SECTION 3
PM2800M SPECIFICATIONS

3.1 General Specifications

Total Harmonic Distortion
Less than 0.1%, 20 Hz – 20 kHz, at +14 dBu output into 600 ohms

Frequency Response
+1, -3 dB, 20 Hz – 20 kHz, at +4 dBu output into 600 ohms

Hum & Noise
(20 Hz – 20 kHz, R_s = 150 ohms, Input Pad @ 0 dB, Input Sensitivity @ -60 dB, except as noted)

-128 dBu	equivalent input noise.
-93 dBu	residual output noise (balanced outputs).
-74 dBu	at MIX OUT with Master fader at nominal level and all channel Mix Send Control at minimum level.
-64 dBu	(68 dB S/N) at MIX OUT with Master fader and one channel fader and Mix Send Control at nominal level.
-74 dBu	at STEREO OUT with Stereo Master fader at nominal level and all channel St Send Control at minimum level.
-64 dBu	(68 dB S/N) at STEREO OUT with Stereo Master fader and one channel fader and St Send Control at nominal.
-84 dBu	at MATRIX OUT with MATRIX MASTER Control at nominal level and all matrix mix controls at maximum level, an all MIX-TO-MATRIX switch is off.
-70 dBu	(74 dB S/N) at MATRIX OUT with MATRIX MASTER control at nominal level, one Matrix Mix control at maximum level, the corresponding mix Master fader at nominal level, MIX-TO-MATRIX switch on and all channel switch is off.
-74 dBu	at AUX OUT with Aux Master fader at nominal level and all channel AUX Send controls at minimum.
-64 dBu	(68 dB S/N) at AUX OUT with Aux Master fader, one channel Fader and AUX Send control at nominal level.

Maximum Voltage Gain
94 dB CH IN to MIX OUT
104 dB CH IN to MTRX OUT
74 dB CH IN to CUE OUT
10 dB SUB IN to MIX OUT
94 dB CH IN to STEREO OUT
94 dB CH IN to AUX OUT
20 dB AUX RTN to MIX OUT
10 dB SUB IN to AUX OUT

Input Channel Gain Control
34 dB variation in gain stop-to-stop

INPUT CHANNEL PAD SWITCH
0, 20, 40 dB of attenuation

INPUT CHANNEL EQUALIZATION
15 dB maximum boost or cut in the each of four bands

HIGH:	1.6 kHz	—	16 kHz	(shelving)
HI-MID:	800 Hz	_	8 kHz	(peaking)
LO-MID:	160 Hz	—	1.6 kHz	(peaking)
LOW:	40 Hz	—	400 Hz	(shelving)

INPUT CHANNEL HIGH PASS FILTER
12 dB/octave roll off below 20 Hz — 400 Hz (adjustable -3 dB point)

CROSSTALK
-60 dB at 1 kHz

OSCILLATOR/NOISE GENERATOR
Switchable sine wave at 100 Hz, 1 kHz, or 10 kHz (less than 1% THD at +4 dBu output level), or pink noise.

VU METERS
STEREO L & R: 2 large, illuminated meters with Peak LEDs. Other meters are smaller size without Peak LEDs. All meters calibrated for 0 VU = +4 dBu = 1.23 V_{RMS} output; Peak LEDs turn on 10 dB before clipping

Meters 1 – 8	MIX 1 – 8
Meter 9	STEREO L
Meter 10	STEREO R
Meters 11 – 14	AUX 1 – 4 / MTRX 1 – 4
Meter 15	CUE L / TB
Meter 16	CUE R / OSC

SIGNAL/CLIP INDICATORS
3 LEDs built into each input module monitor levels in the module: SIGNAL (green) turns on when pre-EQ signal is 10 dB below nominal level. CLIP (red) turns on when pre-EQ signal is 3 dB below clipping. EQ CLIP (red) turns on when post-EQ level is 3 dB below clipping.

PHANTOM POWER
48 V DC is applied to electronically balanced inputs or optional transformer- isolated inputs (via 6.8 kohm current limiting/isolation resistors) for powering condenser microphones. May be turned on or off via rear-panel phantom master switch; when on, individual channels may be turned off via +48 V switch on each input module.

OPTIONS

IT1800 Input Transformers; may be installed in individual input modules. Changes actual input impedance from 3K ohms to 1K ohm.

OT1800 Set of 4 output transformers, or OT3000 Set of 8 output transformers in rack-mountable external chassis with male and female XLR connectors on the front panel. Occupies 2 rack spaces (3-1/2" or 88 mm) in a 19" (480 mm) wide rack; 3-1/2" (88 mm) depth. May be used to isolate any PM2800M XLB outputs.

Miniature lamps on flexible supports to mate with 4-pin XLB (NC4FP) TYPE sockets in console; 3 sockets for 32 CH console, 4 sockets for 40 CH console.

Dust cover.

POWER REQUIREMENTS

Requires Yamaha PW2800 power supply; see specifications for that unit.

CONSOLE DIMENSIONS

HEIGHT	both models	12-33/64 inches	(318 mm)
DEPTH	both models	37-23/32 inches	(958 mm)
WIDTH:	32 channel	62-31/64 inches	(1587 mm)
	40 channel	73-1/4 inches	(1863 mm)

	32 CH	**40 CH**
NET	234.8 lbs	283.3 lbs
WEIGHT	106.5 kg	128.5 kg
(excl. power supply)		

NOTES: 0 dBu is referenced to 0.775 V_{RMS}. 0 dBm is referenced to 1 milliwatt. Specifications are subject to change without notice or obligation.

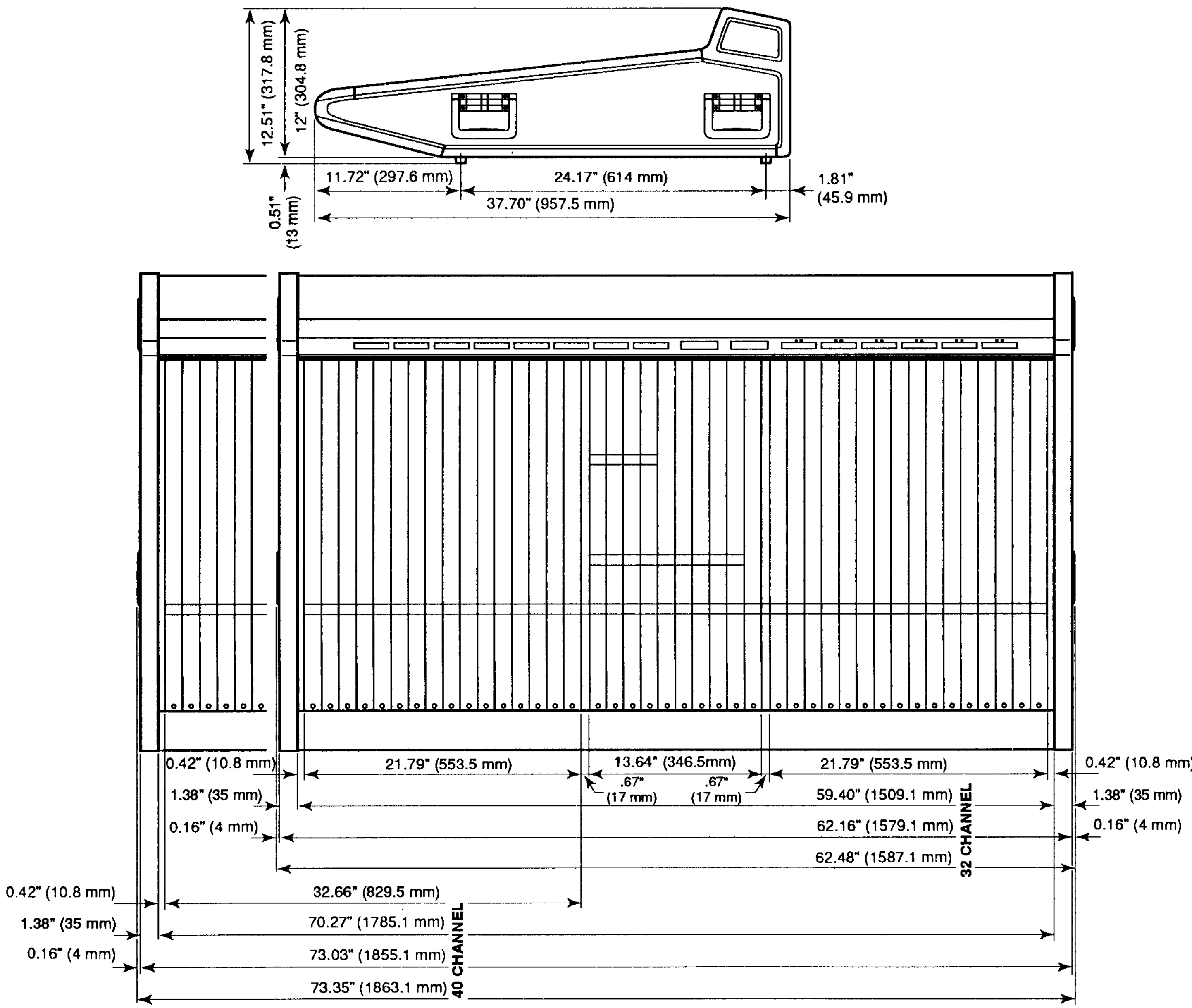


Figure 3-1. PM2800M Dimensions.

3.2 POWER SUPPLY (PW2800A) SPECIFICATIONS

DIMENSIONS:

HEIGHT 5-1/5 inches (132 mm) (excluding rubber feet; add 7/16" (10.5 mm) for feet).
DEPTH Overall, 16-9/10 inches (429.2 mm); Behind panel, 15-1/2 inches (394 mm).
WIDTH 18-9/10 inches (480 mm); for standard rack mounting.

NET WEIGHT

33.4 pounds (15.2 kg).

FUSE

Primary fuse 7 amp, slo-blo.

AC REQUIREMENTS

U.S.A./Canada models: 105 to 130 V, 50/60 Hz.
General Export models: 220 or 240 V, ±10%, 50/60 Hz.

UMBILICAL CABLE

Multi-conductor cable with locking, multi-pin connector conveys power to the PM2800M console. Cable is approximately 10 feet (3.6 meters) long.

COOLING

Internal fan, pulls air through foam grille on front panel, exhausts via top and side vents.

NOTE: Specifications are subject to change without notice or obligation.

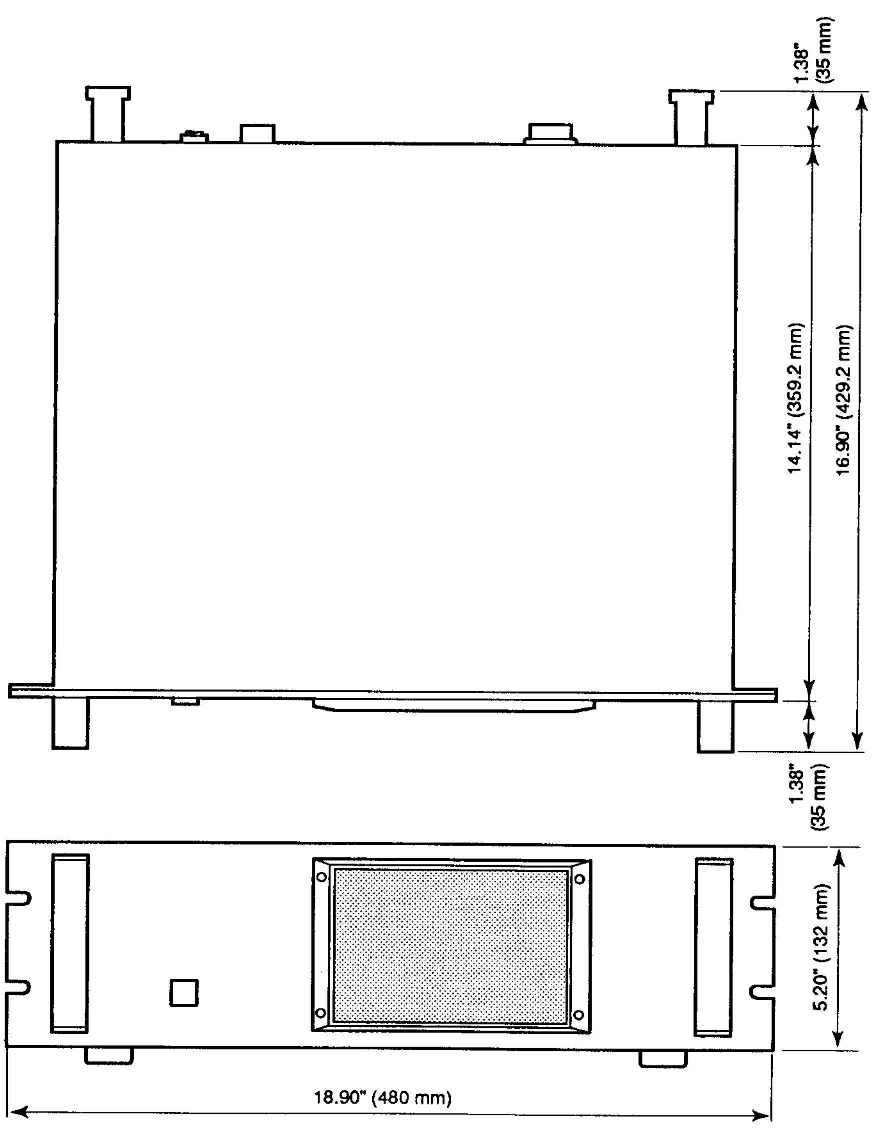


Figure 3-2. PW2800 Dimensions.

3.3 INPUT CHARACTERISTICS

Connection	Pad	Gain Trim	Actual Load Impedance	For Use With Nominal	Input Level			Connector In Console[2]
					Sensitivity[1]	Nominal	Max. Before Clip	
CH INPUT[3] 1 ~ 32, or 1 ~ 40	0 0 20 40	-60 -26 ~	3K ohms if electronically balanced; 1K ohms if Xfmr bal.	50 ohm to 600 ohm mics and 600 ohm lines	-90 dBu (0.025 mV) -56 dBu (1.23 mV) -36 dBu (12.3 mV) -16 dBu (123 mV)	-60 dBu (0.75 mV) -26 dBu (38.8 mV) -6 dBu (388 mV) +14 dBu (3.88 V)	-34 dBu (13.5 mV) 0 dBu (775 mV) +20 dBu (7.75 V) +24 dBu (12.3 V)	XLB-3-31 type
AUX RETURN 1 ~ 4 (stereo)			10K ohms	600 ohm lines	-16 dBu (123 mV)	+4 dBu (1.23 V)	+24 dBu (12.3 V)	XLB-3-31 type
SUB IN: MIX 1 ~ 8 AUX 1 ~ 4 CUE			10K ohms " 33K ohms	600 ohm lines " "	- 6 dBu (388 mV) - 6 dBu (388 mV) - 6 dBu (388 mV)	+4 dBu (1.23 V) +4 dBu (1.23 V) +4 dBu (1.23 V)	+24 dBu (12.3 V) +24 dBu (12.3 V) +24 dBu (12.3 V)	XLB-3-31 type Phone Jack[4]
TALKBACK IN	-50 +4		3K ohms "	50 – 600 ohm mics 600 ohm lines	-70 dBu (0.25 mV) -16 dBu (123 mV)	-50 dBu (2.45 mV) +4 dBu (1.23 V)	-24 dBu (48.9 mV) +24 dBu (12.3 V)	XLB-3-31 type
COMM IN	-50 +4		3K ohms "	50 – 250 ohm mics 600 ohm lines	-70 dBu (0.25 mV) -16 dBu (123 mV)	-50 dBu (2.45 mV) +4 dBu (1.23 V)	-24 dBu (48.9 mV) +24 dBu (12.3 V)	XLB-3-31 type
INSERT IN: CH 1 ~ 32 or 40 MIX 1 ~ 8 AUX 1 ~ 4			10K ohms	600 ohm lines	-16 dBu (123 mV)	-6 dBu (3883 V)	+20 dBu (7.75 V)	Phone Jack[4]

3.4 OUTPUT CHARACTERISTICS

Connection		Actual Source Impedance	For Use With Nominal	Output Level		Connector In Console[2]
				Nominal	Max. Before Clip	
MIX OUT STEREO OUT MATRIX OUT AUX OUT CUE OUT OSC OUT TALKBACK OUT	1 ~ 8 L, R 1 ~ 4 1 ~ 4 L, R	150 ohms 150 ohms 150 ohms 150 ohms 150 ohms 150 ohms 150 ohms	600 ohm lines 600 ohm lines 600 ohm lines 600 ohm lines 600 ohm lines 600 ohm lines 600 ohm lines	+4 dBu (1.23 V) +4 dBu (1.23 V) +4 dBu (1.23 V) +4 dBu (1.23 V) +4 dBu (1.23 V) +4 dBu (1.23 V) +4 dBu (1.23 V)	+24 dBu (12.3 V) +24 dBu (12.3 V) +24 dBu (12.3 V) +24 dBu (12.3 V) +24 dBu (12.3 V) +24 dBu (12.3 V) +24 dBu (12.3 V)	XLB-3-32 type XLB-3-32 type XLB-3-32 type XLB-3-32 type XLB-3-32 type XLB-3-32 type XLB-3-32 type
INSERT OUT:	CH 1 ~ 32 or 1 ~ 40	600 ohms	10K ohm lines	-6 dBu (388 mV)	+20 dBu (7.75 V)	Phone Jack
MIX OUT AUX OUT	1 ~ 8 1 ~ 4	600 ohms "	10K ohm lines "	-6 dBu (388 mV) "	+20 dBu (7.75 V) "	Phone Jack "
PHONES OUT		15 ohms	8 ohm phones 40 ohm phones	75 mW 65 mW	150 mW 130 mW	Phone Jack (TRS, Stereo Wired)

NOTES:

1. Sensitivity is the lowest level that will produce an output of +4 dBu (1.23V), or the nominal output level, when the circuit is set to maximum gain.
2. All XLB connectors are electronically balanced, except as noted below. Phone jacks are unbalanced.
3. IT1800 Input Transformers; may be installed in individual input modules. Changes actual input impedance from 3K ohms to 1K ohms.
4. Tip/Ring/Sleeve type phone jack.

3.5 PERFORMANCE GRAPHS

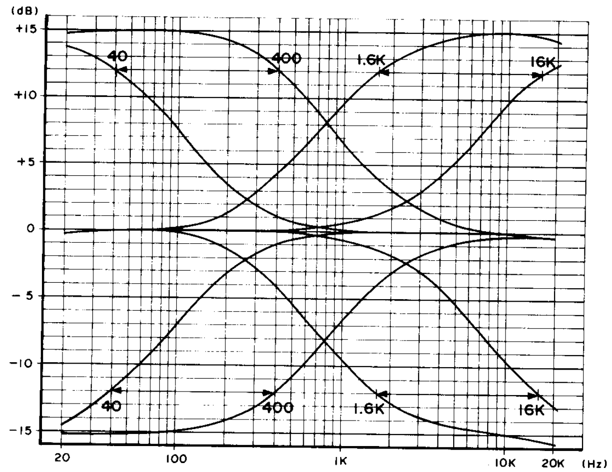


Figure 3-3. High and Low Frequency Band Equalizer Characteristics (shelving).

(a) ***(b)***

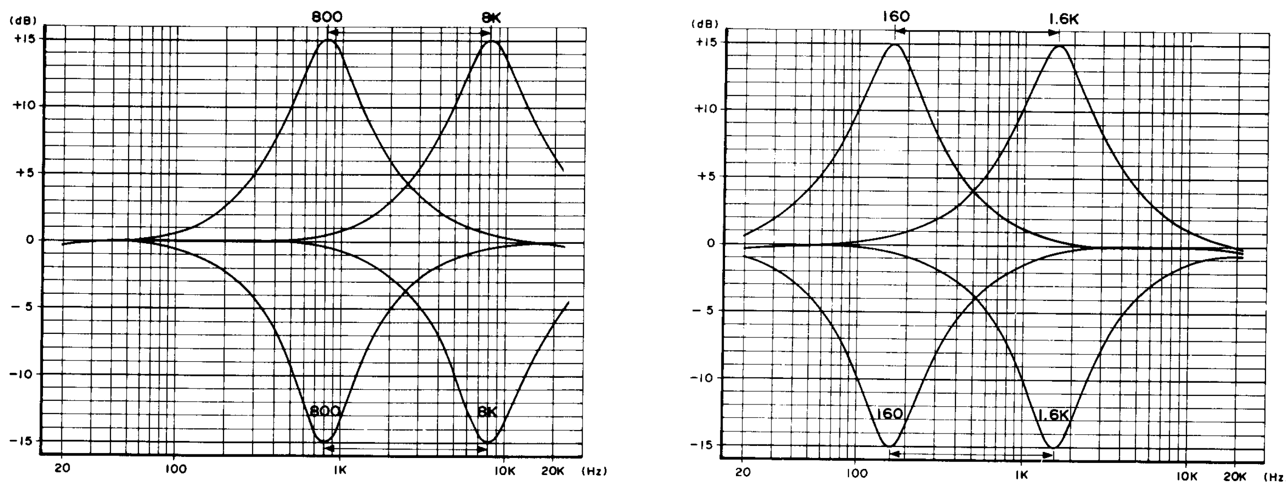


Figure 3-4. Hi-Mid (a) and Lo-Mid (b) Band Equalizer Characteristics (peaking).

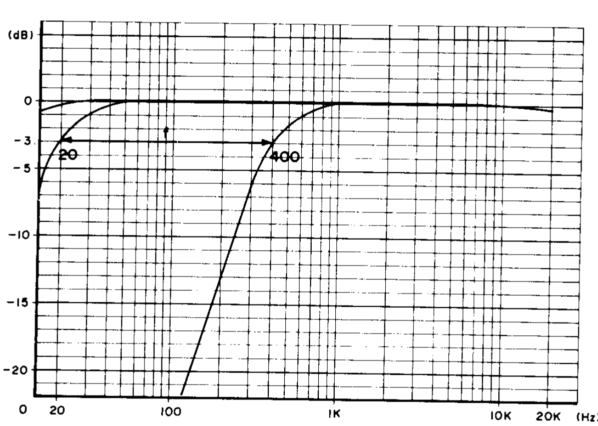


Figure 3-5. High Pass Filter Characteristics.

3.5.1 Input Channel 1 to Mix Output 1 Performance Graphs with Input Gain Control @ Max

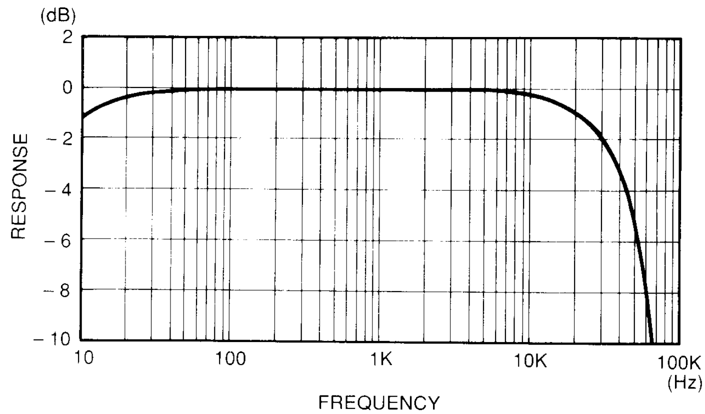


Figure 3-6. Frequency Response
At +4 dBu & +14 dBu output levels, PAD at 0 dB.(Curves would be identical with PAD at 20, or 40 dB.),

A) PAD at 0 dB

B) PAD at 20 dB

C) PAD at 40 dB

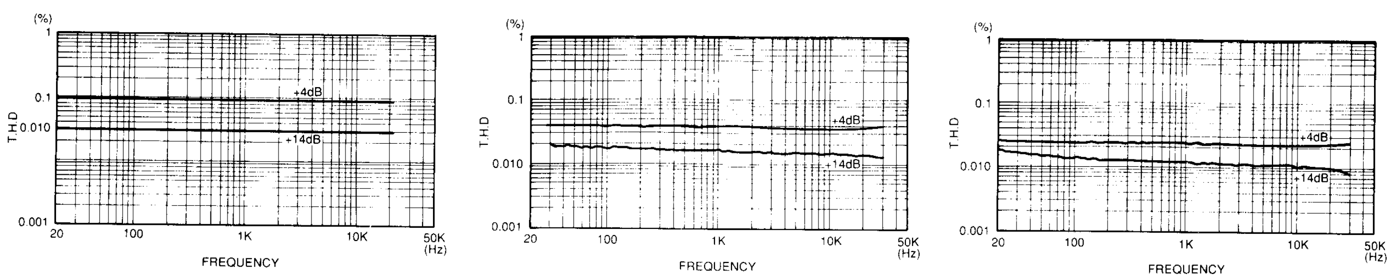


Figure 3-7. Frequency vs. THD Curves
At +4 dBu & +14 dBu output levels.

A) PAD at 0 dB

B) PAD at 20 dB

C) PAD at 40 dB

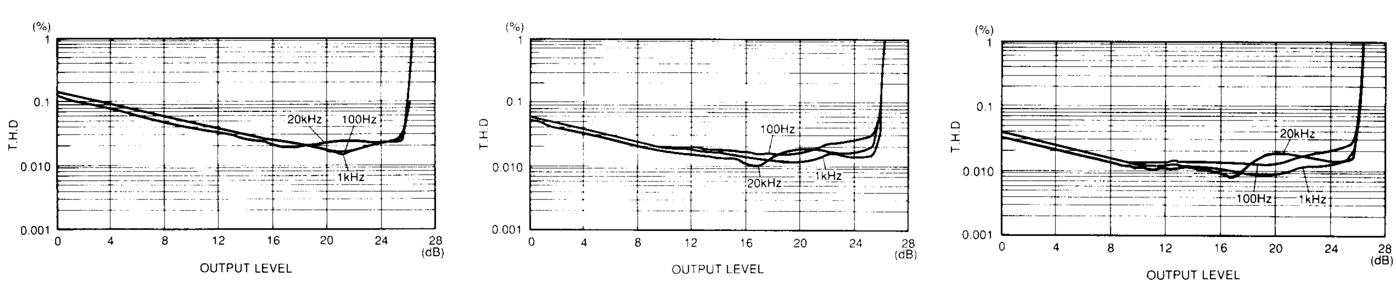


Figure 3-8. Output Level vs. THD Curves
At 100 Hz, 1 kHz & 20 kHz.

3.5.2 Input Channel 1 To Mix Output 1 Performance Graphs With Input Gain Control @ Min

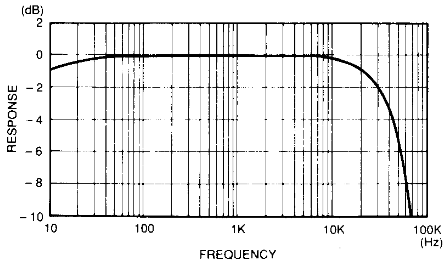


Figure 3-9. Frequency Response
At +4 dBu & +14 dBu output levels, PAD at 0 dB. (Curves would be identical with PAD at 20 or 40 dB.)

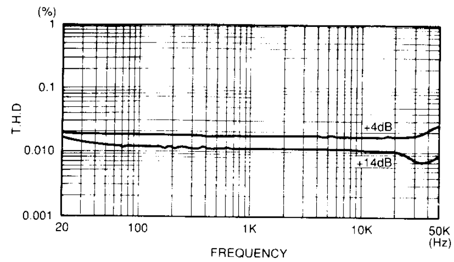


Figure 3-10. Frequency vs. THD Curves
At +4 dBu & +14 dBu output levels, PAD at 0 dB.

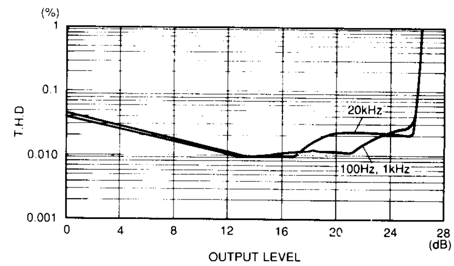


Figure 3-11. Output Level vs. THD
At 100 Hz, 1 kHz & 20 kHz, PAD at 0 dB.

3.5.3 Aux Return 4 (I) To Mix Output 1 Performance Graphs

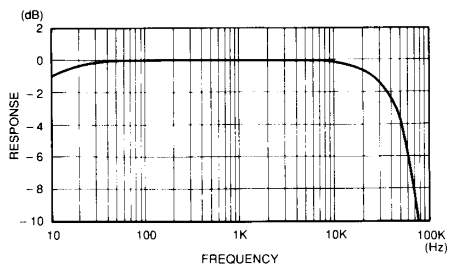


Figure 3-12. Frequency Response
(At +4 dBu output level.)

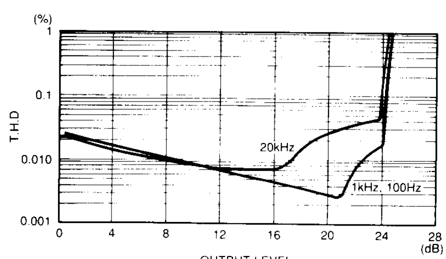


Figure 3-13. Output Level vs. THD
(At 100 Hz, 1 kHz & 20 kHz.)

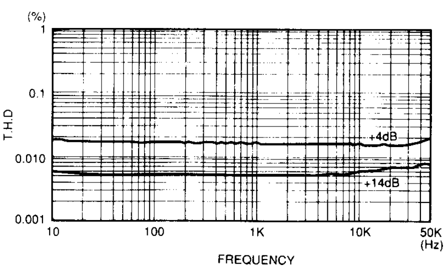


Figure 3-14. Frequency vs. THD Curves
(At +4 dBu & +14 dBu output levels.)

3.5.4 Channel 1 Input to Phones Output Performance Graphs with Input Pad @ 40 dB, Gain @ Min.

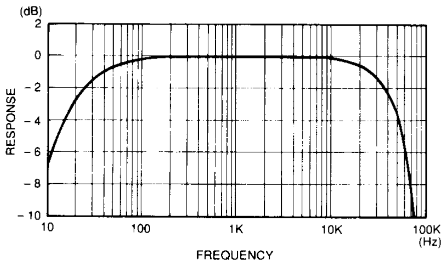


Figure 3-15. Frequency Response.

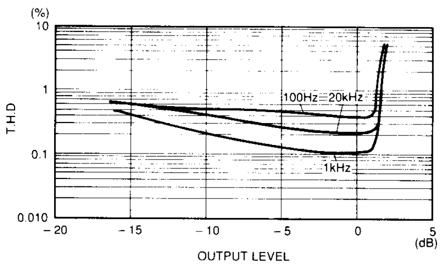


Figure 3-16. Output Level vs. THD At 100 Hz, 1 kHz & 20 kHz.

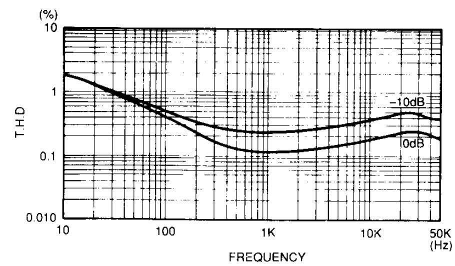


Figure 3-17. Frequency vs. THD Curves.

3.5.5 Crosstalk Performance Graphs

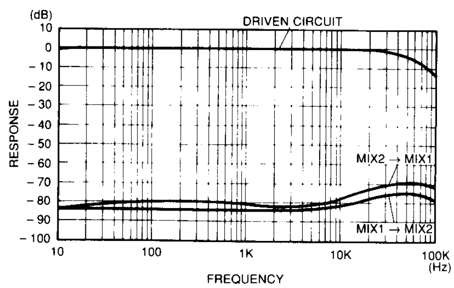


Figure 3-18. Crosstalk of Mix 1 Into 2 or 2 Into 1 with Input Pan Control at Full CW & Full CCW Positions

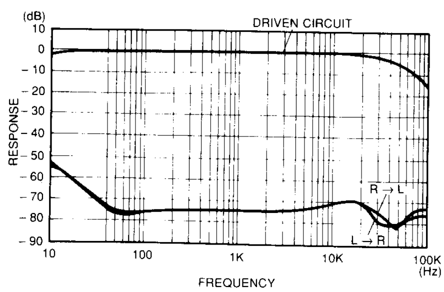


Figure 3-19. Crosstalk of Stereo L Into R or R Into L with Stereo Pan Control at Full CW & Full CCW Positions

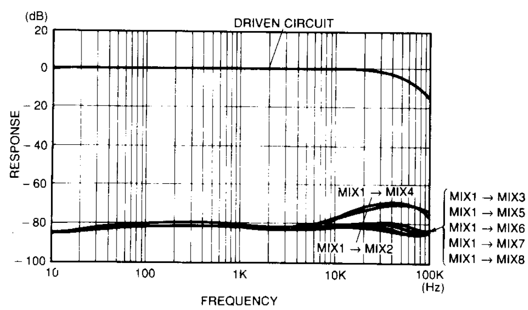
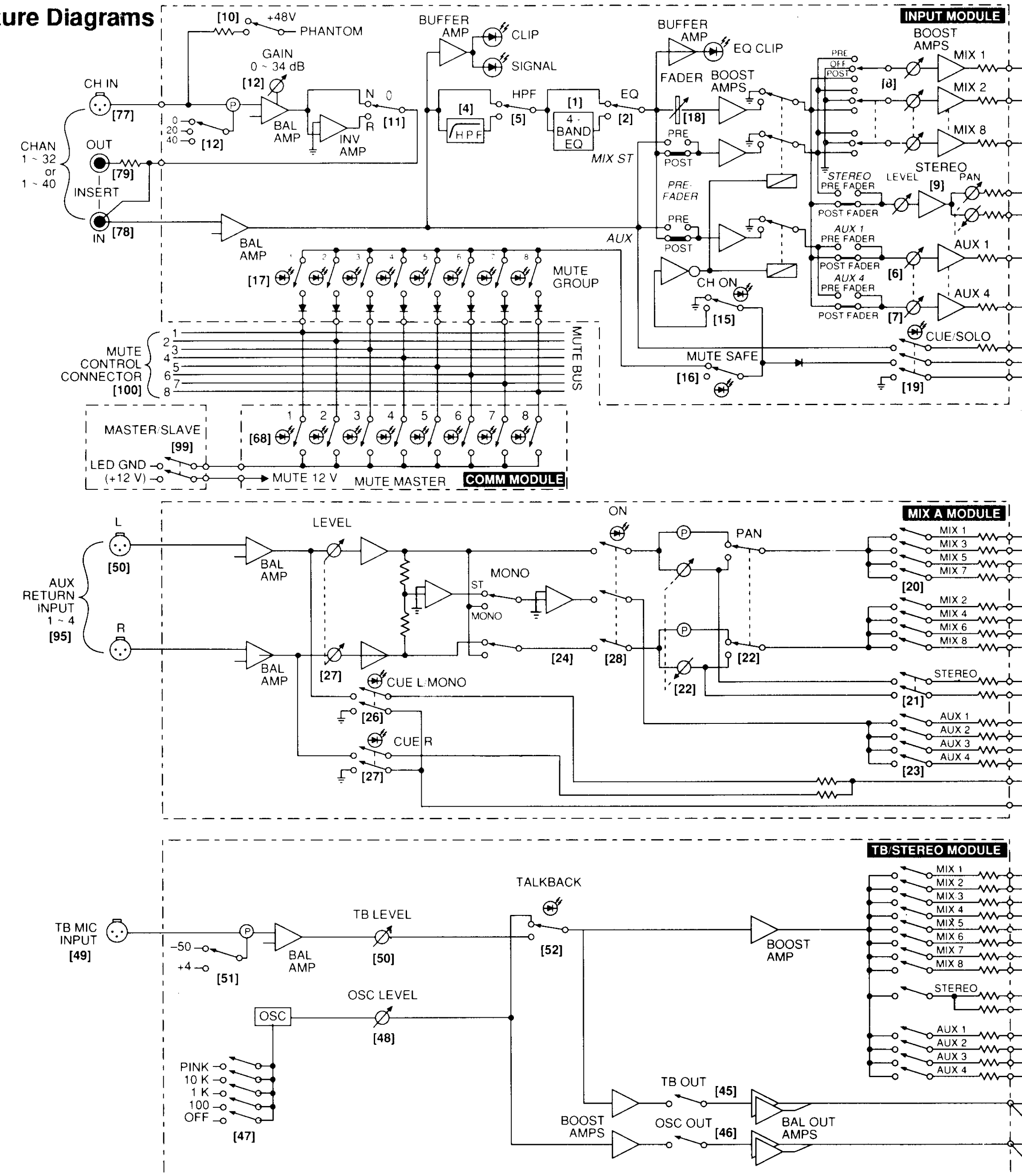


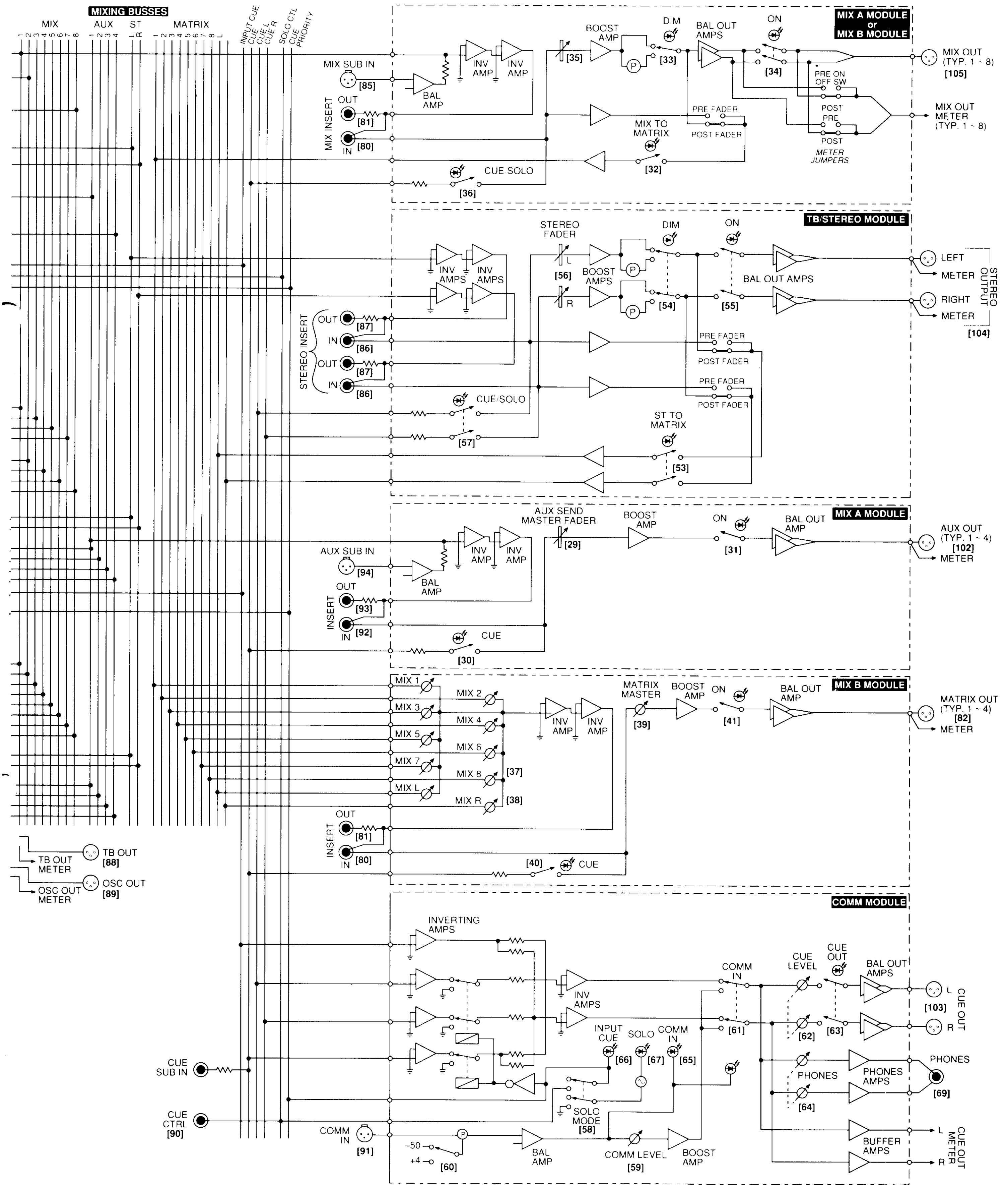
Figure 3-20. Crosstalk of Mix Bus 1 Into Mix Busses 2 through 8

3.6 Block & Gain Structure Diagrams



Note: The numerals in square brackets on this diagram refer the associated control or connector to the callout numbers on the diagrams and text in Section 2 of this manual.

Figure 3-21. PM2800M Signal Flow (Block Diagram).


MIXING BUSSES
MIX
AUX
ST
MATRIX
INPUT CUE
CUE L
CUE R
SOLO CTL
CUE PRIORITY
MIX A MODULE or MIX B MODULE
MIX SUB IN [85]
MIX INSERT
BOOST AMP
DIM
BAL OUT AMPS
ON
INV AMP
BAL AMP
MIX OUT (TYP. 1 ~ 8) [105]
MIX OUT METER (TYP. 1 ~ 8)
MIX TO MATRIX
PRE FADER
POST FADER
METER JUMPERS
CUE SOLO
TB/STEREO MODULE
STEREO FADER
INV AMPS
BOOST AMPS
BAL OUT AMPS
STEREO INSERT
LEFT
METER
RIGHT
STEREO OUTPUT [104]
CUE/SOLO
ST TO MATRIX
MIX A MODULE
AUX SEND MASTER FADER
AUX SUB IN [94]
INSERT
CUE
AUX OUT (TYP. 1 ~ 4) [102]
MIX B MODULE
MATRIX MASTER
MATRIX OUT (TYP. 1 ~ 4) [82]
TB OUT [88]
TB OUT METER
OSC OUT [89]
OSC OUT METER
COMM MODULE
INVERTING AMPS
COMM IN
CUE LEVEL
CUE OUT
INPUT CUE
SOLO
SOLO MODE [58]
PHONES [69]
PHONES AMPS
CUE SUB IN
CUE CTRL [90]
COMM IN [91]
COMM LEVEL [59]
BOOST AMP
BUFFER AMPS
CUE OUT METER

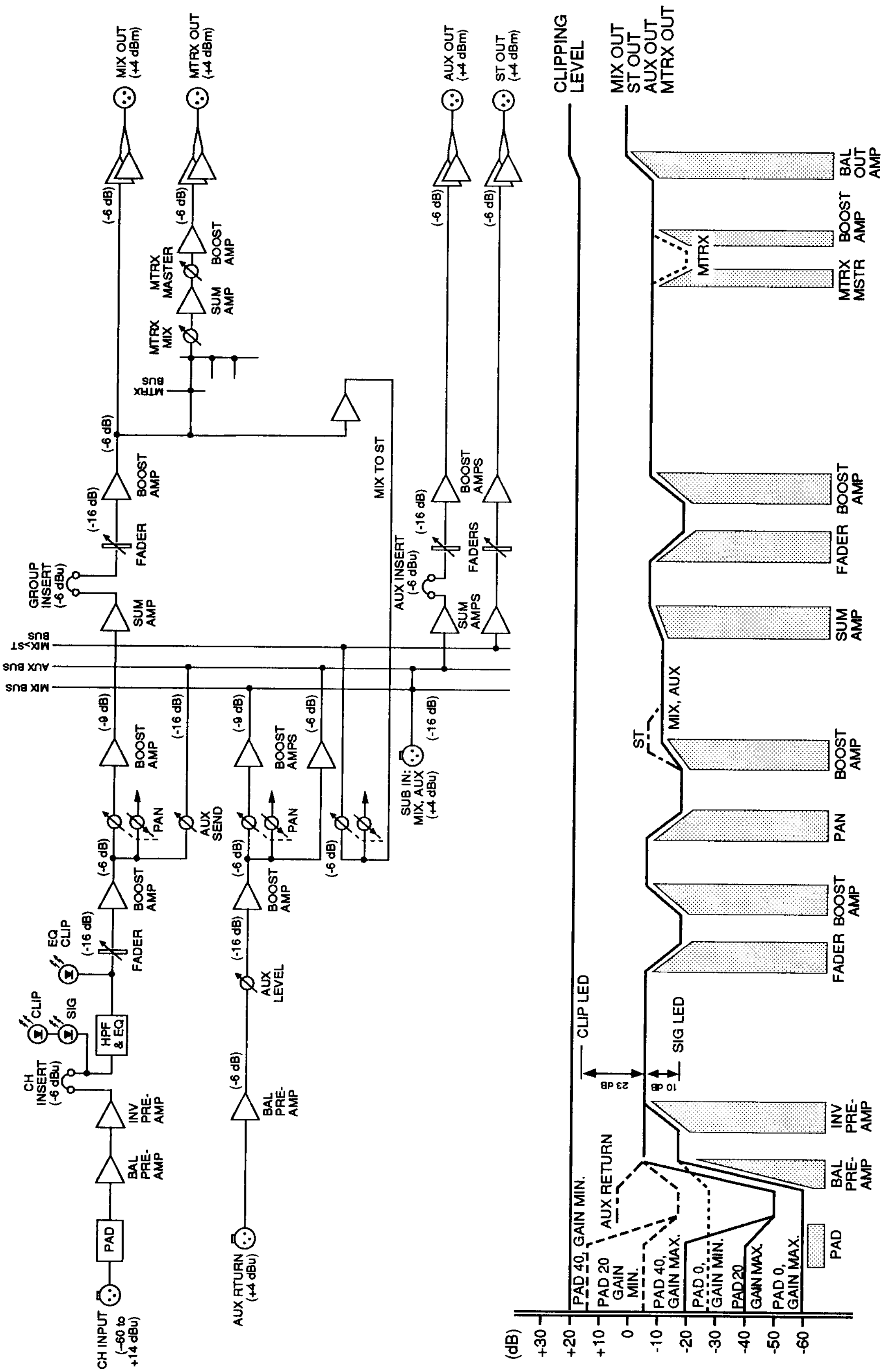


Figure 3-22. PM2800M Gain Structure.

SECTION 4
INSTALLATION NOTES

4.1 PLANNING AN INSTALLATION

Before installing the PM2800M, it is worthwhile considering how it will be used, how it is going to be connected, and what is the best way to implement the installation.

To begin with, there must be a surface upon which the console can be mounted. A desk or table top can be constructed to support the console. It should be capable of supporting at least the weight of the console plus a human console operator leaning on the arm rest; the sturdier, the better. There should be adequate access behind the console to allow for cable connections and "service loops" of extra cable so that the console can be moved without disconnecting everything. The dimensions listed in the SPECIFICATIONS section of this manual can be given to the carpenter or other personnel responsible for building the console support.

Be sure to provide a location within 10 feet (3.6 meters) of the console for housing the PW2800 power supply. This supply may be rack mounted, or it may be placed on a shelf. For touring or critical fixed applications, it may be advisable to purchase a spare PW2800 supply and to keep it next to the main supply for rapid changeover in the rare event of a problem.

Experienced sound system installers will prepare a detailed block diagram of the entire sound system prior to installation. They will figure out all the necessary cables, where they run, and the required length so that the cables can be prepared ahead of time. In fixed installations, this will enable appropriate conduit to be installed (be sure to allow some extra "breathing room" in the conduit to allow for cable replacement or future additions. For open-air installations, such as outdoor amphitheaters, there is no substitute for waterproof conduit (it excludes moisture in the event of rain or when the venue is washed down, thereby preventing deterioration and short circuit of audio and power cables). Refer to additional wiring information under AUDIO CONNECTORS AND CABLE TYPES.

4.2 POWER MAINS

4.2.1 Verify the Correct Mains Voltage

PW2800 power supplies sold in the U.S.A. and Canada are designed to operate with 110 to 120 volt, 50 or 60 Hz AC power mains. The General Export model operates on 220 or 240 volt, 50 or 60 Hz AC mains. If you are traveling with this equipment, be sure to test the power mains, and to use the appropriate power supply. Consult your Yamaha PM2800M dealer for assistance.

4.2.2 Ensure There is a Good Earth Ground

The console must be grounded for safety and proper shielding. A three-wire power cable is provided for this purpose. Use a special circuit tester to insure that the outlet is properly grounded, and that the "neutral" is not weak or floating. If a grounded, three-wire outlet is not available, or if there is any chance the outlet may not be properly grounded, a separate jumper wire must be connected from the console chassis to an earth ground.

In the past, cold water pipes often were relied upon for an earth ground, although this is no longer the case in many localities. Modern building codes often specify that the water meter be isolated from the water mains by a length of plastic (PVC) pipe; this protects water company personnel working on the water mains from being shocked. It also insulates the cold water pipes from the earth ground. While an electrical wire bypasses the water meter in some locations, this ground path should not be assumed. For similar reasons, avoid hot water pipes. Gas pipes should not be used because if there is a poor electrical connection between two sections of pipe, and if a ground current is being dissipated through the pipe, there exists the potential for a heat or spark-generated fire or explosion. The safest and most reliable approach is to provide your own ground. Drive at least 5 feet (1.5m) of copper jacketed steel grounding rod into moist, salted earth, and use that for a ground, or use one of the specially made chemical-type ground rods available for this purpose.

CAUTION: Connect the PW2800 power supply to the power mains only after confirming that the voltage and line frequency are correct. At the least, use a voltmeter. It is also a good idea to use a special outlet tester that will also indicate reversed polarity, weak *or missing neutral, and weak or missing ground connections in the outlet. Test the power supply before connecting the umbilical cables to the console.*

Severe over voltage or under voltage in the power mains can damage your equipment. For U.S.A. and Canadian models, the power line must measure more than 105V and less than 130V RMS. The tolerance for General Export models is plus or minus 10%. Some lines are "soft," meaning that the voltage drops when the line is loaded due to excessive resistance in the power line, or too high a current load on the circuit. To be certain the voltage is adequate, check it again after turning on the PW2800 with the PM2800M connected, and with any power amplifiers turned on if they are connected to the same power mains.

If the power line voltages do not fall within the allowable range, do not connect the PW2800 to the mains. Instead, have a qualified electrician inspect and correct the condition. Failure to observe this precaution may damage the power supply and console, and will void the warranty.

NOTE: The following discussions of AC outlet wiring are written for U.S.A. and Canadian power systems, although the principles generally apply worldwide. In other areas, however, be sure to check local codes for specific wiring standards.

4.2.3 How to Obtain a Safety Ground When Using a Two-wire Outlet

Two-wire AC outlets do not have a hole for the "safety ground" prong of a three-wire power cord. A two-wire to three-wire AC adaptor is required if you want to use one of these two-wire outlets with the three-wire AC plug on your sound equipment. These adaptors can maintain a safe ground for the sound system if you connect the loose green wire on the adaptor to a grounded screw on the two-wire outlet. How do you know whether or not the screw is grounded?

1. Connect the adaptor's green wire to the screw on the two-wire outlet.
2. Plug the adaptor into the outlet.
3. Plug in your three-wire AC outlet tester into the adaptor. The AC outlet tester will indicate whether the screw is grounded.

If the screw is not grounded, connect the adaptor's green wire to some other ground point in order to maintain a safe ground for your system. If the outlet tester indicates a good ground but reversed polarity on your two-wire to three-wire adaptor, sometimes you can reverse the adaptor in the outlet by pulling it out, twisting it a half-turn and reconnecting it; this may not be possible if the outlet or adaptor is "polarized" with one prong larger than the other.

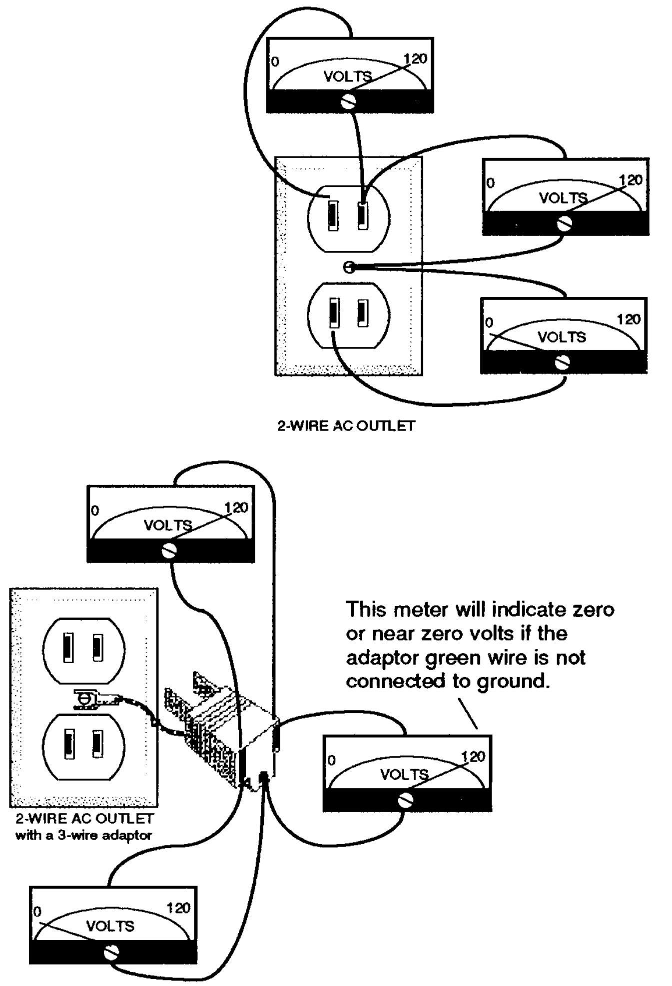


Figure 4-1. Testing a Two-wire AC Outlet.

4.2.4 Improperly Wired AC Outlets: Lifted Grounds

A "lifted ground" condition exists if the ground or green wire from the outlet's safety ground is disconnected or missing. In older wiring, the heavy green wire was sometimes omitted from internal wall wiring in favor of letting the flexible metal conduit or rigid conduit (pipe) suffice as the ground path from the electrical service entrance. This method of grounding is generally acceptable, as long as the metal conduit in the wall is intact and all the screws holding the joints together are secure. However, a single loose screw in a conduit joint inside a wall can remove the safety ground from the next outlet box in the line, and from all the subsequent boxes on that same line.

4.2.5 Improperly Wired AC Outlets: Lifted Neutral

If the neutral becomes lifted at a power outlet, it is possible that items plugged into the outlet will be fed the full 220 to 240 volts available from the power service instead of the desired 110 to 120 volts.

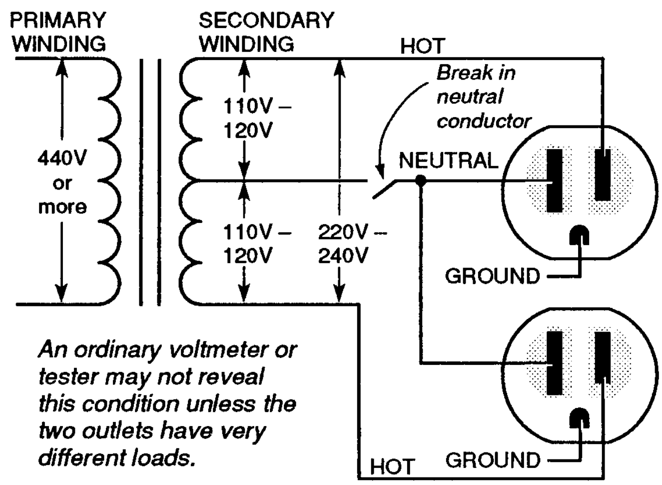


Figure 4-2. Schematic of an Outlet with a Lifted Neutral.

Such outlets may operate, but the voltage can swing from 0 volts to 220 or 240 volts AC (or whatever the maximum voltage at the service entrance), creating a shock hazard and possibly damaging your equipment.

If the PW2800 is plugged into one socket of the two outlets with lifted neutral, and a rack of signal processing equipment or power amplifiers is plugged into the other, fuses would probably blow upon turning on the system, and some of the sound equipment could be destroyed.

If you detect any voltage between the larger slot (white wire) in an outlet and the ground terminal (round prong, green wire) when there is no load on that line, you should contact a licensed electrician to check it out and correct the situation.

WARNING: In AC power wiring, black is hot, and white is neutral–the opposite of most audio signal wiring and speaker wiring. It is safer to consider all AC wiring as potentially lethal. It is possible someone miswired the system, or that a short circuit has developed. Test the voltages yourself, and be safe.

Although the white wires (neutral) and the green wires (ground) in the AC wiring are technically at the same potential (voltage), and should measure the same potential using a voltmeter, the ground prong connections at the outlets should be connected to the grounding bar that was driven into the earth as an additional safety precaution in case something should happen to the wires running from the service entrance transformer to the building or within the equipment itself. If a short should occur within the equipment, hopefully the electricity will find its way to ground via the safety ground, instead of via a person's body. When checking AC power lines at the outlet, be sure you have proper testing tools and some familiarity with the danger of shock hazards from AC power. Follow the diagram shown here, being careful not to touch metal with your hands. Do not short the test leads together. If you are not familiar with AC power distribution, don't experiment; have a licensed electrician perform these tests and correct any discrepancies.

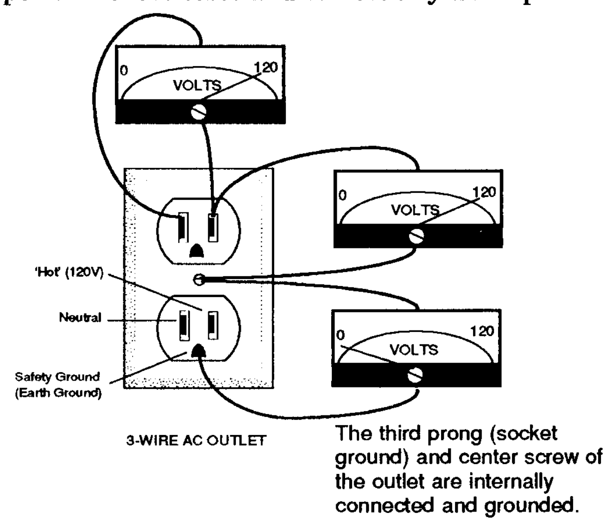


Figure 4-3. Testing a Three-wire AC Outlet.

4.2.6 AC Safety Tips

1. If you are going to verify the quality of AC wiring, there are two inexpensive items you should carry. One of these is a commercial outlet tester, the other is a neon lamp type AC voltage tester. These items are inexpensive and available at most hardware stores, electrical supply houses and some lighting stores. It is advisable to also have an RMS (or averaging) voltmeter to measure the exact AC line voltage.
2. The outlet tester should be used on all power outlets, The neon voltage tester should be used to check for voltage differences between microphone and guitar amps, microphones and electric keyboard chassis, and so forth.

3. If you're not sure whether an outlet is good, don't use it. Just in case, carry a long, heavy duty extension cord. A good extension should be made of #12-3 (12 gauge, 3 wires), and no longer than 15 meters (about 50 feet).
4. If there is no suitable power source at a venue, don't plug in your equipment. Any fault in the wiring of the AC outlet is potentially hazardous. Rather than take a chance with damage to equipment and possibly lethal shock, it is best to refuse to use a faulty outlet until it has been repaired by a licensed electrician. Don't take unnecessary risks.

4.2.7 Turn-On Sequencing

In larger systems, it is often difficult to obtain a sufficient number of 20-amp circuits to accommodate the power surges that may occur when the equipment is turned on. Many modern power amplifiers, for example, each require the full capacity of a 20-amp circuit at turn-on, though their operating current requirement is usually much lower. The solution to this problem is to use a stepped turn-on sequence; in fixed installations, the turn-on sequence is sometimes automated with timing and control circuitry.

4.2.8 Power Source Integrity

Finally, make every effort to assure that your source of power is clean and reliable. Synthesizers, computer sequencers and other digital equipment, in particular, normally require a filtered power source with surge protection in order to avoid glitches, system hangups and possible component damage. Power distribution strips with such protection built in are widely available commercially. The ultimate protection is provided by using a power line isolation transformer, such as the "Ultra Isolation" transformers sold by Topaz. Such devices are designed not only to exclude noise and distortion in the AC signal, but also to hold the voltage at the device's output to a nearly constant value regardless of major fluctuations of the line voltage at its input.

4.3 THEORY OF GROUNDING

Ground is the electrical reference against which potentials (voltages) are expressed. In a practical audio system, a number of different independent references exist in various local subsystems. These may or may not be at the same electrical potential; if handled properly, they certainly need not be at the same potential.

For purposes of clarity in discussing audio connection practices, we will distinguish among three specific ground references:

- **Signal Ground** — the reference point against which signal potentials in a specific piece of equipment or group of components are expressed.
- **Earth Ground** — the local electrical potential of the earth. In practice, earth is the potential of the central, rounded terminal in a US standard three-prong 120-volt outlet. Earth is sometimes obtained from a metal cold-water pipe (though this practice has been criticized recently as unreliable due to increasing use of non-conductive ABS plastic pipe sections), or from a chemical earthing rod sunk into the moistened ground.
- **Chassis Ground** — the chassis connection point of a specific component. In equipment fitted with a three-prong AC plug, the chassis is normally connected to earth, with provision to connect signal ground to earth as well. Equipment having a two-prong AC plug will normally have the chassis connected to signal ground.

As we shall see, connections among these various reference points are an all-important factor in assembling a successful audio system.

Grounding has been an area of "black magic" for many sound technicians and engineers, and certainly for most casual users of sound systems. Everyone knows that grounding has something to do with safety, and something to do with hum and noise suppression, but few people know how to set up a proper AC power distribution system, and how to connect audio equipment grounds so that noise is minimized. This subsection of the manual won't make anyone an expert, but it does point out a few of the principles and precautions with which everyone should be familiar. Whether you read this material or not, before you start cutting shields and lifting grounds, read the warning on the following page.

WARNING: 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. Yamaha shall not be liable for incidental or consequential damages, including injury to any persons or property, resulting from improper, unsafe or illegal installation of a Yamaha mixing console or of any related equipment; neither shall Yamaha be liable for any such damages arising from defects or damage resulting from accident, neglect, misuse, modification, mistreatment, tampering or any act of nature. (IN PLAIN WORDS... IF YOU LIFT A GROUND, THE RESULTING POTENTIAL FOR ELECTRICAL SHOCK IS YOUR OWN RESPONSIBILITY!)

Never trust any potentially hazardous system, such as an AC power system of any type, just because someone else tells you that it's okay. People can get killed by faulty or improperly wired sound equipment, so be sure you check things out yourself.

4.3.1. Why Is Proper Grounding Important?

In practical operating environments, any signal conductor is susceptible to induced currents from several types of sources such as radio-frequency (RF) emissions, AC power lines, switching devices, motors and the like. This is why audio signal cables are invariably shielded: the function of the shield is to "intercept" undesirable emissions. A major goal of grounding technique is to keep unwanted signal currents that are induced in the shield away from the signal conductor(s), and drain them to ground as directly as possible.

Beyond minimizing noise and hum, an equally important consideration in grounding is safety. The connection between a chassis and earth is commonly referred to as a "safety ground" – and with good reason. For example, consider the possibility that a chassis might become connected to "hot" leg of the AC mains (120 volts RMS AC) due to faulty wiring, an inadvertent short or moisture condensation. Suddenly, that innocuous-looking box could be transformed into what engineers gruesomely call a "widow maker." Someone who is touching a grounded guitar, mic stand, or other equipment will complete the circuit when touching the now electrically charged chassis, and receive the full brunt of whatever power is available. If the chassis is connected to earth, however, it will simply blow a fuse or circuit breaker.

Dangerous potential differences may also occur without such shorts. Two individual localized ground points, if they are not directly connected, cannot be assumed to be at the same potential – far from it, in fact. Virtually anyone who has played in a band has, at one time or another, experienced a shock when touching both the guitar and the microphone. The guitar may be grounded on-stage; the mic may be grounded at the console on the other side of the room; but the two grounds are at very different potentials. By completing the circuit between them, the performer gets "zapped." Good grounding practice seeks to control such potential differences for the comfort and longevity of all concerned.

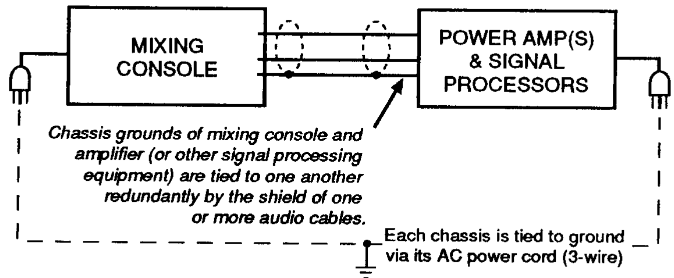
4.3.2. Ground Loops

Handled properly, cable shields are very effective in controlling unwanted induced signal currents. Yet line-frequency hum is, without question, the single most common problem in sound systems, and the most common cause of hum is ground loops.

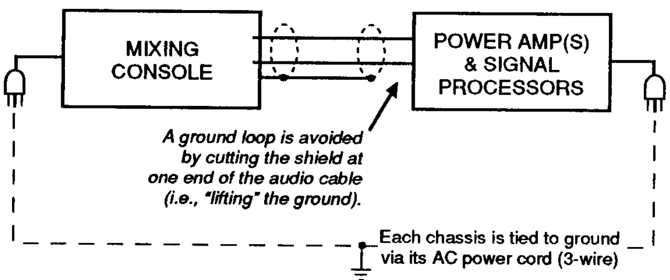
A ground loop can occur when there is more than one ground connection path between two pieces of equipment. The duplicate ground paths form the equivalent of a loop antenna which very efficiently picks up interference currents that are transformed by lead resistance into voltage fluctuations. As a consequence, the reference in the system is no longer a stable potential, so signals "ride" on the interference.

If all connections are balanced and the equipment is properly designed and constructed, such ground loops will not, in fact, induce noise. Unfortunately, much of the so-called "professional" sound equipment sold today is not properly grounded internally, so system-created ground loops can create very real problems.

Figure 4-4 shows a typical ground loop situation. Two interconnected pieces of equipment are plugged into grounded AC outlets at separate locations, and signal ground is connected to earth in each of them. The earth ground path and duplicate signal ground path form a loop which can pick up interference. Normally, this kind of ground loop should not cause any noise in the audio circuits if (a) the circuits are truly balanced or floating, and (b) the audio common is maintained separately from the chassis ground within the equipment. If one of these conditions is not met, then instead of going directly to earth ground and disappearing, these circulating "ground loop" noise currents (which act like signals) travel along paths that are not intended to carry signals. The currents, in turn, modulate the potential of the signal-carrying wiring (they are superimposed on the audio), producing hum and noise voltages that cannot easily be separated from program signals by the affected equipment. The noise is thus amplified along with the program material.



The ground path between the two AC plugs provides a redundant ground (ground loop) since the audio cable shield(s) already does the job.
This is a typical sound system ground loop caused by redundant audio shield and AC mains ground paths.



The dual ground path created by the AC cords does not create a ground loop since the two chassis are not grounded redundantly via the cable shield(s).
Elimination of the typical ground loop by cutting the shield of the audio cable retains AC safety.

Figure 4-4. Typical Ground Loops in Sound Systems.

Ground loops often are difficult to isolate, even for experienced audio engineers. Whenever you hear hum from a sound system, there is a strong possibility that it is being caused by a ground loop. Sometimes, in poorly designed sound equipment (which sometimes includes expensive sound equipment), ground loops occur inside the chassis even though the equipment has balanced inputs and outputs. In this instance, little can be done to get rid of the hum short of having a skilled audio engineer re-design the ground wiring inside. It's better to avoid this kind of equipment. It is also best to avoid unbalanced equipment in professional sound systems (unless the equipment is all going to be very close together, connected to the same leg of the AC service, and not subject to high hum fields).

Equipment does not have to be grounded to prevent noise from entering the system. The main reason we ground a sound system is for safety; proper grounding can prevent lethal shocks. The next reason for grounding a system that includes AC powered equipment is that, under some conditions, proper grounding may reduce external noise pickup. While proper grounding doesn't always reduce external noise pickup, improper grounding can increase external noise pickup.

The AC power cord ground (the green wire and the third pin on the AC plug) connects the chassis of electronic equipment to a wire in the wall power service that leads through building wiring to an "earth" ground. The earth ground is required by electrical codes everywhere, and can contribute to ground loops.

If there is only one path to ground, there can be no ground loop. However, one must look carefully. For example, suppose there is just one audio cable joining a console to a power amplifier... can there be a ground loop? Yes! A ground connection through the AC cables and the chassis of the two units makes the second connection. This, along with the audio cable shield, constitutes a continuous "ground loop" in which noise currents can flow. One way to break this ground loop is to "lift" the AC ground on one piece of equipment, typically the power amplifier, with a two-wire to three-wire AC adaptor. Leaving the loose green wire on the adaptor unconnected breaks the ground loop, but also removes the AC safety ground. The system now relies upon the audio cable to provide the ground, a practice that can be hazardous. In fact, this type of ground loop will not automatically cause noise, as stated previously, unless the equipment is unbalanced or improperly grounded internally.

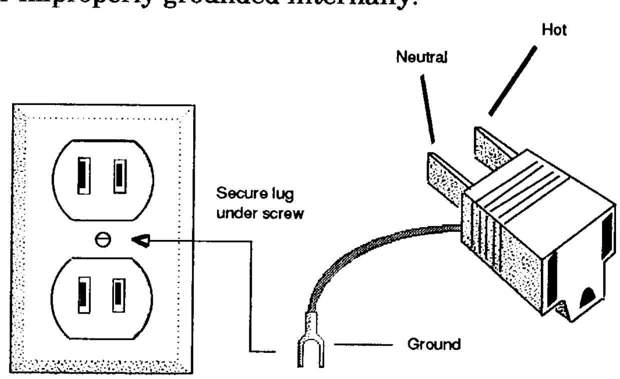


Figure 4-5. Avoid Use of AC Power Cord Ground Adaptor to "Break Ground;" Connect Green Wire to Outlet Box.

Here are some suggestions to minimize the safety conflict while avoiding noise caused by ground loops:

1. Don't lift the safety ground on any piece of equipment unless it significantly reduces the noise level.
2. NEVER defeat the AC safety ground on your console or any other piece of equipment connected directly to your microphones. Microphones take priority in grounding safety because they are handled by performers (who may touch other grounded items, including a wet stage).
3. Where practical, plug all affected equipment into the same AC service "leg." This includes the console, signal processors, and electric instruments such as guitar amps, keyboards, etc. This not only reduces the noise potential if a ground loop occurs, it also reduces the danger of electric shock. Lighting, air conditioning, motors and so on should be connected to a completely different "phase" or "leg" of the main power distribution system.

4.3.3. Basic Grounding Techniques

We will discuss four basic approaches to handling grounds within audio systems: single-point, multiple-point, floating, and telescoping shield. Each has specific advantages in different types of systems.

Figure 4-6a illustrates the single-point grounding principle. Chassis ground in each individual component is connected to earth; signal ground is carried between components and connected to earth at one central point. This configuration is very effective in eliminating line-frequency hum and switching noise, but is most easily implemented in systems (or subsystems) that remain relatively fixed. Single-point grounding is very often used in recording studio installations; it is also effective in the wiring of individual equipment racks. It is almost impossible to implement in complex, portable sound reinforcement systems.

Multiple-point grounding is shown in Figure 4-6b. This situation is common in systems that use unbalanced equipment having the chassis connected to signal ground. It has the advantage of being very simple in practice, but it is not very reliable – particularly if the connection configuration of the system is changed frequently. Multiple-point ground systems which include unbalanced equipment are inherently rife with ground loops. Hum and noise problems consequently can appear and disappear unpredictably as pieces of equipment are inserted or removed; when they appear, problems are very difficult to isolate and fix. Multiple point ground systems that employ balanced circuits with properly designed equipment may present no special noise problems.

Figure 4-6c shows the floating ground principle. Note that signal ground is completely isolated from earth. This scheme is useful when the earth ground system carries significant noise, but it relies on the equipment input stages to reject interference induced in cable shields.

The principle of telescoping shields is illustrated in figure 4-6d. This scheme is very effective in eliminating ground loops. Furthermore, if shields are connected only to earth, unwanted signals that are induced in them can never enter the signal path. Balanced lines and transformers are required to implement this approach, since ground is not carried between components. One drawback is that cables may not all be the same – some having shields carried through at both ends, and others not, depending on the equipment – so it becomes more complicated to sort out the cabling upon setup and breakdown of a portable system.

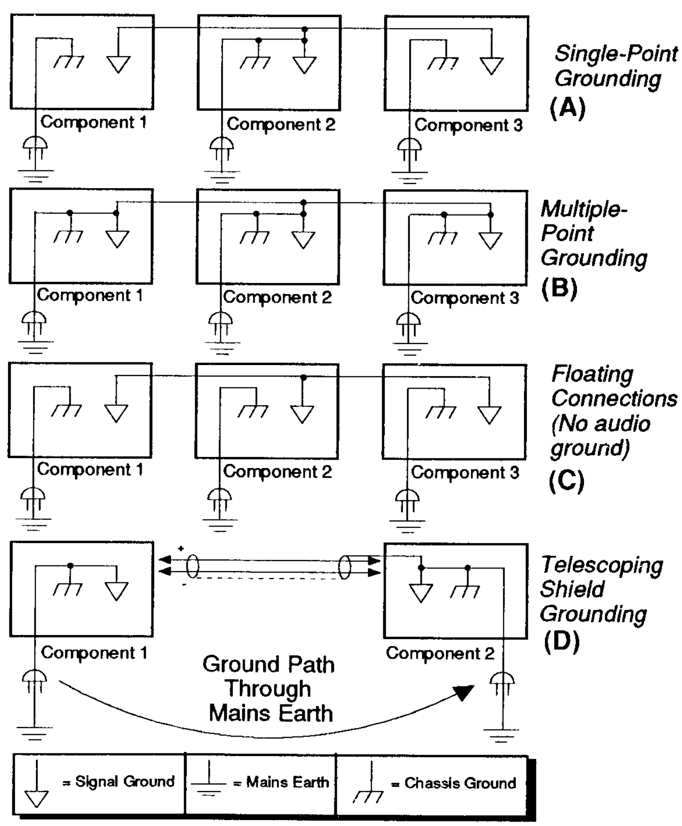


Figure 4-6. Standard Grounding Techniques

Figure 4-7 illustrates a typical audio system in which various grounding techniques are combined. The basic rules that guide the choice of grounding schemes may be summarized as:

1) Identify separate sub-systems (or equipment "environments") that may be contained within an electrostatic shield which drains to earth.
2) Connect signal ground within each separate sub-system to earth at one point only.
3) Provide maximum isolation in connections between sub-systems by using transformer-coupled floating balanced connections.

4.3.4 Balanced Lines, Unbalanced Lines, and Ground Lift Switches

In certain situations you can lift (disconnect) the shield at one end (usually at the output) of an audio cable and thus eliminate the most likely path that carries ground loop currents. In a balanced line, the shield does not carry audio signals, but only serves to protect against static and RFI, so you can disconnect the shield at one end without affecting the audio signal on the two inner conductors of the cable, and with little or no effect on the shielding. Unfortunately, this is not a very practical solution to the ground loop problem for portable sound systems because it requires special cables with shields disconnected on one end. Fortunately, some professional audio equipment is equipped with ground lift switches on the balanced inputs. Ground lifting can be used when multiple unbalanced audio cable join two pieces of equipment; in this case, all but one of the shields can be lifted, thus maintaining the "low" side of the audio connection without unnecessary duplication, which avoids

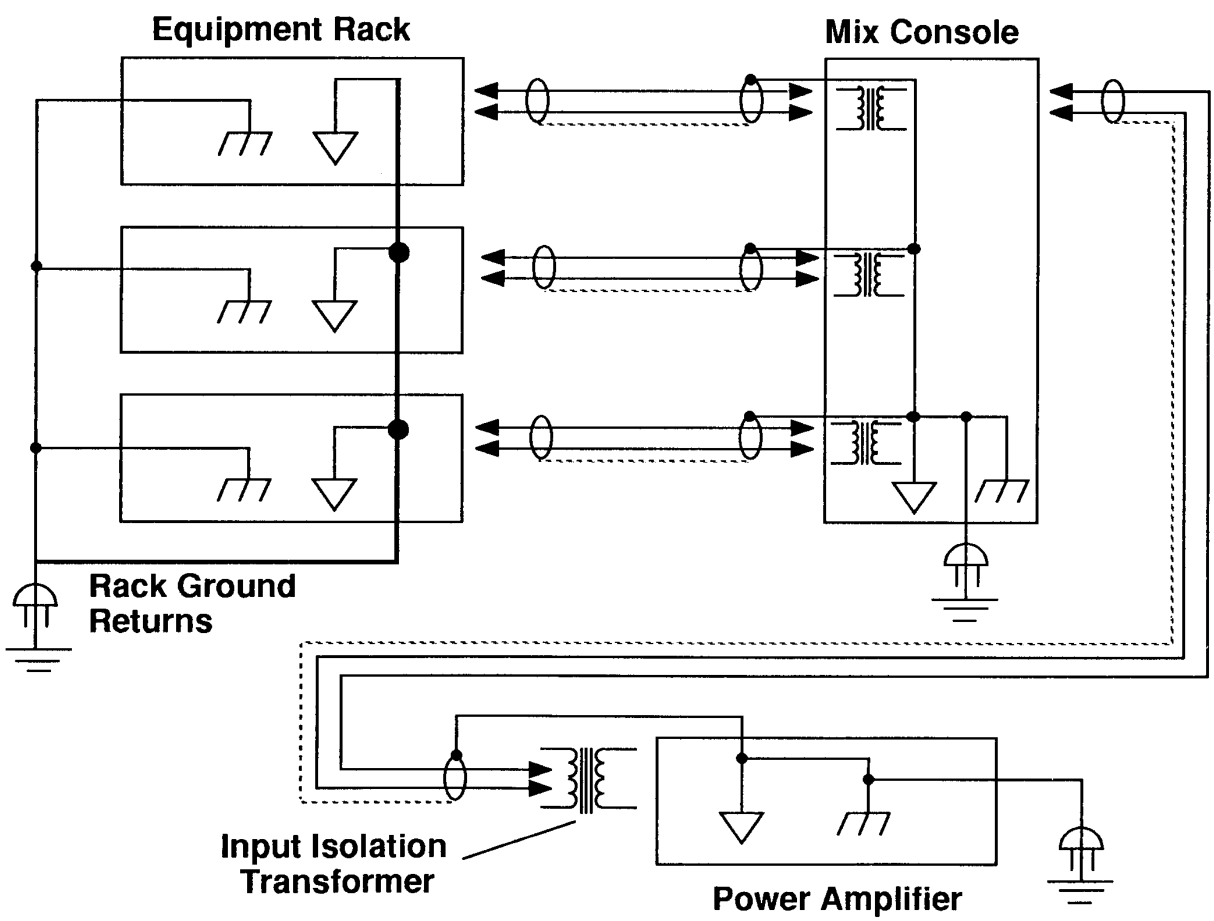


Figure 4-7. Combination of Grounding Techniques in a Typical Audio System

the ground loops and induced noise. If you want to avoid the ground lifting, try tightly bundling the cables.

CAUTION: Microphone cases typically are connected to the shield of the cable, and the shield is tied to the console chassis via pin 1 of the XLB connector. If there is any electrical potential on any external equipment, such as a guitar amp chassis, then a performer who holds the mic and touches the other equipment may be subject to a lethal electrical shock! This is why you should avoid "ground lift" adaptors on AC power connections if there is any other way to eliminate a ground loop.

In those audio devices which anticipate ground loops by providing "ground lift" switches next to XLRs or three-wire phone jacks, the ground lift switch makes and breaks the connection between the connector's shield and the chassis of the particular device. Ground lift switches are usually found on "direct boxes", which are used when an electric musical instrument is to be plugged directly into a console whose inputs are not designed to accommodate direct connection of such instruments (a direct box also includes a transformer and/or isolation amplifier, as discussed in Section 4.5).

One of the best ways to exclude noise from a microphone input is to use a high-quality, low-impedance microphone and to connect it to the console's low-impedance, balanced (or "floating") input. Use high-quality microphone cables fitted with XLR connectors, and keep microphone cables as short as possible. Also, physically separate mic cables from line-level (console output) cables, speaker cables and AC cables.

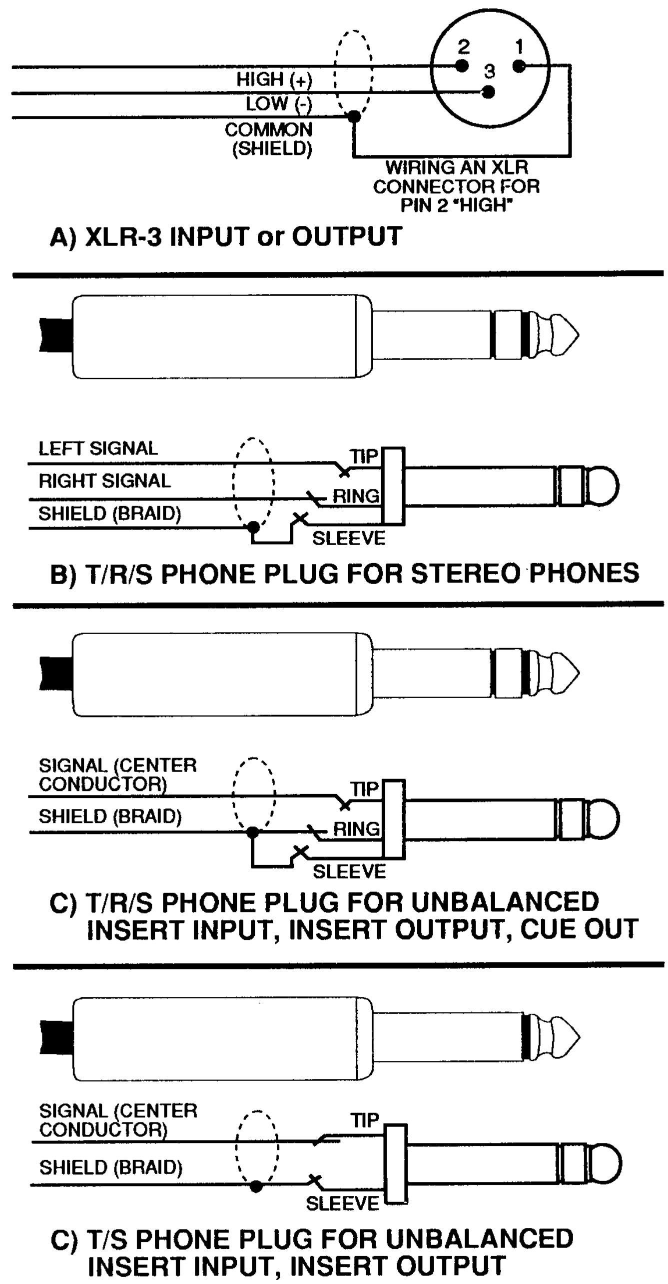


Figure 4-8. Connector Wiring for PM2800M

4.4 AUDIO CONNECTORS AND CABLES

The signal-carrying cables in a sound system are as much an audio "component" as any other part of the system. Improper cables between the equipment can result in exaggerated or deficient high frequency response, degradation of signal-to-noise ratio, and other problems. Use of the proper cables is essential if the full potential of high quality sound equipment is to be realized.

The PM2800M is fitted with only two types of audio connectors: 3-pin XLRs, both male and female, and 3-circuit (tip/ring/sleeve) 1/4" phone jacks (also known as stereo phone jacks, although their function is generally to carry an unbalanced mono signal rather than a stereo signal).

4.4.1 Types of Cable to Use

2-conductor (twisted pair) shielded cable is best for all XLR connections. Belden 8412, Canare L4E6S, or an equivalent are excellent choices due to their heavy duty construction, multiple strands that avoid breakage, good flexibility, and good shielding. Such cables are suitable for all portable applications, and for microphones. For permanent installation or for cables confined to portable racks or cases, a lighter duty cable such as Belden 8451, Canare L-2E5AT or an equivalent are suitable. "Snake" type multi-core cables containing multiple shielded pairs must be

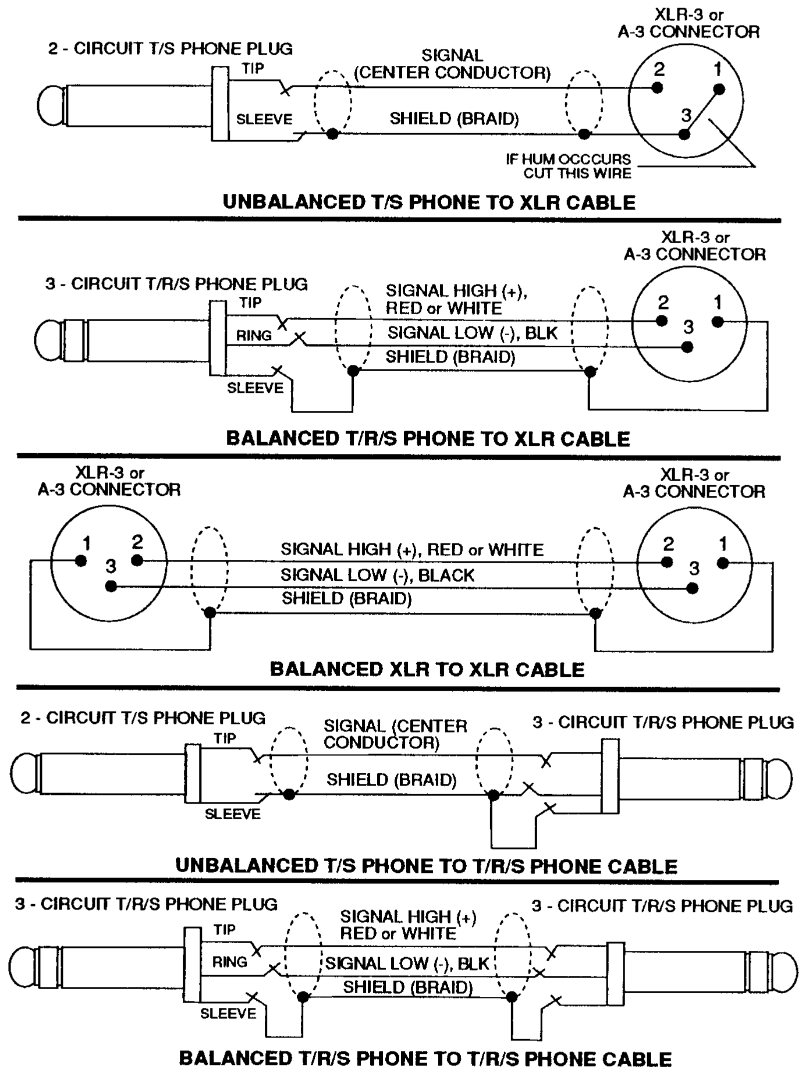


Figure 4-9. Cables for Unbalanced and Balanced Lines.

handled very carefully because the leads tend to be fragile, and a broken conductor cannot be repaired. If you are using a "snake," allow at least one or two spare channels that can be used in case of breakage in one of the channels in use.

4.4.2 Cable Layout

Never run AC power lines in the same conduit, or even closely bundled, with audio cables. At the very least, hum can be induced from the relatively high voltage AC circuits into the lower voltage audio circuits. At worst, a fork lift or other object rolling or dropped across the cables could cut through insulation, shunt the AC into the audio cable, and instantly destroy the audio equipment. Instead, separate AC and audio lines by as wide a distance as is practical, and where they must cross, try to lay them out to cross at as close to a right angle as possible.

Similarly, avoid closely bundling the line-level outputs from the PM2800M with any mic-level inputs to the console. Specifically, avoid using a single multi-core "snake" cable for running mic lines from the stage and power amp feeds up to the stage. The close proximity of such cables promotes inductive and/or capacitive coupling of signals. If the stronger output signal from the console "leaks" into the lower-level mic or line feeding a console input, and that weaker signal is amplified within the console, a feedback loop can be established. This will not always be manifest as audible "howling," but instead may be manifest as very high frequency (ultrasonic) oscillation that indirectly causes distortion of the signal and that can lead to premature component failure. The best solution is to widely separate mic input cables from line-level output cables or, if not practical, to at least bundle them loosely.

For the same reasons that mic and line level cables should be separated, so, too, should speaker cables (the cables run between the power amp output and the speakers) be separated from mic or line level cables. If speaker cables cross other audio cables, they should do so at right angles. If they must be run along the same path, they should not be bundled tightly.

4.4.3 Balanced Versus Unbalanced Wiring

In a general sense, there are two types of signal transmission systems for low to medium level audio signals: the balanced line, and the unbalanced line. Either type can be used with high or low impedance circuits; the impedance of a line bears no necessary relationship to its being balanced or not.

The unbalanced line is a "two-wire" system where the shield (ground) acts as one signal-carrying wire, and the center (hot) wire enclosed within that shield is the other signal-carrying wire.

The balanced line is a three-wire system where two signal wires carry an equal amount of potential or voltage with respect to the shield (ground) wire, but of opposite electrical polarity from each other. The shield (ground) in a balanced line does not carry any audio signal, and is intended strictly as a "drain" for spurious noise current that may be induced in the cable from external sources.

The shield in balanced and unbalanced cables is typically a shell made of fine, braided wires, although some cables have "served" (wrapped) shields or foil shields instead.

Balanced wiring is more expensive to implement than unbalanced wiring. It is often used, however, because it offers useful advantages, especially in portable sound systems. There is nothing inherently "better" or more "professional" about balanced wiring; the application dictates whether one system or the other is appropriate.

Unbalanced wiring works best when high-quality cable is used, the cable extends over relatively short distances, and one leg of the AC power system feeds all the equipment. Unbalanced wiring is often used for radio and TV signal transmission, computer data transmission, and laboratory test equipment.

Balanced wiring helps eliminate some types of externally-generated noise. The two wires of the "balanced" cable carry the same signal, but each wire is opposite in signal polarity to the other. In a balanced input, both of the signal-carrying wires have the same potential difference with respect to ground (they are "balanced" with respect to ground), and the input is designed to recognize only the difference in voltage between the two wires, and (hence the term "balanced differential input"). Should any electrostatic interference or noise cut across a balanced cable, the noise voltage will appear equally – with the same polarity – on both signal-carrying wires. The noise is therefore ignored or "rejected" by the input circuit. (This is why the term "common mode rejection" applies; signals in common to the two center wires are rejected.)

Not all balanced wiring has a shield. In older telephone systems, many miles of cable were run with no shielding in order to save money (now fiber optic cables are replacing costly copper with inexpensive glass or plastics). Out in the open, wires are subjected to radio interference and to hum fields emitted by power lines. Balancing the two signal hot wires with respect to ground gives long lines immunity to external interference. Twisting two wires together theoretically subjects each wire to the same amount of electrostatic or electromagnetic noise. A balanced input will then cancel the unwanted noise signals common to both wires, while passing the desired audio signal, as illustrated in Figure 4-10 (next page).

The RFI (radio frequency interference) cuts across both conductors, inducing equal voltages in the same direction. These voltages "meet" in the differential amplifier (or transformer), and cancel out, while the signals generated by the microphone flow in opposite directions in each conductor, and hence do not cancel out. Thus, in a theoretically perfect balanced system, only the desired signal gets through the differential amplifier or transformer.

4.4.4 The Pro's and Con's of Input Transformers

As illustrated, there are two means to achieving a balanced input; either with a transformer or with a differentially balanced amplifier (an "electronically balanced input"). The latter approach is used in the PM2800M, and was chosen for several reasons: (1) it is more "transparent" sounding than most transformer inputs, (2) it cannot be saturated by low frequency, high-level signals as can a transformer, (3) it is lighter in weight.

There are a number of reasons why input transformers are used in some installations. In the case of certain audio equipment which has an unbalanced input (not this console), a transformer converts the unbalanced input to a balanced input. Beyond that, there are cases where a transformer is desirable even if the input is electronically balanced. For example, where there is a significant amount of electrostatic or

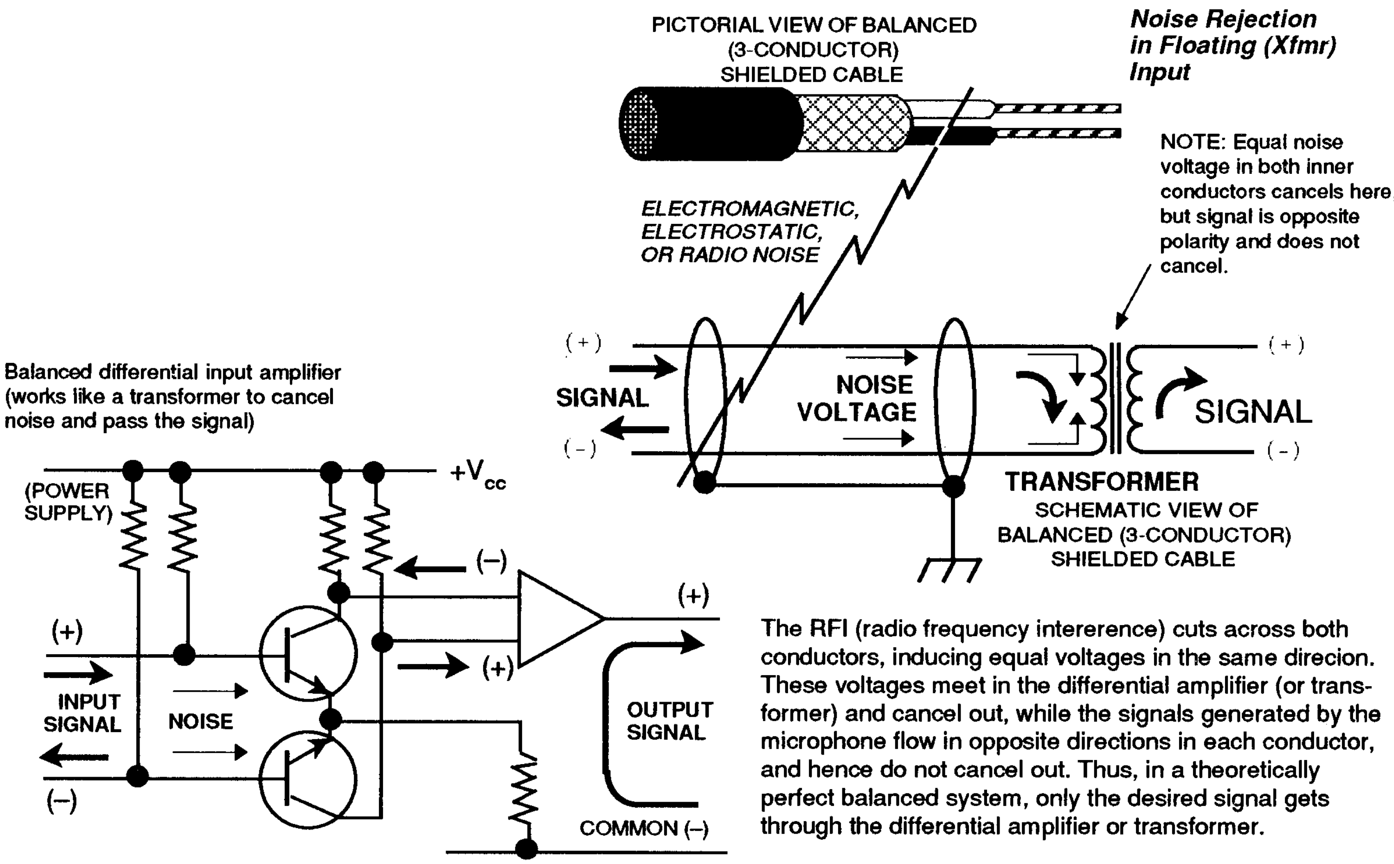


Figure 4-10. Noise Rejection in a Balanced Line.

electromagnetically induced noise, particularly high-frequency high-energy noise (the spikes from SCR dimmers, for example), the common mode voltage (CMV) of an electronically balanced input may be insufficient to handle the noise induced in the cable. In such cases, when the noise voltage exceeds the "rails" of the power supply in the electronic input, the common mode rejection ratio (CMRR) drops drastically, and noise comes right through the now unbalanced circuit. In these cases input transformers can be essential tools for eliminating the noise.

Some people mistakenly believe that an electronically balanced input compromises safety. They suggest that if a performer is touching a mic and also touches an electrically "hot" item such as a guitar which is "live" due to a fault in the guitar amp, AC current will flow through the performer. This is true. The fallacy is the belief that isolating the mic from ground, via an input transformer, can eliminate the shock hazard. This is not so because the mic case is connected to the cable shield, which is generally grounded to the console chassis, and thus the transformer-balanced input does not prevent shock. Instead, grounding isolation to eliminate this type of shock hazard is best done by using an AC power isolation transformer at the input to the guitar amplifier (or for the entire sound system). In reality, the transducer capsule in a microphone is generally isolated and insulated from the mic case, so an electronically balanced input still would not permit a current to flow through the mic — assuming everything is wired correctly in the microphone. If an audio input transformer is used in this situation primarily for ground isolation and to obtain the benefits of a balanced line, it is said to be an "isolation" transformer.

If the transformer is also used to prevent a low impedance input from overloading a high impedance output, it is known as a "bridging" transformer (not to be confused with the "bridged" connections of a stereo power amp output in mono mode).

In general, the PM2800M has no need for input transformers since it already has electronically balanced inputs. In the occasional instances where high common mode voltage exists, as previously described, there is no viable substitute for a transformer, and an optional input transformer kit (Model IT1800) can be installed in individual input modules. Similarly, PM2800M outputs can be transformer isolated by purchasing one or more optional output transformer sets. The Model OT1800 output transformer set contains 4 transformers, and the Model OT3000 contains 8 transformers; each set includes XLR connectors in a compact package that is external to the PM2800M. In this way, those inputs or outputs which require a transformer can be so equipped, and it is not necessary to pay the price, carry the weight or incur the slight performance penalty that comes with the transformers.

NOTE: There are other ways to achieve isolation. The most common means is with a wireless radio mic. One can digitize the audio signal and transmit it by means of modulated light in fiber optics, but this is much more expensive than using a transformer, with no great performance advantage. One can use the audio signal to modulate a light, and pick up the light with an LDR (light dependent resistor), thus achieving isolation at the expense of increased noise and distortion. Some systems, such as those for hearing impaired theatre goers, even do this over 10 to 100 foot distances using infra-red LEDs for transmitters and infra-red sensing photo sensors for receivers. The guitarist who places a microphone in front of the guitar amp speaker, rather than plugging a line output from the guitar amp into the console, has achieved electric isolation between the guitar and console by means of an acoustic link.

4.4.5 Noise and Losses in Low and High Impedance Lines

The length and type of cable can affect system frequency response and susceptibility to noise. The impedance of the line has a major influence here, too.

Signal cables from high impedance sources (actual output impedance of 5000 ohms and up), should not be any longer than 25 feet, even if low capacitance cable is used. The higher the source impedance, the shorter the maximum recommended cable length.

For low impedance sources (output impedances of 600 ohms or less), cable lengths of 100 feet or more are acceptable. For very low impedance sources of 50-ohms or less, cable lengths of up to 1000 feet are possible with minimal loss.

In all cases, the frequency response of the source, the desired frequency response of the system, and the amount of capacitance and resistance in the cable together affect actual high frequency losses. Thus, the cable lengths cited here are merely suggestions and should not be considered "absolute" rules.

Susceptibility to noise is another factor which affects cable length. All other factors being equal (which they seldom are), if a given noise voltage is induced in both a high impedance and a low impedance circuit, the noise will have a greater impact on the high impedance circuit. Consider that the noise energy getting into the cable is more-or-less constant in both instances. The low impedance input is being driven primarily by current, whereas the high impedance input is being driven primarily by voltage. The induced noise energy must do more work when it drives a lower impedance, and because the noise does not have much power, less noise is amplified by the input circuit. In contrast, the induced noise energy is not loaded by a high impedance input, so it is amplified to a greater degree.

4.5 DIRECT BOXES

The so-called "direct box" (or "DI box," where "DI" is an abbreviation for "Direct Injection") is a device one uses to overcome several of the problems that occur when connecting electric guitars and some electronic keyboards to a mixing console.

By using a transformer, the direct box can provide grounding isolation to protect a guitarist from inadvertent electrical shock in the event of a failure in the guitar amplifier or other equipment's power supply. The audible reason for using the direct box is that it matches the impedance of the instrument to that of the console input. Electric guitar pickups are very high impedance devices, and they are easily overloaded by anything less than a 100,000 ohm input termination. Connection of an electric guitar to the typical 600 to 10,000 ohm console input will cause a noticeable loss in signal level and degradation of high frequencies. While the impedance and level mismatch is less of a problem with electronic keyboards, such instruments often have unbalanced outputs which are, nonetheless, susceptible to hum and noise where long cables are required to reach the mixing console. To avoid these problems, a direct box can be connected near the instrument, and the output of the direct box then feeds the console.

NOTE: If a preamplifier-type instrument amp head (or a conventional amp having a preamp output) is used, a direct box may not be necessary since the head may provide a balanced, isolated output to a console.

One further application of the direct box is to isolate and pad the speaker-level output of an instrument amplifier so that signal can be fed to the console input. Normally, one would not connect a speaker-level signal to a console input. However, the reverb, chorus, tremolo, distortion, EQ, and other characteristics of many instrument amps are an integral part of the instrument's sound. If the amp head does not provide a line-level output for a console, then a suitably designed direct box can "tap" the speaker output for feed to the console. Even where a line level output is provided, sometimes the coloration of the signal at the speaker output (due to intentional clipping of the power amp section of the guitar amplifier, and back EMF from the speaker) is desired, and can only be obtained at the speaker terminals.

We present information in this manual for constructing an excellent passive type direct box; credit should be given to the designer, Deane Jensen, of North Hollywood, CA. While this design is believed to work well with the PM2800M, its inclusion in this manual does not represent an endorsement by Yamaha. The specified transformer is available from Jensen Transformers, Inc., 10735 Burbank Blvd., North Hollywood, CA 91601. FAX (818) 763-4574, Phone (213) 876-0059.

4.5.1 Passive Guitar Direct Box

This direct box is not a commercial product, though it can be assembled by any competent technician. It can be used in three ways:

1. At the output of a standard electric guitar, without an amplifier (pad switch open, ground switch closed),
2. At the output of a standard guitar with a guitar amplifier also connected (pad switch open, ground switch open or closed),
3. At the output of a guitar or instrument amplifier (pad switched in, ground switch open or closed).

The filter switch, which only works when the pad switch is closed, simulates the high frequency roll off of the typical guitar amp speaker. Since clipping distortion in a guitar amp creates high frequency harmonics, the filter switch, by attenuating the high frequency response, also cuts distortion. The filter and pad, however, are optional and may be omitted if the box is to be used strictly between the guitar pickup and the console.

The transformer was designed specifically for use in a guitar direct box. When connected to a typical electric guitar pickup, and an XLB channel input on a PM2800M, the transformer reflects the optimum load impedance to both the guitar pickup and the mic preamp input. This preserves optimum frequency response and transient response. The transformer has two Faraday shields to prevent grounding and shielding problems that could cause hum in the PM2800M or the guitar/instrument amplifier. Place the ground switch in whichever position works best.

Assembly can be accomplished in a small metal box. During operation, keep the chassis of the box away from the chassis of any guitar/instrument amp or any other grounded object. If you decide to use a transformer other than the Jensen model JE-DB-E, it should have similar characteristics: an impedance ratio of 20K ohms (primary) to 150 ohms (secondary), dual faraday shields, very low capacitance primary winding, and full audio spectrum frequency response.

The 3k ohm resistor shown connected across the transformer secondary is installed when the direct box is used with a standard PM2800M electronically-balanced input. When paralleled with the 3k ohm impedance of the PM2800M electronic input, this brings the termination impedance to about 1.5k ohm, which produces better response in the direct box. The approximate load on the guitar (pad switched out) is 200k ohm. If the optional input transformers are installed in the PM2800M, then the input impedance of the console is 1k ohm, and the 3k ohm termination resistor on the direct box is not required. The approximate load on the guitar, in this case, is 133k ohm. (The actual termination impedance of the direct box input is 133k ohm with the pad out of circuit, or 11k ohm with the pad in circuit.) The pad should be used if the guitar has an active pickup, or if any active effects devices are connected betweeen the guitar and the direct box.

Each winding, each Faraday shield, and the transformer chassis shield have separate leads. Be sure to connect the transformer leads as indicated (absolute polarity of these transformers DOES make a difference to performance; *even if both* sides of the transformer are swapped in polarity, the sonic results will not be as good as when correct polarity is observed).

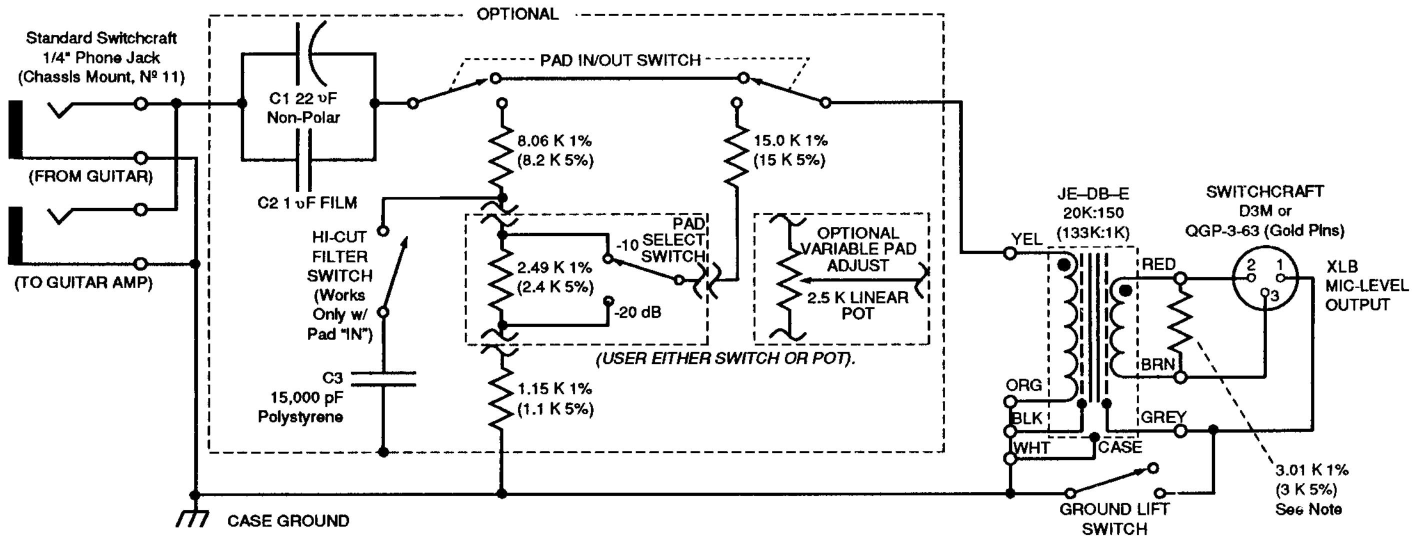


Figure 4-11. Passive Direct Box Schematic Diagram.

Additional Notes Regarding Figure 4-1:

1. C1 is a high quality, non-polar aluminum electrolytic, such as Roederstein type EKU. Voltage rating should be 25 V or higher. If non-polar cap is not available, use two 47μF, 25V polarized electrolytics in series, as shown on the Jensen JE-DB-E Data Sheet. Because of their high distortion, tantalum capacitors are not recommended for C1.
2. C2 is an optional high quality (polypropylene or polycarbonate) film capacitor used together with C1 to improve the sonic quality of the input capacitor.
3. C3 is a high quality (polystyrene or polypropylene) film capacitor. Adjust the value for the desired high-frequency rolloff (filter works only with pad in circuit).
4. Pad circuitry must always be used when the source is line or speaker level (synthesizer, guitar amp output, etc.).
5. 1% metal film resistors such as Roederstein (resista) MK-2 are recommended for their low noise and audio quality, although the nearest 5%, 1/4 watt carbon film (values shown in parentheses) will work with reduced accuracy.
6. Optional 2.5 kΩ linear taper potentiometer allows continuously variable attenuation between –10 dB and –20 dB. Conductive plastic is recommended, but carbon will work OK.
7. Pin 2 of the microphone-level output connector is "Hi", Pin 3 is "Lo", in order to comply with I.E.C. standards. This is compatible with Neumann, AKG, Beyer, Shure, Sennheiser, Crown, EV, and Shoeps microphones, all of which are Pin 2 "Hot."
8. 3 kΩ resistor across transformer secondary should be installed when the direct box is used with inputs having greater than 2 kΩ actual termination impedance (i.e., standard PM2800M input). It is OK to leave the resistor in circuit with 1 kΩ inputs (i.e., PM2800M input with optional isolation transformer installed), although better results will be obtained if the resistor is omitted in this case.
9. Parts kit DB-E-PK-1 containing all resistors and capacitors needed to build above circuit available from Jensen Transformers for nominal fee.

4.6 CONFIGURING EQUIPMENT RACKS

The great majority of audio equipment manufacturers make provision for their electronic products to be mounted in EIA standard 19 inch wide equipment racks. (The equipment may be only 17 to 18 inches in width, or even less; the "rack ears"that mount to the rack rails extend to 19 inches.) Panel heights for rack-mounting equipment are standardized on multiples of a single-unit space of 1.75 inches.

When selecting electronic equipment, it is important to bear in mind eventual rack mounting. Not only the height but also the depth of the unit should be considered. Particularly in portable applications, the integrity and strength of the front panel and/or rack mounting ears also must be examined in relation to the chassis weight. Heavy components such as power amplifiers should be supported at the rear as well, rather than relying only on the front rack ears. Even if a piece of equipment seems secure when you screw its front panel to the rack rails, the vibration and shock encountered in the back of a semi-trailer may quickly bend metal or break it right out of the rack.

Before actually mounting the selected components, it is wise to carefully plan out each rack with an eye to both signal flow, heat flow, and weight distribution. It might be best to mount together components that function as a group: the equalizer, active crossover and power amplifier for a single loudspeaker or array, for example. On the other hand, some prefer to mount all the equalizers for the system in one rack, all the power amplifiers in another, and so on. If you select the latter approach, however, you may find that the power amplifier racks are dangerously heavy. Also, if one "all the same" rack is damaged, you could be out of business, whereas loss of a "mixed" rack will only partially impair the system. It is far better to put some thought into such matters beforehand than to do all the work, and then correct mistakes after they cause major problems.

At its best, configuring equipment racks is a true craft combining a focus on practical utility and careful engineering with a concern for clean appearance. In a well-prepared rack, electronic devices are accessible yet protected, and are neatly and consistently mounted with proper hardware. Interior and exterior work lamps, integral power distribution, ground fault indication and a well-stocked spare fuse compartment are among the extra touches that are usually provided. Equipment that may generate strong electromagnetic fields (power amps with large transformers) should be separated from equipment that has high gain (microphone and phono cartridge preamplifiers or cassette decks).

The hallmark of a professional rack is the care that is taken with the internal wiring. Color coding and/or clear and logical cable marking facilitate troubleshooting and reflect an understanding of the electronic signal flow. Related groups of connections are neatly routed and bundled with cable ties. Audio signal cables are kept separate from power cords, and low-level signal cables are separated from high-level signal cables. Excess cable (including any service loop) is neatly stowed and tied down, and all connections are secured so that they stay in place in shipment.

Finally, touring sound professionals protect their equipment racks in foam-lined flight cases equipped with wheels and handles to facilitate handling. Given the considerable investment in equipment, materials and time that a fully-loaded rack represents, such protection is essential. Flight cases in standard sizes are available from a number of manufacturers, and it is generally not necessary or economical to make them yourself.

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	=	10.0 microwatts	=	0.00001 watts
0 dBm	=	1.0 milliwatts	=	0.001 watts
+4 dBm	=	2.5 milliwatts	=	0.0025 watts
+24 dBm	=	250.0 milliwatts	=	0.25 watts
+30 dBm	=	1000.0 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 useable 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.

5.2.3 A Discussion of Headroom

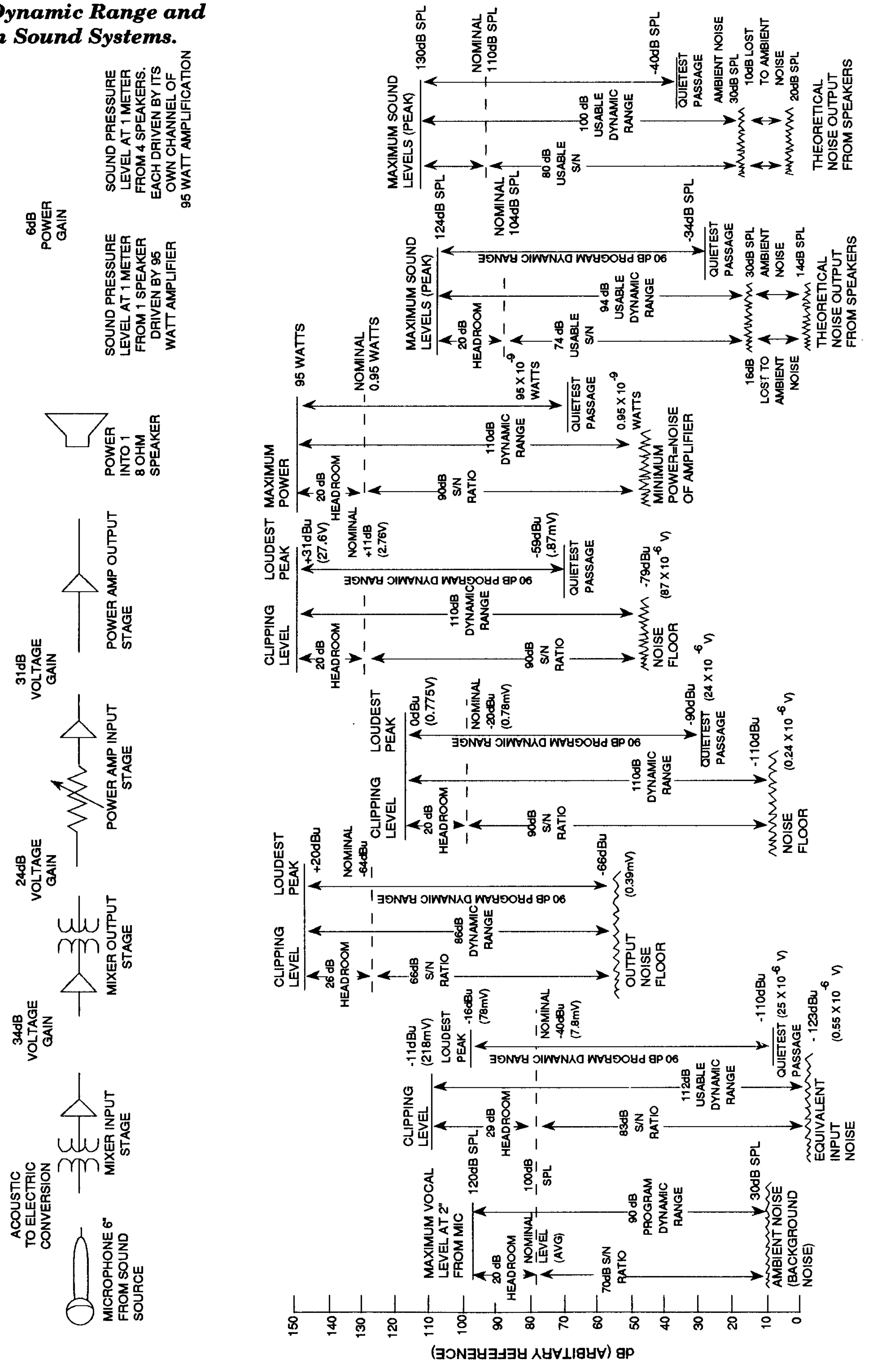
The average line level in the typical commercial sound system just described is +4 dBu (1.23 volts), corresponding to an average sound level of 100 dB SPL. This average level is usually called the "nominal" program level. The difference between the nominal and the highest (peak) levels in a program is 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. and dbx, Inc.). A compander

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



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.

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 overloading 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."

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 PM2800M meters (0 VU = +4 dBu, which allows 20 dB headroom before the console reaches its maximum +24 dBu output level).

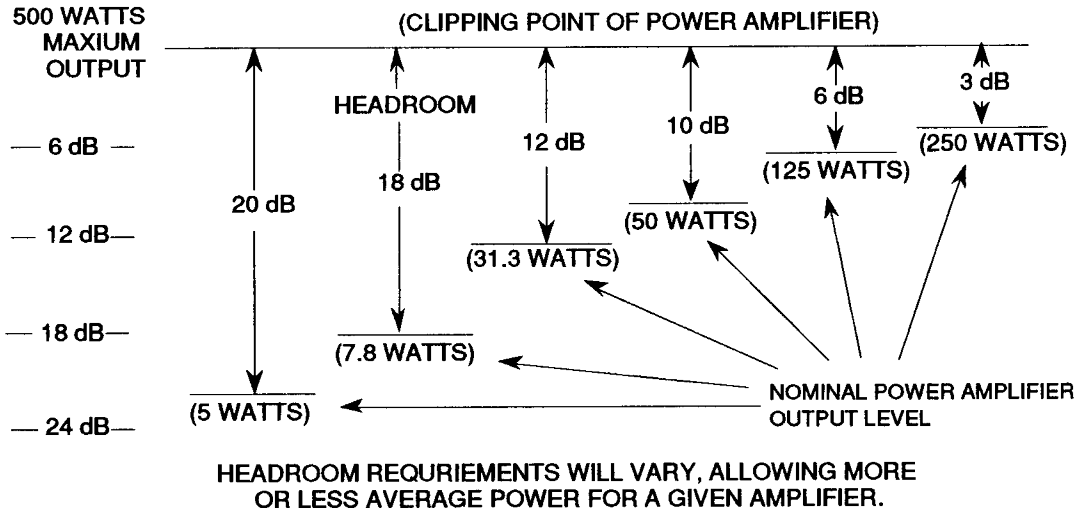


Figure 5-2. Headroom in Different Applications

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 level control on the console at its marked "nominal" setting, and adjust the master level control 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 levels hould 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 output per factory calibration).
3. Assume that the rated maximum input level for the graphic equalizer to which the console output is connected is +14 dBu (3.88 volts). Subtracting +4 dBu from +14 dBu leaves only 10 dB of headroom, so in order to maintain the desired 20 dB of headroom, a 10 dB resistive pad should be inserted between the console output and the equalizer input. The signal level at the input to the equalizer should now be -6 dBu (388 mV),which can be confirmed with a voltmeter.

   NOTE: If the graphic equalizer is inserted in the console's group or stereo INSERT IN/OUT loop, that signal level is already a nominal -6 dBu when the VU meters are at 0 VU, so no pad would be required.
4. 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 -6 dB, so the equalizer output can be reduced to that level, providing another 4 dB of headroom, which is a good idea anyway since it will allow for more EQ boost without overloading the equalizer output.
5. 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 or aux 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 PM2800M 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 PM2800M. 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 PM2800M 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 PM2800M 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 PM2800M 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 tape recorders is so great that all the headroom a console can provide is advantageous.

SECTION 6
OPTIONAL FUNCTIONS

The PM2800M 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 PM2800M, a number of optional functions have been built in, and can be selected by moving jumper wires within certain modules.

WARNING: Underwriters' Laboratories (UL) requires that we inform you there are no user-serviceable parts inside the PM2800M. 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; 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 internal jumpers 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

The modules in the PM2800M are designed for easy removal.

1. Turn the Power OFF first, before removing or installing a module.
2. Remove the screws which secure the meter bridge to the console, and tilt the bridge back to expose the tops oft he modules.
3. Remove the Philips head screws at the top and bottom of the module, but do not yet attempt to lift the module out.
4. Disconnect the connectors from the rear edge of the module to be removed. Input modules have two ribbon connectors(30 pin & 10 pin), and one Molex type low-profile connector. Other modules have two 30-pin ribbon connectors and one or two cable connectors.
5. Pull up gently on one or more control knobs, and as you feel the module connectors release, slide the module forward toward the front of the console slightly.
6. Now lift the module the rest of the way out of the console.
7. Installation of a module should be done by reversing the order of this procedure. Work slowly to make sure that edge connectors mate properly.

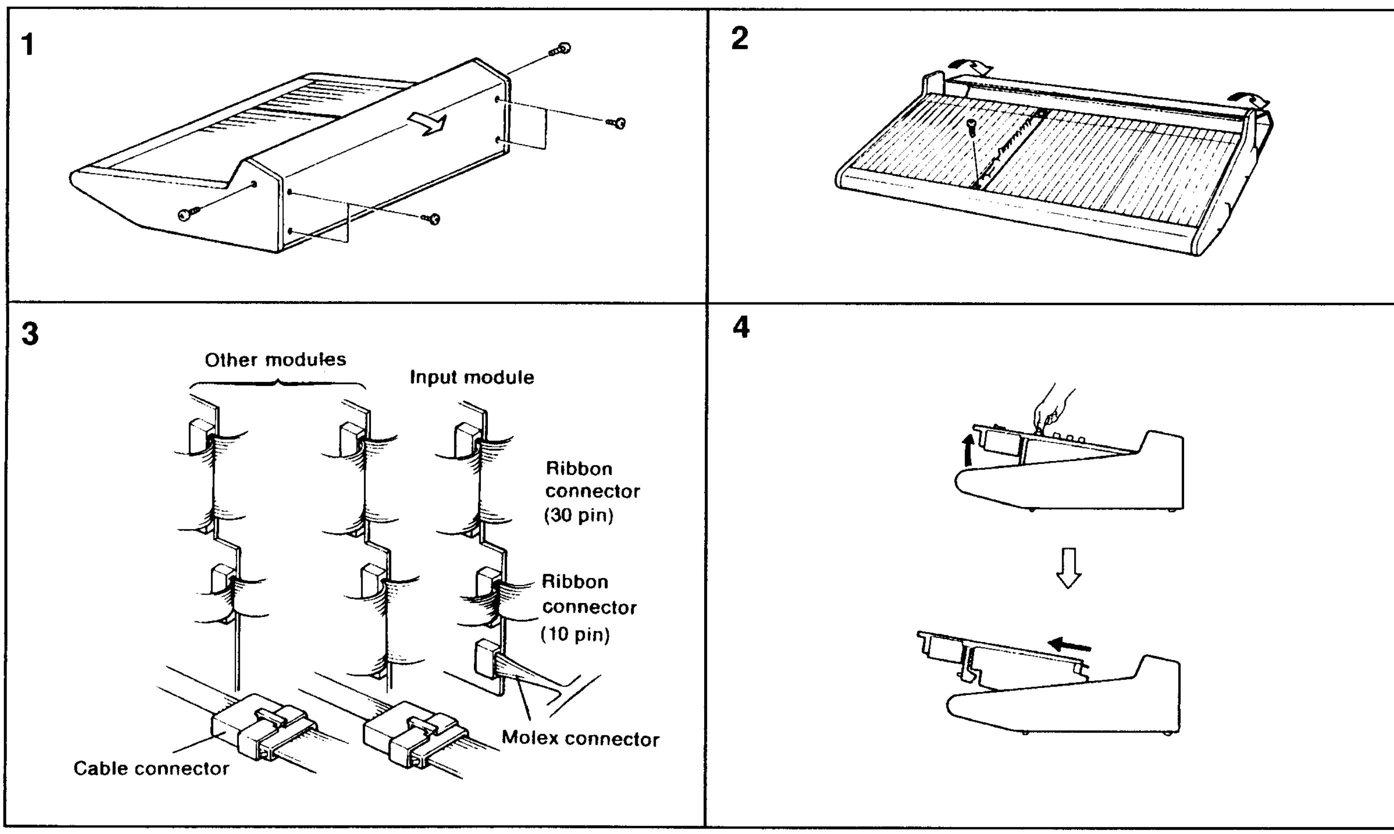


Figure 6-1. Removal of Module From PM2800M.

6.2 INPUT CHANNEL AUX SENDS: PRE OR POST... FADER, EQ & HP FILTER

As shipped, the console is wired so that the AUX 1 – 2 and AUX 3 – 4 send controls in each input module derive signal after the fader, equalizer and high pass filter. If desired, individual sends can be altered, by moving an internal jumper, so they are derived before ("Pre") the fader, EQ and filter.

Note that the "Pre" function can be altered to be post EQ & filter, but still pre fader, as described in Section 6.4.

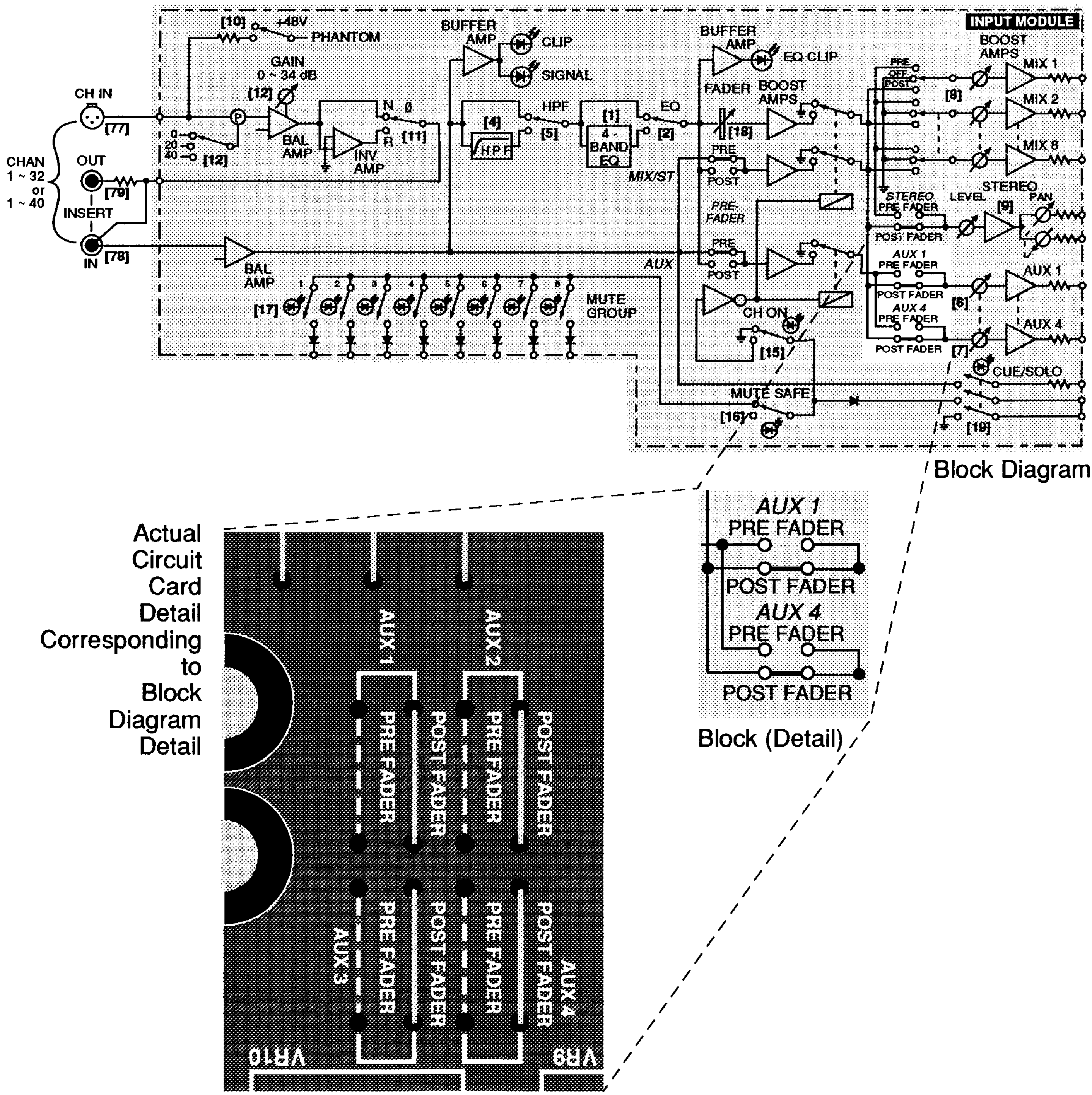


Figure 6-2. Internal Jumper Wiring for Pre or Post Fader, EQ & Filter Aux Sends

6.3 INPUT CHANNEL STEREO MIX: PRE OR POST... FADER, EQ & HP FILTER

As shipped, the console is wired so that the STEREO mix controls (PAN/LEVEL) in each input module derive signal after the fader, equalizer and high pass filter. If desired, individual channel stereo mixes can be altered, by moving an internal jumper, so they are derived before ("Pre") the fader, EQ and filter.

Note that the "Pre" function can be altered to be post EQ & filter, but still pre fader, as described in Section 6.6.

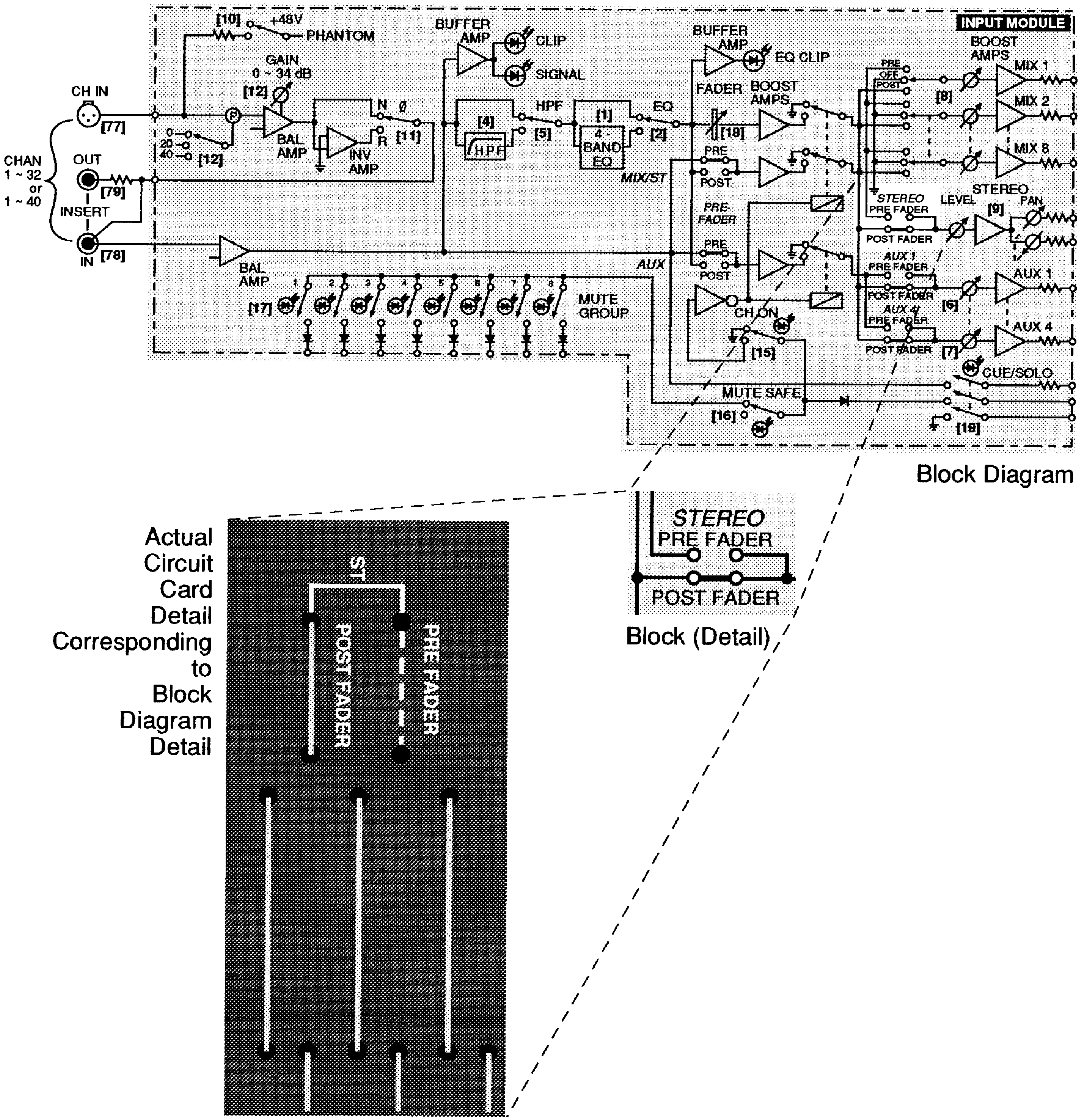


Figure 6-3. Internal Jumper Wiring for Pre or Post Fader, EQ & Filter Stereo Mix

6.4 INPUT CHANNEL AUX SEND PRE FUNCTION: PRE FADER & EQ OR PRE FADER/POST EQ

A jumper wire in each input module permits all four auxiliary sends to be altered. As shipped, the console is wired so that the AUX SEND function is post fader, EQ and high pass filter. This is generally useful for effects mixing, where fading the channel in the primary mix also fades the send to the effects devices (e.g., so a reverberation unit will not continue to feed reverb to the mix after the primary signal has been faded). Also, channel EQ and filtering will affect the signal going to the effects unit.

On the other hand, if the auxiliary sends are used for recording or remote mixes, where one does not want channel level changes which are made to satisfy stage requirements to alter the recording/remote mix, then a pre-fader setting is appropriate. If an aux mix is jumpered to be a pre-fader effects send, it may still be desirable to apply channel EQ and HP filter processing to that send. Jumpering individual Aux Sends to the POST function would achieve this goal, but also would cause the channel fader to affect those sends. To solve the problem, a single jumper can be moved so that the PRE function remains pre-fader, but is taken after the EQ and HP filter; *this jumper affects all four aux sends* on a given module.

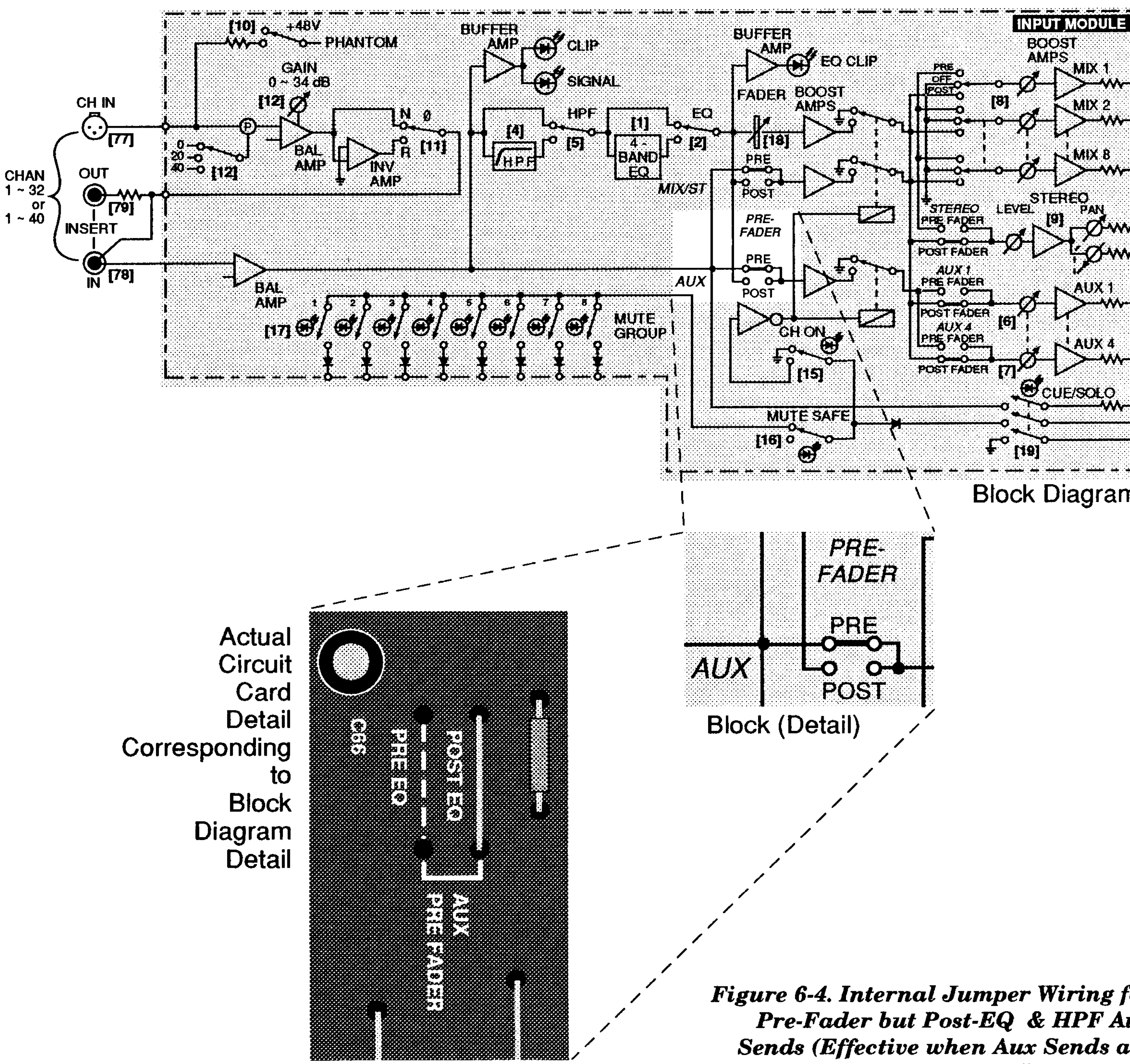


Figure 6-4. Internal Jumper Wiring for Pre-Fader but Post-EQ & HPF Aux Sends (Effective when Aux Sends are Jumpered to "Pre" Function).

6.5 INPUT CHANNEL PRIMARY MIX 1–8 & STEREO MIX PRE FUNCTION: PRE FADER & EQ OR PRE FADER/POST EQ

A jumper wire in each input module permits the function all eight primary mix controls and the stereo mix controls to be altered. As shipped, the console is wired so that the PRE position on the PRE/OFF/POST switch is pre fader, EQ and high pass filter. (Similarly, if the stereo send is jumpered to the "pre" function, it is pre fader, EQ & HPF.) In applications where it is desirable to apply channel EQ and HP filter processing to that mix, a single jumper can be moved so that the PRE function remains pre-fader, but is taken after the EQ and HP filter, *this jumper simultaneously affects all eight primary mix controls and the stereo mix controls* on a given module (provided they are set to the "pre" function).

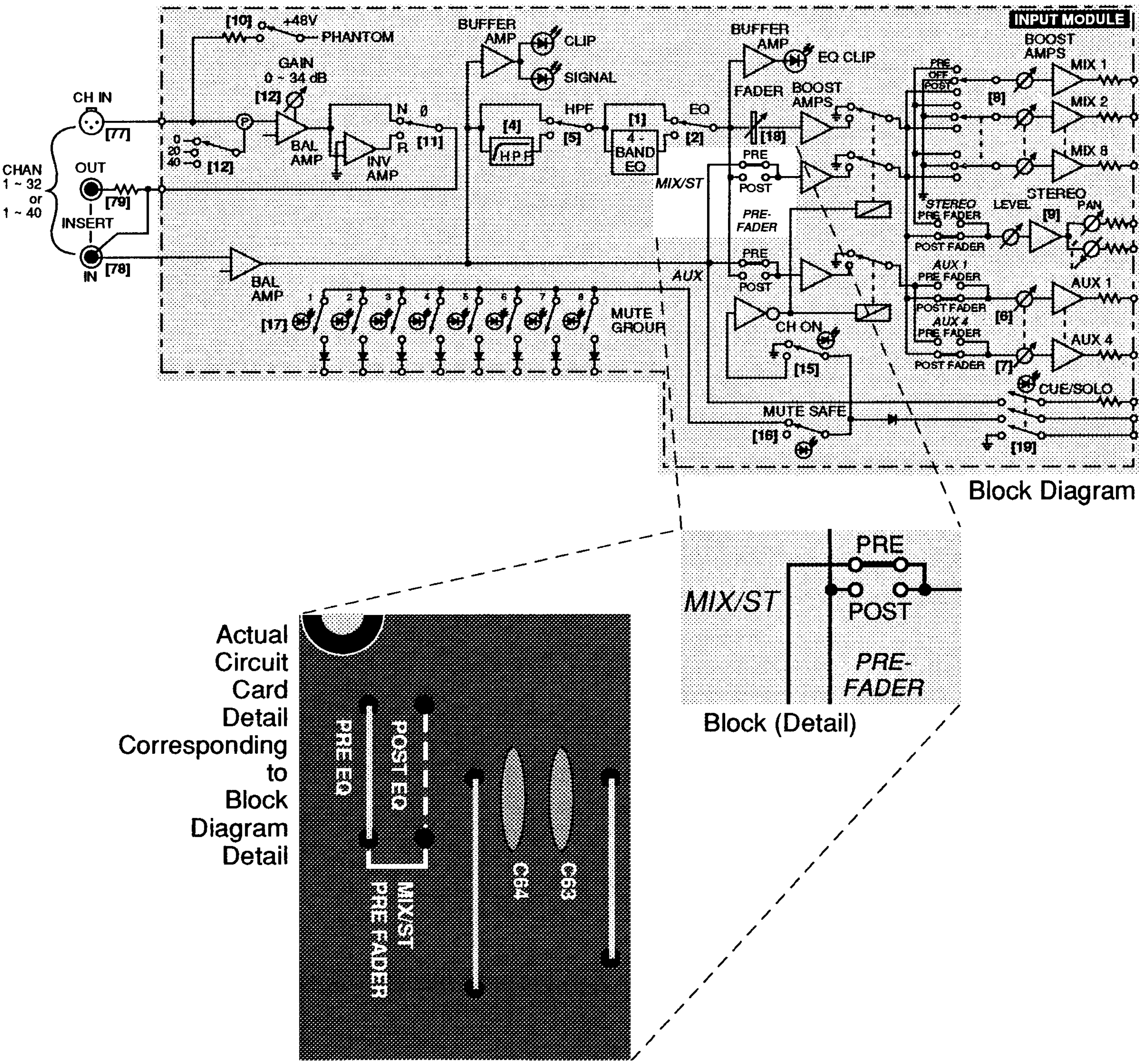


Figure 6-5. Internal Jumper Wiring for Pre-Fader but Post-EQ & HPF Primary & Stereo Mix Controls (Effective when Primary Mix Switches are Set to PRE, or Stereo Mix is Jumpered to "Pre" Function).

6.6 MIX-TO-MATRIX & ST-TO-MATRIX: ASSIGNED PRE OR POST MASTER FADER

A jumper wire in each Mix A and Mix B module permits the eight mix bus sends to the mix matrix to be altered. A similar jumper exists in the Stereo/TB module affecting the stereo bus send to the matrix. As shipped, the console is preset so that when the MIX-TO-MATRIX switch [32] or ST-TO-MATRIX switch [53] is on, the matrix is fed signal after the corresponding Master Fader [35 or 56] (but before the ON/off switch [34 or 55]). The internal jumper in each of these modules can be repositioned so that the matrix is fed before the Master Fader.

In the factory preset configuration, the matrix follows the primary mixing bus. If one mix bus, for example, is used for lead vocals, another for backup vocals, and other for keyboards, etc., then all lead vocals going to all matrix outputs can be adjusted with one Mix Master Fader... all keyboards going to all matrix outputs can be adjusted with another mix Master Fader, etc. Several matrix outputs can then be altered to provide similar, but not identical, remixes of these sources to suit different soloists. Suppose, however, that you plan to feed a stereo sidefill mix from the Stereo Bus, yet you need one or more additional stereo mixes for remote feed to a house console, a stereo keyboard monitor mix, etc.

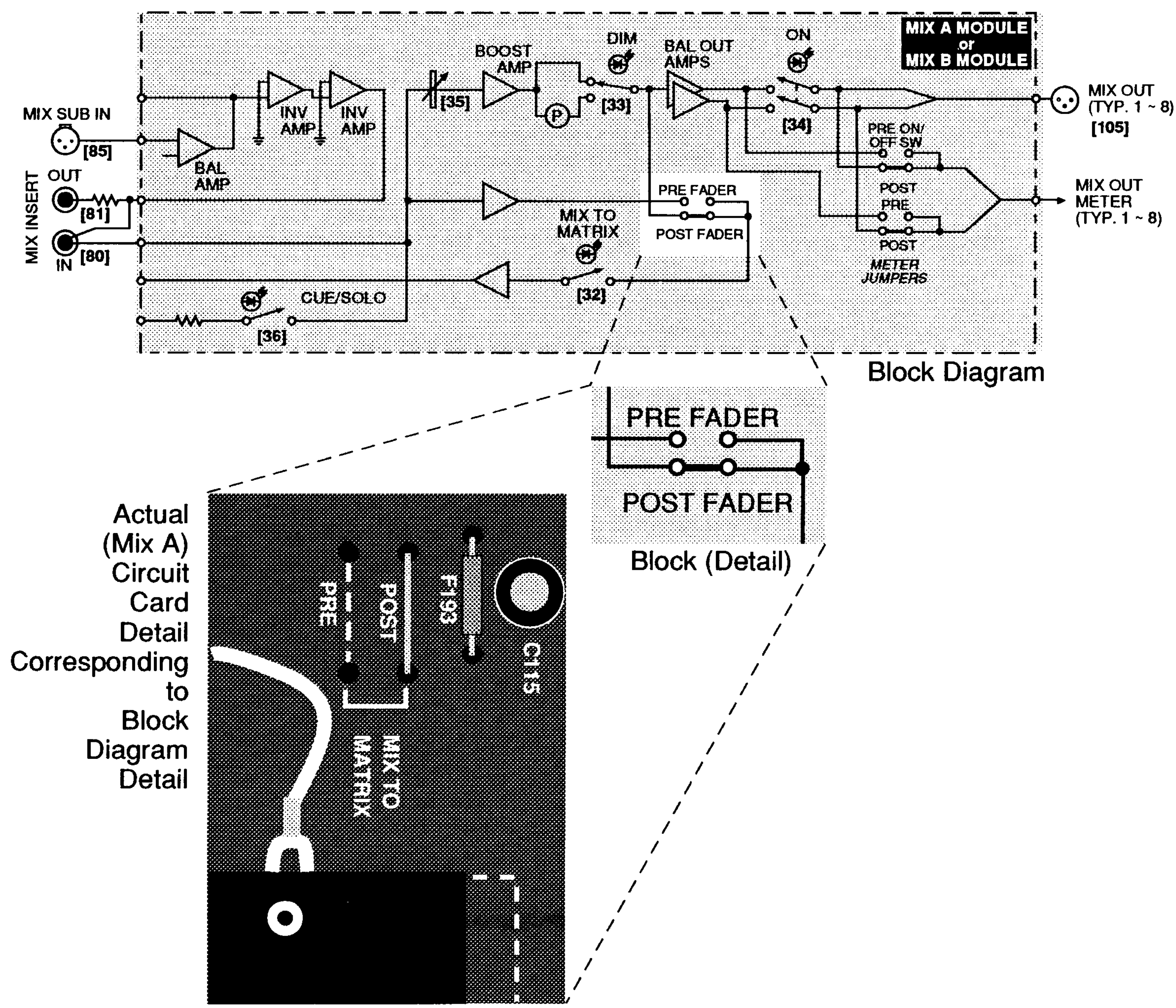


Figure 6-6. Internal Jumper Wiring for Pre- and Post- Mix Master Fader or Stereo Master Fader Feeds to Mix Matrix.

The Stereo Bus can be used to create the sidefill mix, but this mix will not be suitable for the other stereo requirements; even if it were, level changes required for one stereo output would cause problems with the others. Normally, the mix matrix alone can be used to create one stereo and two mono mixes, or two stereo mixes, but these will be affected by level changes in the Mix Master Faders [35] and/or the Stereo Master Fader [56], so the mixes will not be truly independent.

A greater number of independent output mixes can be obtained, however, by wiring the Mix A, Mix B and/or Stereo/TB modules to alternate jumper positions whereby their matrix sends are pre Mix Master Fader or pre Stereo Master Fader. You can then assign the Mix Bus outputs or the Stereo Bus outputs to the matrix without these signals being affected by the bus master faders. The matrix controls on each master module will provide a unique 10:1 mix of the same busses; that matrix channel's Nº 1 – Nº 8 mix controls [37] and STereo L & R controls [38] will serve as submasters, and the MATRIX MASTER control [39] will be the final determinant of the mixed output. In this way, you can obtain one additional stereo and two additional mono mixes, or two additional stereo mixes (or, for that matter, four additional mono mixes).

6.7 MIX OUT METER FUNCTION: PRE OR POST MIX ON/OFF SWITCH

There are eight VU meters which monitor the MIX output levels. Specifically, as factory wired, these meters normally monitor the Mix output after the Mix Master Faders [35] and ON/Off switches [34]. Thus, if a Mix Output is switched off, there will be no meter deflection. In some cases, you may wish to preview mix levels before turning on the output. A pair of internal jumpers on each Mix A and Mix B module permits you to do this for the individual Mix output meters.

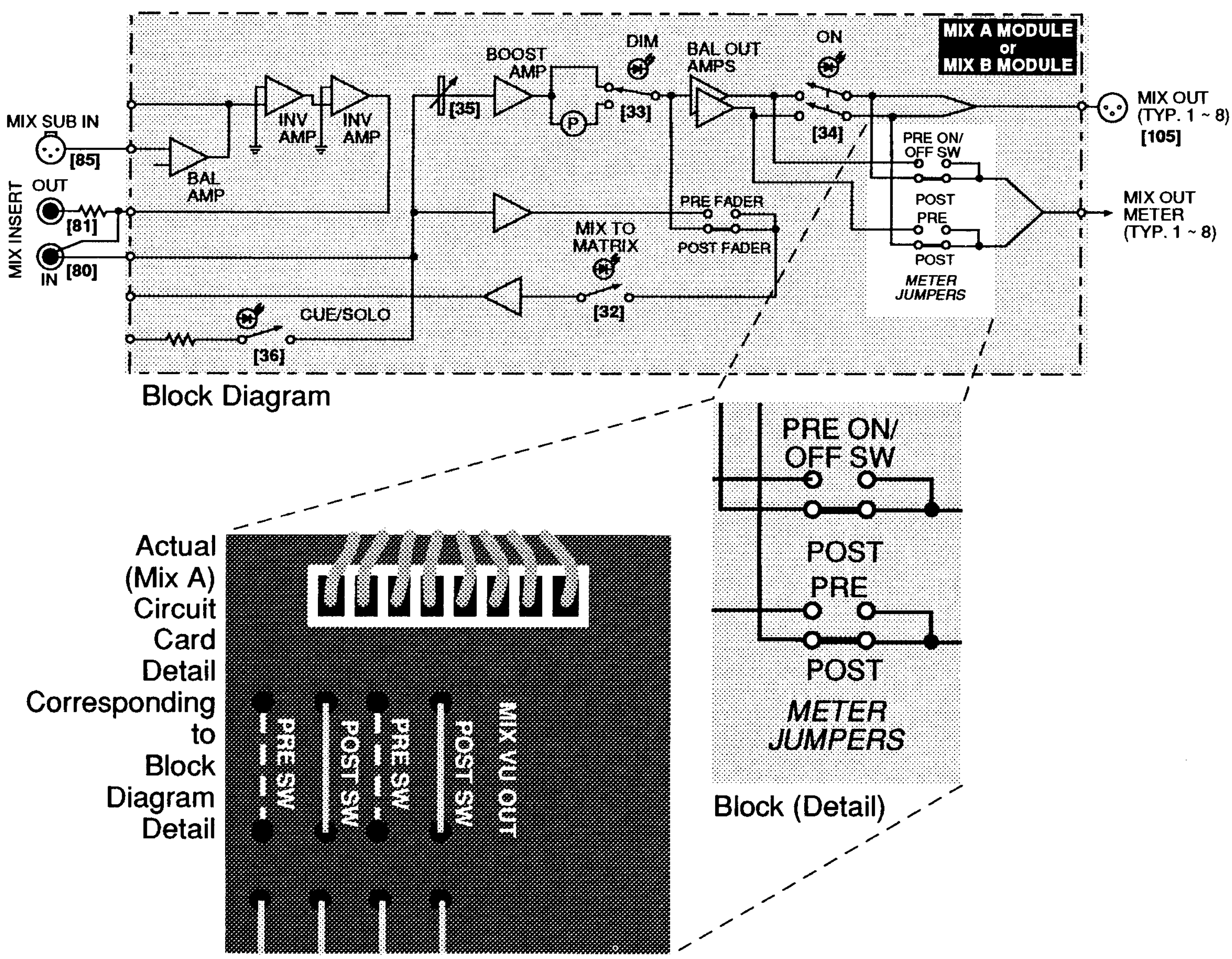


Figure 6-7. Internal Jumper Wiring for Mix Metering Before or After the Mix On/Off Switch.

6.8 INSTALLATION OF OPTIONAL INPUT TRANSFORMERS

The PM2800M standard input module is equipped with a balanced, differential input preamplifier for the XLB 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 modules main circuit board. Refer to Figure 6-8A.

An optional transformer balancing option may be installed by a Yamaha PM2800M 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 converted from the console mainframe.
3. Hold the module with the fader to the left, and lay the module on its side, controls facing away from you.
4. Remove the "IN1 4/4" board.
5. Install the new board (that comes wired to the transformer) in place of the "IN1 4/4" board.
6. Install the transformer by securing its bracket to the lower left edge of the module frame with the screw provided. Dress the cable that joins the transformer and its circuit board neatly. You may wish to tie it to the board so that after the module is reinstalled, the cable does not become pinched between modules or the module and mainframe. Refer to Figure 6-8B.
7. Reinstall the input module into the mainframe.

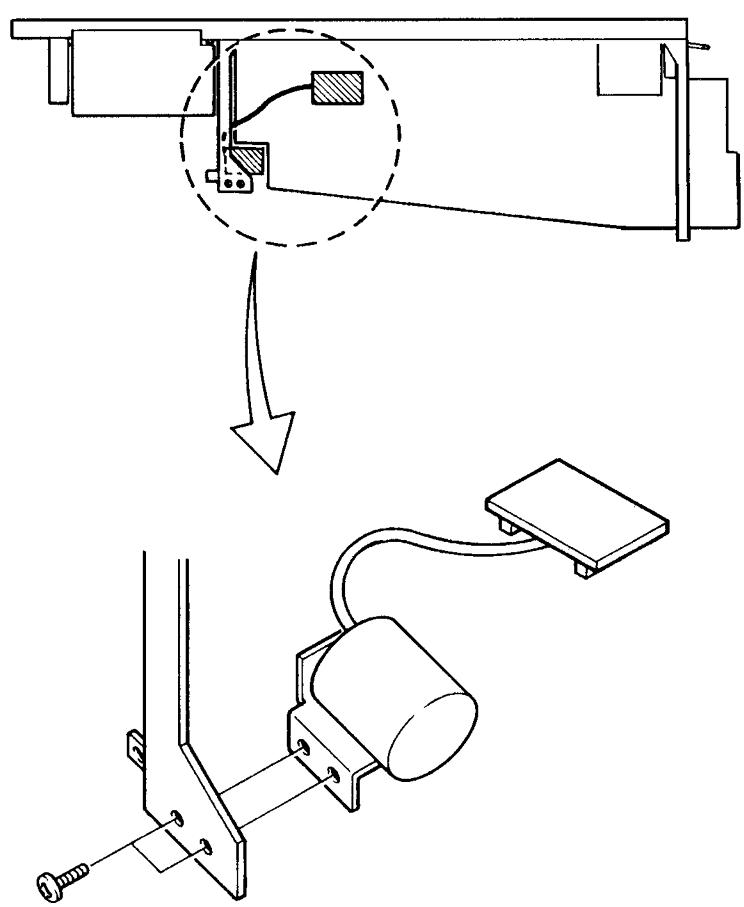


Figure 6-8. Optional Input Transformer Installation.

6.9 HINTS ON CIRCUITRY FOR REMOTE CONTROL OF THE MUTE GROUPS

The MUTE CONTROL connector on the PM2800M rear panel is provided primarily so that two consoles may be linked, and just one console's MUTE MASTER switches will affect both consoles input channels. However, it is possible to create an independent controller so that this function can be remoted from the console. One 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 information which will allow a competent technician to do the job.

The MUTE connections are illustrated in Figure 2-9. In order to mute a group, simply ground the conductor corresponding to that group. The console's MUTE MASTER/SLAVE switch must be set to the SLAVE position in order for the corresponding remote control to take effect. The MUTE connections not only permit remote muting by an external switching system, they also permit two PM2800M consoles, or a PM2800M and PM1800 or PM3000 to be interconnected so that one console's master mute switches can control both console's muting functions.

WARNING: Only qualified service technicians should attempt to construct and connect any circuit to interface with the PM2800M MUTE CONTROL connector. A circuit or wiring error could damage the console, and such damage is not covered under the terms of the PM2800M 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 PM2800M Service Manual should be consulted before actually building any remote control device.