereo field. If the internal switch for that bus is set for L+R (the factory default setting), you will hear that bus equally in both sides of the cue mix.

If you want to cue several group busses together, make sure LAST CUE mode is not selected (i.e., switch [70] is not illuminated).

With the various signals applied to the Stereo 1 or Stereo 2 mix, bring up the corresponding Stereo Master Faders to nominal position and check the bus levels on the bus' L and R VU meters; if they are too high, you can lower all Group Master Faders (if the Group-to-Stereo switches are engaged, or lower the input channel Faders or channel-to-stereo assign controls (if the input channels' direct-to-stereo assign capability is in use). Lower all the affected source faders by a similar amount so as to preserve the mix balance.

7.1.5 How VCA Control Affects Gain Structure

Use of the VCA Master fader can complicate the gain structure considerably. It is important to set up the input PAD switch and GAIN controls using the technique previously described, including any level compensation for EQ boost. The channel Faders initially should be set at nominal position, and any VCA Masters to which the input channel is assigned should be set at nominal position as well. When all VCA Masters are at their nominal position (green "NOMINAL" pointer illuminated), the gain structure can be approached pretty much as outlined previously. If, however, a given input channel is assigned so that it is affected by several VCA Masters, and any of those VCA Masters is raised in level, then the input channel Fader levels is effectively increased. If enough VCA Masters are raised to the point where input channel VCA gain can go no higher (as indicated when the red VCA MAX LED turns on), then the offending VCA Masters should be lowered slightly to correct the situation, or the channel Fader should be lowered. If the adjustments adversely affect the balance between VCA groups, all VCA Masters then can be lowered, or the input Faders of the other channels can be lowered somewhat.

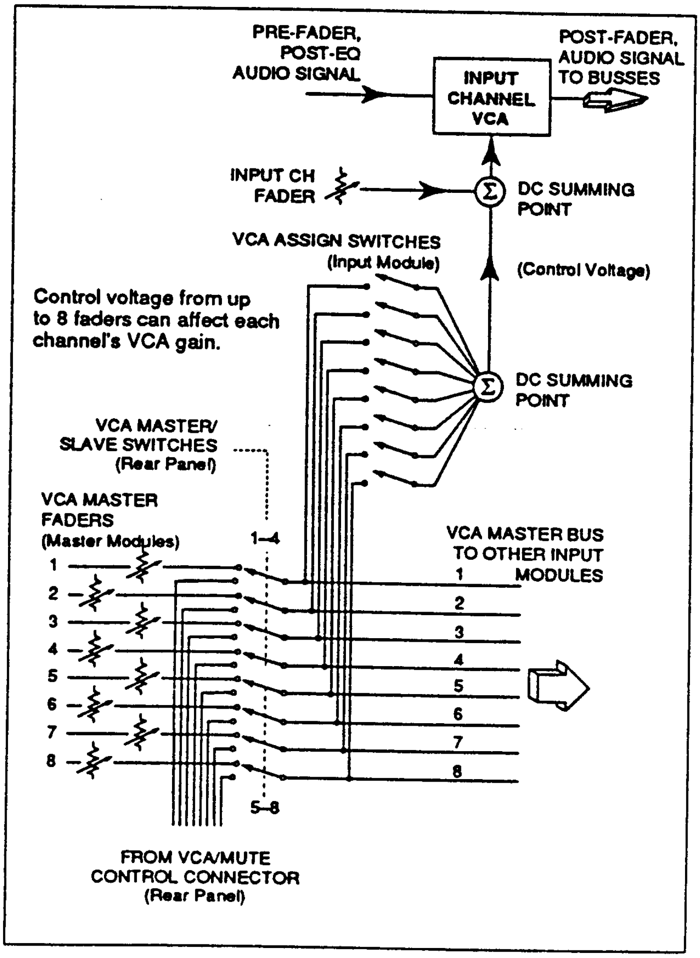


Figure 7-1. Control Voltages From up to 9 Different Points (the Channel Fader Plus 8 VCA Master Faders) Can Affect any Channel's VCA Gain

CAUTION: If you assign or unassign an input channel to a VCA Master group during a performance, the channel gain will jump up or down unless the corresponding VCA Master Fader is set precisely to the nominal position (green LED "NOMINAL" indicator illuminated).

7.1.6 Channel Muting and Gain Structure

As pointed out earlier, adding inputs to a mix will increase mix levels. If optimum mix levels are established with some input channels muted, and those channels are later turned on (either with the channel ON/off switch or with the channel MUTE and MASTER MUTE switches), then the bus levels may increase unacceptably, and all input channels' levels applied to the offending bus or busses may have to be reduced. Similarly, if some Groups are added to the Stereo Master mix after those gains have been calibrated, then Stereo bus levels may increase unacceptably, requiring either a reduction in all Group Master levels or minor adjustments of the Stereo Master Fader(s).

7.2 Further Hints & Conceptual Notes

7.2.1 What Is a VCA, and Why Is It Used?

A VCA, or Voltage Controlled Amplifier, is a special type of amplifier whose gain (the amount of amplification) is adjustable by means of an externally applied DC voltage. This is in contrast to a conventional amplifier, whose effective gain may be adjusted by means of altering a feedback resistance or by attenuating audio signal before or after the amplifier.

In a conventional console, mixer or other audio processor, a channel fader (or level control) is generally a variable resistor which attenuates the audio signal flowing through it. The Fader is usually preceded a buffer stage and followed by a booster stage, both of which are fixed gain amplifiers. The buffer keeps the fader's changing resistance from loading the input preamplifier, and the booster stage makes up for the fixed insertion loss of the fader resistance when the fader is set to its nominal position (typically 6 dB). The signal then may be routed to a submaster (Group Master) Fader, where it is again subject to insertion loss so that some gain must be "made up" by an additional booster amplifier stage. If the signal path becomes complex, with one or more levels of "submaster" control, more noise and distortion can result due to thermal resistor noise and residual amplifier aberrations. Also, because the audio signal must be physically routed over a longer, more involved path, there is more opportunity for crosstalk, electrostatically or electromagnetically induced noise, and further signal quality degradation.

An alternate approach involves the use of a VCA. In the PM4000M, there is one VCA in each input module. That VCA takes the place of the post-Fader booster amplifier in a conventional console configuration. The PM4000M channel Fader is a variable resistor, but it does not have audio flowing through it. Instead, it adjusts a DC voltage output (from 0 volts at nominal position, to -0.5 volts at maximum gain, to +10 volts at "infinite" attenuation position). The DC output voltage from the channel Fader is applied to the channel's VCA control input.

The VCA is a special amplifier that is designed to operate at unity gain when the fader is at nominal position, can provide some gain with the channel and/or VCA Master Faders set above nominal, but primarily is designed to attenuate the signal as the fader is lowered. (You can think of VCA as Voltage Controlled Attenuator, although technically that is a distinctly different device.) So far, there is no big advantage to this VCA approach over the conventional console, where the audio flows through the channel fader.

The VCA's advantage is realized when grouping is used. The VCA Master Faders are really just like the channel faders in that they output a DC voltage. When one or more input channel VCA Assign switches are engaged, the voltage(s) output from the corresponding VCA Master Fader(s) combine with the channel fader output voltage, and the sum of these voltages determine the channel's VCA gain. The audio signal does not actually flow through any VCA Master Fader, and no matter how many VCA Masters affect the channel, the audio path remains the same... simple and direct with no added noise, distortion or crosstalk.

For reasons described in Section 7.2.2, conventional group master Faders are also provided in the PM4000M.

7.2.2 The Distinction Between The Group Busses and The VCA Master "Groups"

The PM4000M affords the operator with two different means to control multiple input channels from a single fader. One approach is to assign multiple inputs to a given Group with the Input-to-Group Assign controls [14], and to then use the Group Master Fader [34] to control those signals. With this approach, the actual audio output signal from each of the assigned input channels is applied to a bus wire via 18K ohm summing/isolation resistors. The signal on the group bus is then fed into a combining (summing) amplifier in the Master module, is routed through the GROUP INSERT IN/OUT jacks [90], is then controlled by the Group Master Fader, and is fed to GROUP OUT [100] and any other post-Group Master Fader circuits.

An alternate approach to control multiple input channels from a single fader is to use the VCA system. The audio signal in each input channel does not actually pass through the channel Fader [23]. Instead, that fader applies a DC control voltage to a VCA (Voltage Controlled Amplifier) in the input module. The audio signal flowing through that VCA is, in turn, increased or decreased in level according to the control voltage applied to the VCA. One advantage of the VCA is that the control voltage applied to it can come from more than one point. In fact, when one or more of the input channel's VCA ASSIGN switches [20] is engaged, control voltage from the correspondingly numbered VCA Master Faders [39] is also applied to the channel VCA. The circuitry is such that the VCA Master will cause the assigned input channel(s) post-fader output levels to ride up and down, scaled to the channel Fader setting. Of course, the channel(s) output signal must still be assigned somewhere.

NOTE: It may not be obvious, but VCA master faders *and VCA assign switches have nothing at all to do with where the audio signal goes. They only affect its level. The signal must be assigned via the bus assign controls.*

If the signal on several channels is assigned directly to the stereo bus using the channels' Stereo assign controls [15] [16], then the VCA Master to which those channels are assigned will act like a Group-to-Stereo fader. If the channels' output is assigned to a Group bus using a Group assign control [14], then the VCA Master [39] to which those channels are assigned will control the level applied to the Group Master [34], which is somewhat redundant but does serve some useful purposes.

What cannot be done with a Group Master Fader [34] that can be done with a VCA Master [39] is controlling the post-fader Group Assign levels from groups of input channels. While it's true that Group Master Faders [34] affect the overall bus output level on the 18 Group busses, each of these busses can be considered a discrete output. Of the many input channel controls that may be feeding a given Group Master Fader, some can be controlled by one VCA Master, and others by another VCA Master. Thus, when "subgrouping" is accomplished with the VCA Master Faders, the output of affected input channels is controlled more completely. That is, the channels' Group and Stereo outputs are all affected by the assigned VCA Master(s).

What cannot be done with a VCA Master Fader [39] that can be done with a Group Master Fader [34] is the processing of a single, mixed signal. Consider, for example, that a given group of signals must be compressed... say the backup vocal mics or the drummer's monitor feed. If the several input channels which accommodate backup vocals are all assigned to a single Group Master Fader, then one compressor/limiter can be inserted in the Group INSERT IN/OUT patch point [90], affecting the mixed signal on that group mixing bus. On the other hand, if those same input channels were instead controlled as a "group" by a VCA Master Fader, and the channel outputs were assigned to various group mixing busses, then it would be impossible to compress the backup vocal mix. Instead, multiple compressor/limiters would have to be inserted in the individual channel INSERT IN/OUT patch points [82]. The latter approach is more costly, and also applies the effect to all the channel's outputs, rather than just to a specific group.

VCA Master Fader grouping is often useful for control of scenes, songs or sets, whereas conventional Group Master Faders are often useful for control of related groups of mics and instruments. For example, one VCA Master might be assigned to control all drum microphones. Another VCA Master might also be assigned to the same drum microphones, plus any percussion and guitar mics. One VCA Master would then affect drum levels, while the other would affect the entire rhythm section.

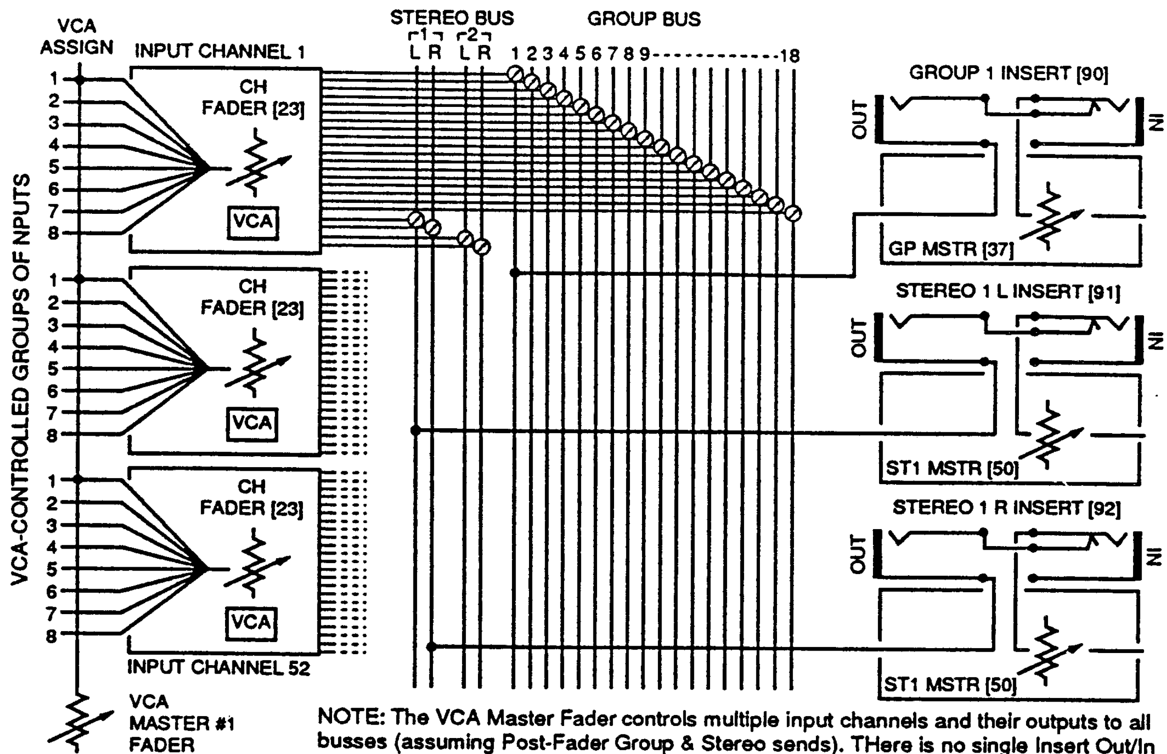
In some cases, multiple channels that are assigned direct to the stereo bus can be controlled in groups by the VCA Masters, while other channels can be assigned to different Group Master Faders, and the Group Masters, in turn, can be assigned to stereo; using this approach, one has the equivalent of a maximum of 19 groups mixed to stereo (9 pair of Group Busses, Two Stereo Busses, and 8 VCA Groups).

There is one further distinction between VCA groups and conventional groups. If one were to use conventional groups to control scenes, sets or songs, a given input channel might well be assigned to several group mixing busses. The Group-to-Stereo assign function would then be used to combine those Group busses to stereo mixes, with the group master faders serving as scene controllers. If, in this instance, two Group Master Faders were raised to nominal position, and the same input channel was assigned to both of those groups, that channel's level could rise 3 dB in the combined stereo output, throwing it out of balance with other single-assigned channels. This is because that channel signal is being added together twice in the stereo mix.

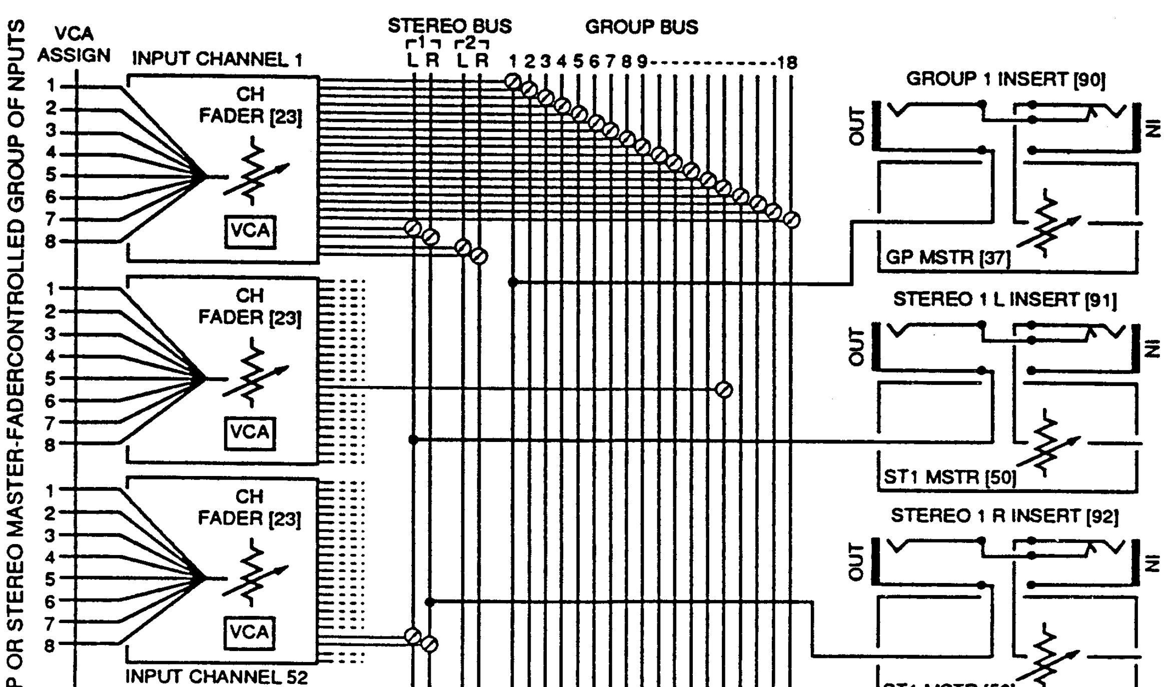
If instead of using conventional Group Master Faders, VCA Master Faders were used to control the scenes, and one input was assigned to two (or more) VCA Masters, the above level "build up" would not occur, and the correct balance would be retained. That's because when VCA Master Faders are set to nominal position, they output zero volts... which means they don't change the level coming from the input channel. Whether one, two or all eight VCA Master Faders are assigned to a given input channel, the channel's output level will not change so long as the VCA Masters are at nominal.

On the other hand, if one "pulls down" the conventional Group Master Fader in the first example above, the level of the double-assigned input will only drop 3 dB, whereas pulling down a VCA Master Fader will completely kill any input channel assigned to that VCA group.

Ultimately, the selection of VCA or conventional Group Master Fader assignments should be dictated by the specific requirements of the application.



NOTE: The VCA Master Fader controls multiple input channels and their outputs to all busses (assuming Post-Fader Group & Stereo sends). THere is no single Insert Out/In point that can process this VCA-controlled group of inputs, however.



NOTE: Channels and outputs are selected at random in these illustrations. While every input channel has controls for routing to all Group and Stereo busses, only some are turned up. The Group 1 Master Fader controls the Post-Input Fader signals from all these input channels. Similarly, the Stereo 1 Master Fader adjusts the stereo 1 output contribution of all these input channels. In this way, a single effects unit or dynamic processor can process the grouped input signals if it is placed in the Group Master Insert [90] or Stereo Master Insert Out/In [91] [92] point.

Figure 7-2. Signal Processing of The Mixed Program Is a Major Difference Between The VCA-controlled "Groups" and The Conventional Group Masters

7.2.3 Using The Channel Insert In Jack as a Line Input

The input channel INSERT IN jacks [82] are electronically balanced, line level inputs that come after the channel PAD switch and GAIN control. These jacks may be used to accommodate any balanced or unbalanced +4 dBu nominal line input source. Why would one want to use the 1/4" phone jack INSERT IN rather than the XLR channel input? There are several possibilities. Certainly, the most obvious is that if the input source is equipped with a +4 dBu phone jack output, then the INSERT IN jack enables a standard phone plug-to-phone plug cable to be used without any adaptor. However, the INSERT IN jack also can save time.

If the PM4000M is being used for theatre production or TV production, then CD or tape machine returns (playback from the CD machine or tape recorder) can be plugged into the INSERT IN jacks, while microphones or other line level sources can be plugged into the channel XLRs. When recording the tracks or using live mics for other purposes, the channels' PAD switches and GAIN controls can be set, as needed, for the various input sources. When playing back a CD or tape, the PAD switches and GAIN controls need not be readjusted; instead, simply engage the channel INSERT ON switches [13] to select the recorded material. The same concept applies where the console is used for multiple stage setups (as in subsequent scenes in a theatrical presentation, or different sets for a live musical show). Provided one of the sources is a +4 dBu line level source, it can be connected to the INSERT IN, and the other mic or line level source can be connected to the channel XLR; the INSERT switch then permits instantaneous selection of one or the other input source without need to disconnect and connect cables.

NOTE: The INSERT IN / OUT point on is after the channel EQ unless the INSERT PRE switch is engaged [12].

7.2.4 Understanding and Use of The Master Mute Function.

Each input channel is provided with eight MUTE Assign switches [21]. When one of these switches is engaged on a given input channel, that channel becomes subject to control by the correspondingly numbered MUTE MASTER switch [41]. Specifically, when the MUTE MASTER switch is engaged, then the assigned input channel(s) turn Off (assuming they had been turned On in the first place). What this means is that any assortment of input channels can be pre-set to turn off when one or more of the MUTE MASTER switches is engaged (or to turn on when the MUTE MASTER switch is released). This is useful in just about every conceivable application.

In a concert, an entire group of mics can be muted when the instruments and/or vocalists are not using them. The input channel faders and other mix controls can all be left at their previously established settings, and only one MUTE MASTER switch need be engaged to keep these mics (or line level sources) from contributing to the console output. Then, at the precisely required moment, that group of channels can be brought into the mix "on cue" by releasing the MUTE MASTER switch.

For a theatrical presentation, different scenes can be un-muted as required, keeping the number of open mics at a minimum, which reduces the natural tendency for feedback with distant mics in a live sound reinforcement system. For TV or radio production, a group of inputs which are primarily used for solo performances can be kept muted until the moment they are needed, thus minimizing noise. For a church, the choir mics can be kept muted until the moment the choir is called upon, thus reducing noise, the "hollow" sound from those open mics, and removing the extra stress on the choir members of having to keep absolutely still during the entire service. These are but a few of the ways that the PM4000M's ability to mute overlapping groups of input channels can be used to advantage.

NOTE: Although a similar function could be achieved by using the Group ON / off switches, the functions really are quite different. Consider that the MUTE MASTER switch kills all the output of the channels, including the direct-to-stereo bus feed, whereas each Group ON / off switch kills only one group output. Also, consider that some input channels feeding a given group can be killed with one MUTE MASTER, while other input channels may continue feeding that same group output. Thus, the mute function is distinctly different than the Group Output or Stereo output ON / off switches.

Things can become more complex when an input channel is assigned to more than one MUTE MASTER switch. In this case, the mere act of releasing one MUTE MASTER may not turn on the channel... if the channel is still being muted by so much as one other assigned MUTE MASTER. Should the need arise to turn on a particular input channel without unmuting other channels, and you don't want to disturb the previously assigned MUTE switches, you can override the entire muting system by engaging that channel's MUTE SAFE switch [22]. MUTE SAFE, in effect, blocks any of the channel's MUTE ASSIGN switches [21] so that the channel will be on so long as its ON/off switch [18] is engaged.

7.2.5 Signal Assignment Directly to the 18 Group Mixing Busses AND the Two Stereo Mixing Busses

The input channel bus assignment is very flexible. One can assign a channel by varying degrees directly to any of the 18 Group mixing busses by means of individual assign controls [14]. The same can be done directly to either of the two stereo mixing busses using the two sets of stereo LEVEL controls and the PAN pots [15] [16]. However, if the concentric switch around one of the PAN controls is set to the right, then the LEVEL control becomes an assign level control to the left side of that stereo bus, and the PAN control becomes an assign level control directly to the right side of the same stereo bus. This gives an option for up to 22 "Group" busses when no direct-to-stereo mix bus is required.

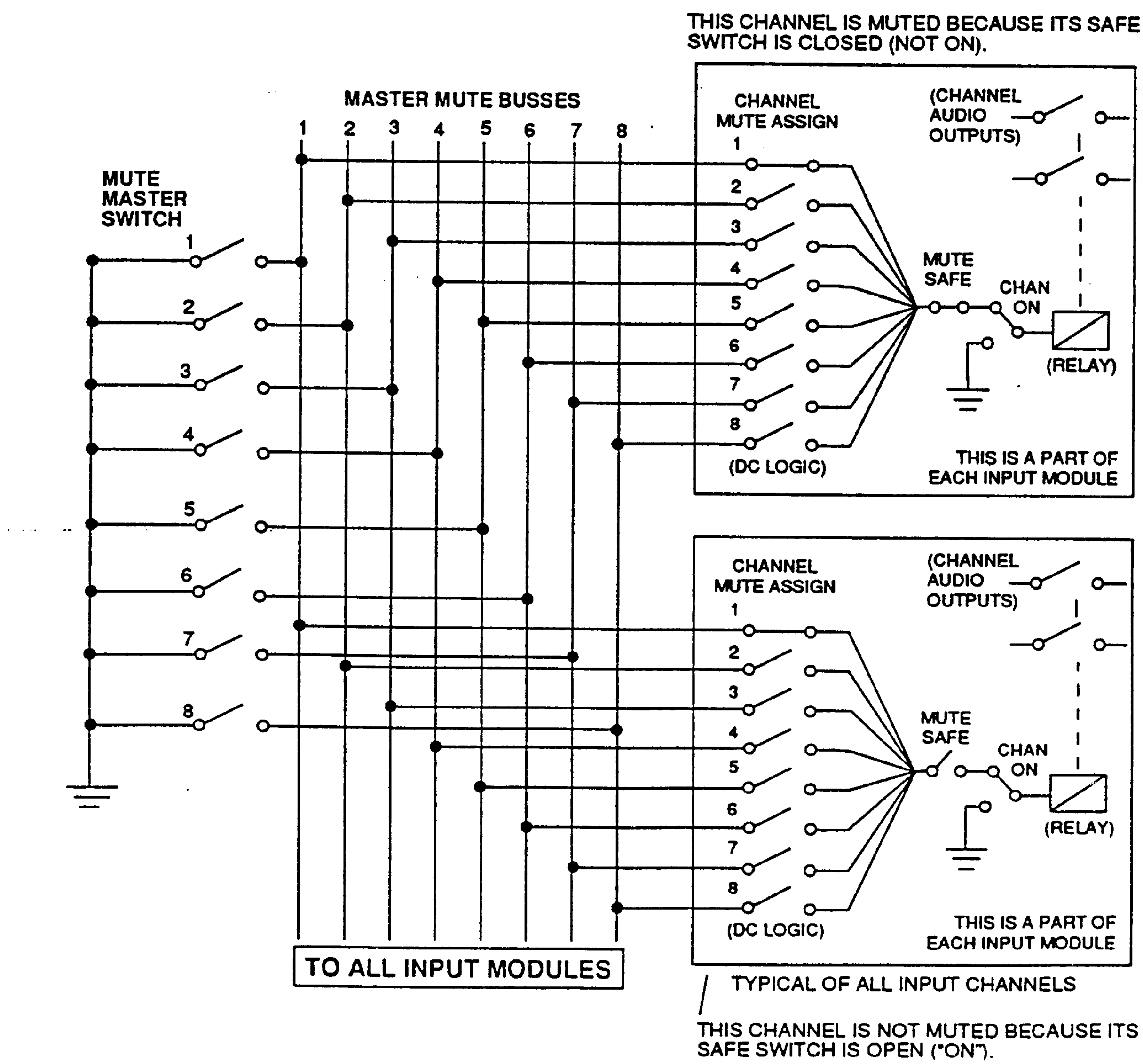


Figure 7-3. Block Diagram of the PM4000M Master Mute System

On the other hand, the 18 individual Assign controls [14] are arranged in pairs that facilitate their use for creation of up to 9 stereo mixes. You can decide to use an even-numbered odd-numbered bus (i.e., 1, 3, 5, etc.) for the left-side of a stereo mix, and the adjacent even-numbered bus (i.e., 2, 4, 6, etc.) for the right-side of that mix. Turning the odd channel level down and the even one up pans the signal right; vice-versa pans the signal left, and turning both bus assigns to 3 dB below maximum places the signal in the center. (You would not typically turn both all the way up because this would cause a 3 dB build-up of level with the two busses adding together in the center.)

Why would one want to utilize so many stereo mixes – up to 9 pairs of groups plus Stereo 1 and Stereo 2?

There are instances when more than two stereo mixes will be required. For example, suppose a given production is being done in stereo, with many input channels assigned directly to the primary stereo bus via the ST 1 (or ST 2) mix controls [15] [16]. Let's assume here that the drum set is being mixed in stereo, and that it must be compressed as a stereo drum mix. One does not want the considerable compression required for the drums to "dip" the other sources in the mix, so using the Stereo Bus Insert Point for primary drum compression is not a viable option. Instead, all drum input channels can be assigned to a pair of odd and even numbered group busses which are used as a stereo pair. The INSERT IN/OUT jacks of those two group busses [90] are then patched to a stereo compressor/limiter, which affects only the stereo drum mix. These two groups are then folded into the main house mix by means of their Group-to-Stereo controls [33] – panning one group fully left and the other fully right.

Alternately, the separate stereo programs mixed on pairs of group busses can be used for completely different purposes and never mixed together (do not use the Group-to-Stereo controls). A typical example is where multiple stereo mixes are required for wireless stage monitor systems.