FOUR DOLLARS

PM-1000

THE YAMAHA PERFORMERS

PM-1000

INTRODUCTION

The PM-1000 is a 4-bus mixing console with 16, 24 or 32 input channels. Yamaha engineers, in cooperation with soundmen, mixers and performers, designed this board to meet with widely varied demands of professional sound reinforcement, as well as studio applications. Modular construction and all solid state circuitry assure reliability and ease of maintenance.

To enhance its appearance, the console is finished with durable, black-anodized aluminum panels and housed in a handsome rosewood cabinet with a padded armrest. Carrying handles on the side panels and a padded leatherette cover are included as standard equipment.

Every console is delivered complete with a full complement of modules, and ready for use. Transformer isolated inputs and outputs, plus a precision, wide-range input sensitivity selector on each channel help simplify installation and optimize performance with virtually any type of audio system. Further convenience is afforded by the many XLR connectors, phone jacks and stereo phone jacks that carry all input and output signals.

The PM-1000 is built to exacting standards. It is light, yet rugged enough to sustain the kind of punishment that often occurs in portable applications. For fixed installations, the console has all the features and appearance of many larger, custom boards. Top of the line performance at an economical price sets the PM-1000 in a class by itself.

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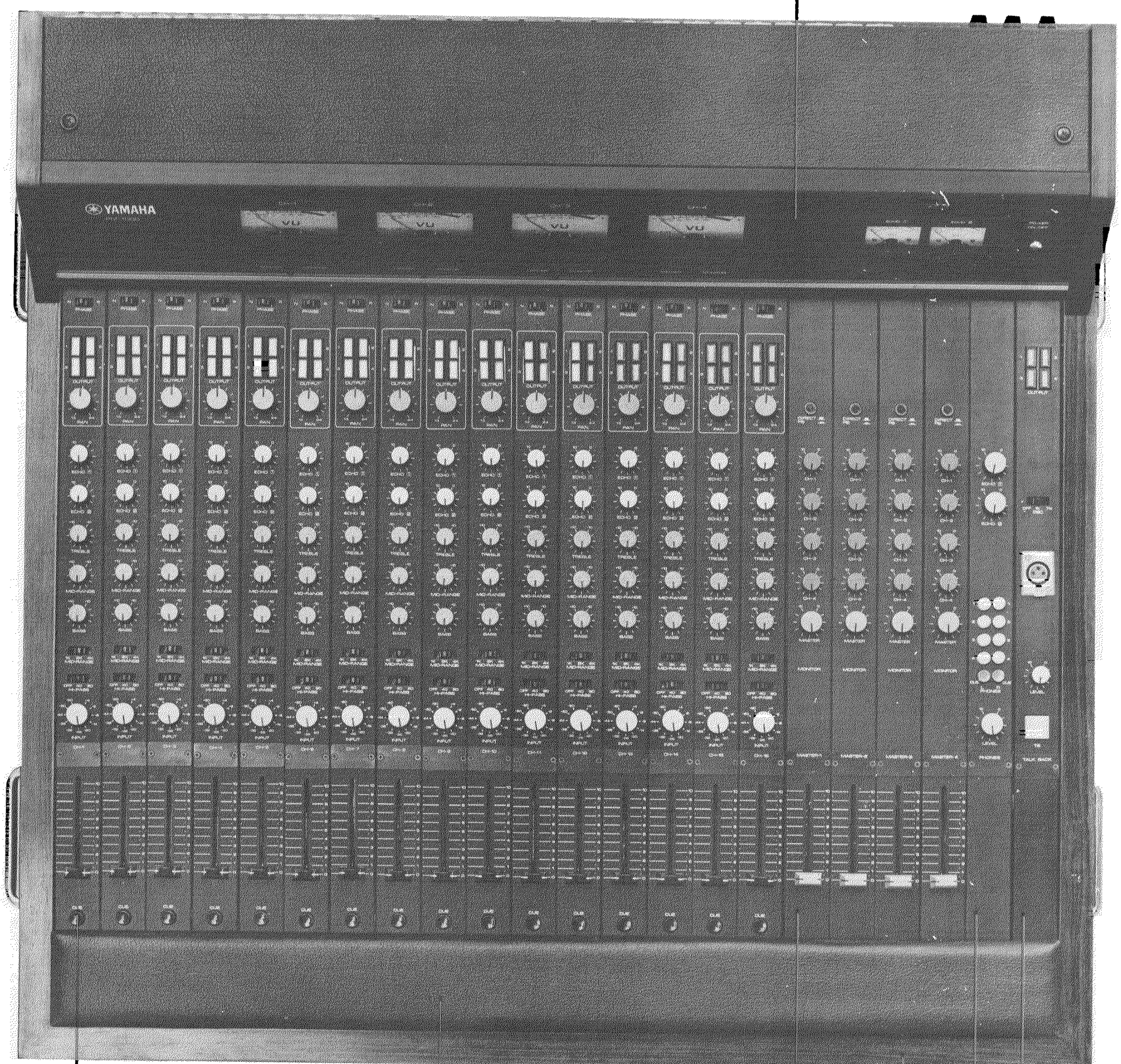
SECTION ONE 1

IDENTIFICATION OF PARTS AND CONTROLS

MAIN COMPONENTS & SUB ASSEMBLIES*

Meter Panel



Input (x 16)

Armrest

Headphone Jacks

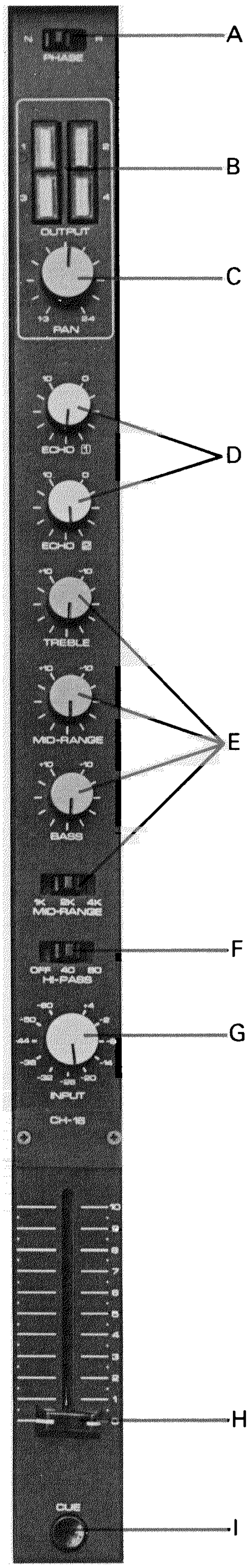
Master and Monitor Module (x 4)

Headphone and Echo Module

Talkback and Oscillator Module

*Many times there is more than one label appropriate to a console control or function. Depending on the specific application or the industry (recording or sound reinforcement), the nomenclature will vary. To avoid confusion in this manual, wherever possible, we will try to refer to the nomenclature actually printed on the console. Where function may be unclear, an alternate term may be printed in parentheses. Some examples of this alternate terminology are: **Echo** (foldback or stage monitor), **Monitor** (speaker feed), **Line Output** (program output), and **Sub In** (auxiliary input).

ONE 2



THE INPUT MODULE

A. PHASE SWITCH Reverse the polarity of the audio signal entering the input module. This switch eliminates the need to rewire connectors for out-of-phase audio sources. Sliding the switch from **N** (normal) to **R** (reverse) interchanges the leads joining pins 2 and 3 of the XLR connector to the input transformer's primary winding.

NOTE: PM-1000 XLR connectors are wired according to DIN Standards; pin 2 high and pin 3 low. Refer to the Installation Section for details. *Normal* phase for this console means that a positive voltage applied to pin 2 at the input causes a positive voltage to appear at pin 2 of the XLR outputs.

B. OUTPUT ASSIGN (BUS ASSIGN) SWITCHES Apply audio from the input module to any combination of the four program mixing buses. Latching switches 1, 2, 3 and 4 either individually or in any combination assign post-equalizer and fader audio to correspondingly numbered buses. As described below, adjusting the **Pan** pot to either side of center alters the level applied to the four program mixing buses.

C. PAN POT adjusts the relative output level available to the four program mixing buses. Panning to the center position provides equal output at full post-fader level to all four **Output** assign switches. In other words, the program is *centered* in four buses. Panning to the left gradually removes audio from the feed to buses 2 and 4, maintaining full output to buses 1 and 3. Conversely, panning to the right gradually removes the output from buses 1 and 3, maintaining full output in buses 2 and 4. The *left* and *right* designations are arbitrary, based on the rotation of the pan pot; they refer to the use of the console's line outputs for driving stereo or 4-channel recorders and/or loudspeaker systems.

D. ECHO 1 AND ECHO 2 (ECHO MIX CONTROLS) Adjust the module's output to each of two auxiliary mixing buses. These controls apply pre-fader, post-equalizer audio to the correspondingly numbered echo mixing buses. The audio on these buses can be fed to external reverbs, echo devices or tape delay units. In addition, the echo outputs are ideally suited to driving performers' cue headphone systems or stage monitor (foldback) systems; these applications require external power amplifiers.

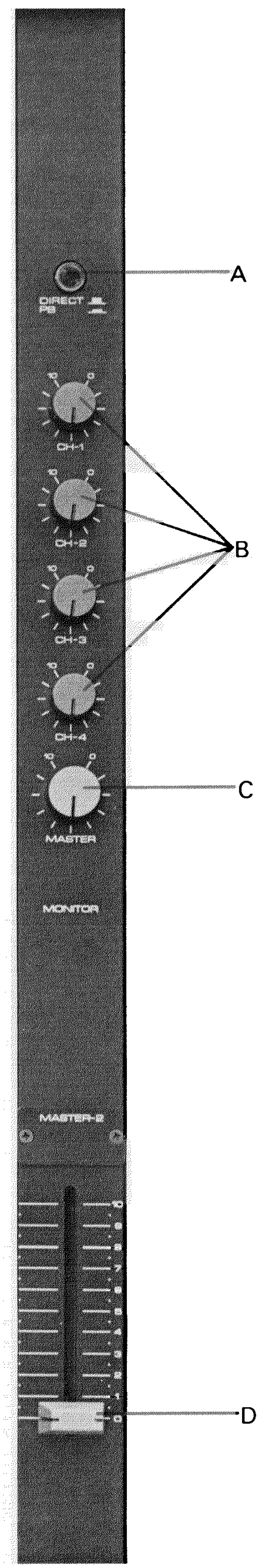
E. EQUALIZER Alters the frequency response of the input module in order to create a tremendous variety of tonal characteristics. The **Mid-Range** control acts on any of three *presence* frequencies (1kHz, 2kHz or 4kHz), as determined by the **Mid-Range** Select Switch. **Mid-Range** provides ±15dB of continuously variable peaking equalization. The **Bass** and **Treble** controls provide ±15dB of continuously variable shelving equalization at 100Hz and 10kHz respectively. Centering the three equalizer controls provides flat audio response by defeating all equalization.

F. HIGH PASS (LOW CUT) FILTER Switch-actuated 12dB/octave high pass filter. The filter follows the equalizer, affecting the output to the cue, echo and program mixing buses. The switch has three positions: **OFF** bypasses the filter entirely: **40** attenuates audio below 40Hz; **80** attenuates audio below 80Hz, the most pronounced filter effect.

G. INPUT LEVEL (INPUT SENSITIVITY) SWITCH A precision 4-stage switch that varies the preamplifier gain and/or attenuates the incoming signal. **Input** Level affects all outputs from the module. It provides optimum results with virtually any input, from -60dB to +4dB. When correctly adjusted, **Input** Level permits the input faders and echo mix controls to be used in their best range — with maximum headroom and minimum noise characteristics. The switch has 11 settings: -60, -50, -44, -38, -32, -26, -20, -14, -8, -2 and +4dB, each corresponding to a nominal input level (i.e., -60dB is not a 60dB pad, but is the most sensitive characteristic for nominal -60dB inputs).

H. CHANNEL FADER (INPUT FADER) A straightline control which provides continuously variable adjustment of the module's output to the program mixing buses, completely killing the signal at the bottom of its travel. The fader has no effect on the echo or cue outputs of the module.

I. CUE (PREVIEW/SOLO) BUTTON) Applies audio to an auxiliary cue mixing bus when the button is depressed. The cue bus is fed with pre-fader, post-equalizer audio. Since the cue feed is unaffected by the channel fader or the echo mix controls, the incoming signal may be previewed prior to assigning any audio to the program mixing or echo mixing buses. **Cue** is monitored via the headphone output.



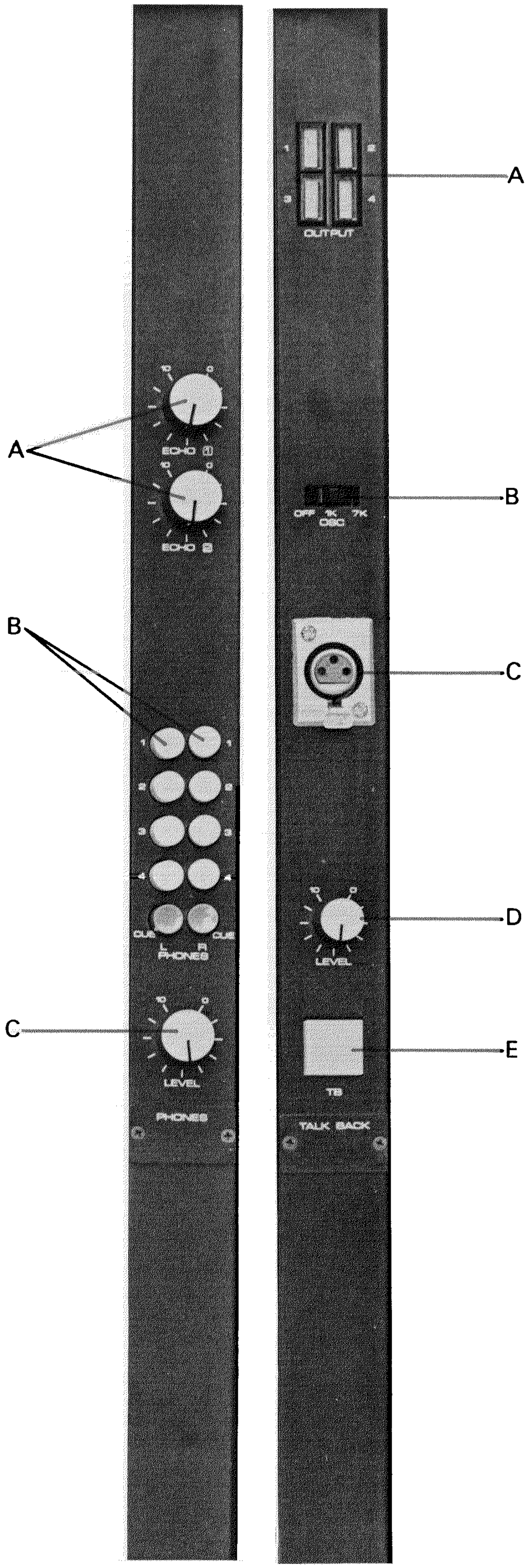
MASTER AND MONITOR MODULE

A. DIRECT/PLAYBACK (OUTPUT SOURCE SELECT) SWITCH Selects the audio source applied to the master module's input. **Direct** mode derives audio from the program mixing bus numbered to correspond to the module (or from any source plugged into the **Master In** jack). Playback **(PB)** mode derives audio from the correspondingly numbered **PB In** jack. In practice, the **Direct/ PB** switch selects between a live program and a recorded program to drive the console output.

B. MONITOR MIX (SPEAKER MIX) CONTROLS These rotary level controls, labeled **CH-1** through **CH-4**, derive audio from the line outputs of master modules 1-4. A monaural mixdown of the line outputs is obtained by adjusting the four monitor mix controls. This mono mix is brought to the **Monitor Out** jack, where it can drive a power amplifier for one or more loudspeakers. Since there are four master modules, each with an identical complement of four monitor mix controls, up to four distinct speaker mixes can be derived from the original four line outputs. This is known as a 4x4 matrix with level controls.

C. MONITOR MASTER (SPEAKER LEVEL) CONTROL Provides overall level control for the mono mix created with the monitor mix controls immediately above it. When the four monitor outputs are driving different loudspeakers, the **Monitor Masters** adjust the balance between speaker feeds. This 4x4 matrix arrangement offers far more flexibility for speaker feeding than would be available if the speakers were driven directly from the four line outputs **(Line A** or **Line B)**.

D. MASTER FADER (GROUP MASTER) Sets the overall signal level applied to the main program outputs of the console, **Line A** and **Line B**. (Line A & B are identical, redundant outputs, each consisting of four connectors that derive audio from the **Direct/ PB** switches.) The **Master** fader governs the correspondingly numbered line outputs, whether derived from the program mixing bus (in **Direct** mode) or the Playback input (in **PB** mode). Together, the four faders are useful for balancing the levels of different groups of inputs that have been assigned to the four mixing buses; in this capacity, the faders are Group Master controls.



Headphone &
Echo Module

Talkback &
Oscillator Module

HEADPHONE AND ECHO MODULE

A. ECHO SEND (FOLDBACK) MASTER CONTROLS Provide overall level control for echo mixing buses 1 and 2. The audio mixes established with the echo mix controls in each input module pass through the **Echo** Send Masters, and drive the **Echo Out** jacks. When the echo outputs are connected to foldback (stage monitor) or performers' headphone systems, these master controls determine the monitor volume. When the echo outputs are connected to an echo or reverb device, the Master controls determine the level of the delay effect.

B. HEADPHONE MIX PUSHBUTTONS This portion of the module consists of two rows of pushbuttons, each row capable of selecting any combination of five audio sources. The left and right rows feed audio to the left and right sides of the headphone output. The five available sources are: program mix bus 1 through program mix bus 4, and the cue bus. When more than one pushbutton per row is latched, the selected audio sources are blended in equal proportions.

C. HEADPHONE LEVEL CONTROL Simultaneously adjusts the overall volume of the left and right headphone feeds. This stereo output is available at two stereo phone jacks. The jacks are wired in parallel so as to drive one or two pairs of stereo headphones.

TALKBACK AND OSCILLATOR MODULE

A. OUTPUT ASSIGN SWITCHES When latched, switches 1 through 4 apply the module's output to program mixing buses 1 through 4. (16-channel consoles manufactured after February, 1976 (*serial no. 1542 & up*), and all 24 and 32- channel consoles have two additional buttons. These assign the module output to Echo Mix buses 1 and/or 2.) Depending on the status of the talkback button, the output will be either a vocal signal from the talkback mic input, or a test tone from the built-in oscillator.

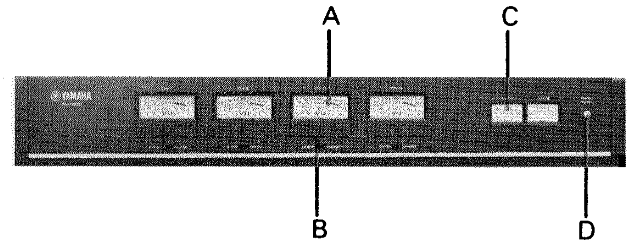
B. OSCILLATOR FUNCTION SWITCH Turns the oscillator **OFF**, or sets it for constant sine wave generation at **1,000 Hz** or at **7,000 Hz.**

C. TALKBACK MICROPHONE CONNECTOR Accepts any low impedance microphone for use with the talkback circuitry.

D. TALKBACK LEVEL CONTROL Adjusts the talkback microphone preamplifier gain. This control affects only the talkback level, not the oscillator.

E. TALKBACK BUTTON Pressing this button activates the talkback mic. If the oscillator is already switched on, pressing **Talkback** interrupts the oscillator and substitutes audio from the talkback preamplifier.

METER PANEL †



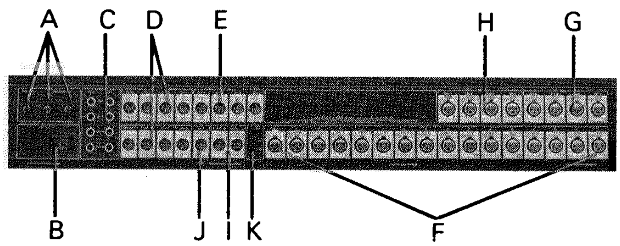
A. MAIN (PROGRAM OUTPUT) VU METER Four illuminated meters that provide a visual indication of the average audio output level from Master and Monitor modules 1 through 4. Depending on the setting of the function switch, each meter displays either the **Master** (Line Out) or **Monitor** Out level. A zero VU indication is equivalent to +4dBm output into a 600-ohm termination.

B. METER FUNCTION SWITCH Two-position slide switch allows use of the associated VU meter for monitoring either of two console outputs, **Master** (Line) or **Monitor**. The Master indication is derived from the **Line Out A** jack, and the Monitor reading is derived from the **Monitor Out** jack that corresponds in number to the VU meter.

C. ECHO OUTPUT VU METER Two illuminated VU meters provide a visual indication of the signal level at the Echo 1 and Echo 2 outputs. When driving monitor (foldback) systems from the echo outputs, these meters indicate the stage monitor level. A zero VU indication is equivalent to a +4dBm output into a 600-ohm termination.

D. MAIN POWER (AC ON/OFF) SWITCH Pushbutton alternately switches the AC power on and off. No audio passes through the console when power is off.

REAR PANEL †



A. POWER FUSES Protect the primary (line) and secondary (low voltage) portions of the console's power supply.

B. POWER CORD CONNECTOR Accepts the AC power cord provided with the console. PM-1000 consoles delivered in the U.S.A. and in Canada are equipped with grounded (3-wire) AC cords for connection to 50 or 60Hz, 110-120Vrms AC power mains. The console draws about 45 Watts, or 0.5 Amps (90 Watts for 24/32 consoles).

C. MASTER OUT & MASTER IN Provide a patch point between the program input to the Master and Monitor module and the input to the **Direct/PB** switch. The jacks are wired to accept standard phone plugs. Their nominal operating level is -20dB (for high impedance circuits).

D. LINE-A-OUT AND LINE-B-OUT JACKS (PROGRAM MIX OUT) These are redundant outputs. **Line A** and **B** each consist of four jacks, and the signal is derived from the Master Faders. The nominal output is +4dBm (for 600-ohm termination), and the connectors accept XLR type 3-pin female plugs.*

E. MONITOR OUT (SPEAKER FEED) JACKS Carry the console output from the Monitor Mix (Speaker Matrix) section. Each **Mon Out** jack, 1 through 4, carries audio from the correspondingly numbered **Monitor Master** control. The outputs are nominal +4dBm (for 600-ohm termination), and the connectors accept XLR type 3-pin female plugs.*

F. CHANNEL INPUT JACKS Numbered 1 through 16, the **Input** jacks accept audio from any low impedance, balanced or floating audio source from -60dB to +4dB nominal level. The inputs are balanced and transformer-isolated, but auxiliary matching transformers are recommended for unbalanced, high-impedance inputs. The connectors accept XLR type 3-pin male plugs.

G. SUBMIXER INPUT JACKS Numbered 1 through 4, these jacks accept audio from a low impedance, low level source, such as the Yamaha PM-400B mixer. The **Sub In** jacks are designed for nominal -20dB levels, and accept XLR type 3-pin male plugs.* Audio from **Sub In** is applied to the program bus ahead of the Master Faders, thus the Master Faders control the Sub In level.

H. PLAYBACK INPUT JACKS Numbered 1 through 4, the **PB In** jacks accept audio from a low impedance, low level tape machine output. This audio is applied to the PB side of the **Direct/PB** switch in the correspondingly numbered Master and Monitor module. The **PB In** jacks are designed for -20dB levels and accept XLR type 3-pin male plugs.*

I. ECHO OUTPUT (FOLDBACK OUT) JACKS Carry the console output from Echo Send 1 & 2 Masters. The **Echo Out** jacks are transformer-isolated, nominal +4dBm (for 600-ohm termination), and they accept XLR type 3-pin female plugs.* As discussed previously, these jacks may be used to drive foldback (stage monitor) amplifiers or performers' headphone amplifiers.

J. TALKBACK OUTPUT JACK Carries the output of the Talkback & Oscillator module, even if the module's bus assign switches are not latched. The **TB Out** jack can be connected to a power amplifier, directly to headphones, or to any point where the oscillator or talkback signal is needed. For calibrated operation with optimum fidelity, **TB Out** delivers +4dBm into a 600-ohm termination. The connector accepts an XLR type 3-pin female plug.*

K. PHANTOM POWER SWITCH Connects the 48Vdc phantom power supply to the center tap of all input transformer primary windings. 24/32 consoles have switches for each input. Recommended procedure is to turn off the phantom power whenever condenser microphones are not used; however, the "phantom" switch may be left on without damaging most standard microphones, line inputs, or the phantom supply.

†*See Section 8 for 24/32 Consoles.*

**Refer to the Installation Section of this Manual for information regarding the polarity of all XLR connectors.*

SECTION TWO

SPECIFICATIONS

GENERAL SPECIFICATIONS *See Section 8 for those specifications which differ on 24 and 32 channel consoles.*

Frequency Response	+0, -4dB, 20Hz - 20kHz; ±0.5dB, 50Hz - 15kHz.	
Total Harmonic Distortion	Less than 0.25% @ +10dB, 20Hz - 20kHz; Less than 0.5% @ +20dB, 70Hz - 15kHz.	
Hum and Noise* **(20Hz - 20kHz)**	-124dBm Equivalent Input Noise (E.I.N.).	
	-69dB (73dB S/N) Line Out A & B: Master Fader at nominal level and all Input Faders down.	
	-60dB (64dB S/N) Line Out A & B: Master Fader and one Input Fader at nominal level.	
	-63dB (67dB S/N) Echo Out: Master Send at nominal level and all Echo Mix Controls down.	
	-54dB (58dB S/N) Echo Out: Master Send and one Echo Mix Control at nominal level.	
Maximum Voltage Gain (Input Selectors at -60dB, where applicable)	PROGRAM —	74± 2dB from Channel In to Line Out A & B. 48± 2dB from Channel In to Master Out.
	MONITOR —	74± 2dB from Channel In to Monitor Out.
	ECHO —	74± 2dB from Channel In to Echo Out.
	SUB IN —	30± 2dB from Sub In to Line Out A & B.
	PB IN —	30± 2dB from PB In to Line Out A & B.
	MASTER —	32± 2dB from Master In to Line Out A & B.
Equalization	BASS —	± 15dB @ 100Hz, shelving.
	MID-RANGE —	± 15dB @ 1kHz, 2kHz or 4kHz; peaking.
	TREBLE —	± 15dB @ 10kHz, shelving.
High Pass Filter	12dB per octave roll-off below 40Hz or 80Hz.	
Oscillator	1kHz or 7kHz sine wave, +4dBm @ < 1.0% THD.	
Talkback	Microphone input jack, preamp, level control, and push-to-talk switch; to pgm. buses and/or direct out.	
Inputs to Console	16 x Channel Inputs (microphone and line sources). 4 x Sub In (Submixer input). 4 x Master In (Hi-Fi auxiliary program input). 4 x PB In (Playback input). 1 x Talkback Mic In.	
Mixing Buses	4 x Main Program (Line Out). 4 x Monitor (Speaker feed). 2 x Echo (Foldback/stage monitor). 1 x Cue (Preview).	
Console Outputs	8 x Line (4 Line A, 4 Line B). 4 x Monitor (Speaker feed). 4 x Master (Hi-Fi auxiliary program output). 2 x Echo (Foldback/stage monitor). 1 x Talkback (Talkback mic or oscillator out). 2 x Stereo Headphone (Console operator's monitor).	
Crosstalk	-60dB at 1,000Hz, adjacent inputs, -50dB at 1,000Hz, input to output.	
VU Meters (0 VU=+4dBm)	4 x large, illuminated meters; switchable for Master (Line Out) or Monitor (Monitor Out).	
	2 x small, illuminated meters; Echo (Foldback) Out.	
Phantom Power	48Vdc applied to balanced channel input transformers for powering condenser microphones. May be turned On or Off with rear-panel switch.	
Power Supply	Self-contained module inside console, fused and fully regulated. Requires 110-120 VAC, 50-60Hz, 45 Watts. May be modified for 220-240V AC operation.	
Finish	Black anodized aluminum panels, padded armrest, rosewood veneer cabinet.	
Dimensions	34¼" (87.2cm) wide X 34½" (87.5cm) deep X 11" (27.7cm) high.	
Weight	110 pounds (50kg).	
Accessories	Integral carrying handles and removable leatherette cover are included with console.	
Warranty	One year, parts and labor.	

**Measured with 6dB/octave filter @ 12.47kHz: equivalent to a 20kHz filter with infinite dB/octave attenuation.*

DEFINITION OF TERMS: UNBALANCED, BALANCED & FLOATING

Unbalanced, balanced and floating circuits may all be transformer-isolated. The distinction between them lies in the way the circuits are referenced to ground (audio common). A FLOATING circuit has no ground reference, as illustrated by the primary of the PM-1000's Sub In, PB In, and Talkback Mic input transformers and by the secondary windings of the Line Out, Monitor Out, Echo Out and Talkback Out transformers.

The PM-1000's input channel transformer primaries are BALANCED by virtue of their center taps. A balanced circuit requires either a center tapped transformer, or resistors from each side of the transformer to ground; either condition places both sides of the transformer at an equal difference from ground potential. In other words, the transformer is *balanced* with respect ground.

Any circuit which causes one side of an input or output to be grounded is considered an *unbalanced* circuit, unbalanced in that both sides of the circuit are at different potentials with respect to ground. Most non-transformer connections, including the PM-1000's **Master In** and **Master Out** jacks, are unbalanced (there are exceptions, but they are rare).

Typical unbalanced audio sources are: direct feeds from electric instruments, from power amplifiers, and from high fidelity tape machines. In order to obtain best results when using these devices with the PM-1000's channel inputs, a matching transformer should be inserted at the remote device. The transformer prevents the console's input transformer from becoming unbalanced, which may induce noise and crosstalk. Also, the matching transformer provides the best impedance match for high impedance sources.

Typical floating sources (which maintain a balanced condition in the PM-1000 channel inputs) are: microphones, transformer-isolated submixers, and virtually any transformer-isolated output from auxiliary professional equipment.

DEFINITION OF TERMS: dB, dBV, dBm and dB SPL

The term dB, which means decibel (1/10th of a Bel) expresses a ratio. More precisely, dB is 10 times the logarithm of a power ratio and 20 times the logarithm of a voltage or sound pressure ratio.

dBV expresses a voltage ratio. It is not directly related to current or circuit impedance. The 0 dBV reference is usually 1V rms.

dBm expresses a power ratio. It is related to the voltage or current across a low impedance. The 0 dBm reference is 0.775Vrms in a 600-ohm circuit, which is equal to 1 milliwatt at 600 ohms.

dB SPL expresses an acoustic pressure (not power) ratio. The 0 dB SPL reference is approximately the threshold of human hearing at 1kHz, which is equal to 0.0002 dynes/cm^2.

dB expresses the difference between two levels (power, voltage etc.) and is a relative term. The difference between +10dBm and +4dBm is 6dB. The difference between -20dBV and -10dBV is 10dB.

dBV and dBm are not numerically equal when dealing with 600-ohm circuits, although they are close (0dBV is 2.2dBm at 600 ohms). As the impedance is changed to other than 600-ohms (given a constant voltage), the value of dBV remains constant while the value of dBm changes. Consider a +4dBm output terminated by 600 ohms; the voltage level is +1.8dBV. This circuit has a voltage drop of 1.23V rms, and a power dissipation of 2.5 milliwatts. If the voltage now remains constant, but the termination is changed to 1200-ohms, the power dissipation drops to 1.23mW, +1dBm.

Since the power dissipation in high impedance circuitry is negligible, "dBV" is sometimes used to express signal levels in high impedance lines. "dBm" is commonly used to express signal (power) levels in low impedance lines, roughly between 4 and 1200-ohms. To avoid confusion, we use the term "dB" to represent a specific voltage, whether it is applied to a low or a high impedance. 0dB is referenced to 0.775V, and 0dBm is referenced to 1mW (0.775V driving a 600-ohm termination). For example, when 12.3V is fed to a high impedance, the level is designated "+24dB." When +24 dB (12.3 volts) drives a 600-ohm termination, the level is designated "+24dBm."

An increase of 3dB is equivalent to 2 X the power.
An increase of 10dB is equivalent to 10 X the power.
A decrease of 3dB is equivalent to 1/2 the power.
A decrease of 10dB is equivalent to 1/10 the power.
An increase of 6dB is equivalent to 2 X the voltage or SPL.
An increase of 20dB is equivalent to 10 X the voltage or SPL.
A decrease of 6dB is equivalent to 1/2 the voltage or SPL.
A decrease of 20dB is equivalent to 1/10 the voltage or SPL.

The open circuit voltage output of a low-Z microphone (in dBV), given a 94dB SPL sound field, is approximately equal to the microphone's rated EIA sensitivity. If the sound level at the mic is higher than 94dB SPL, add the difference to the EIA rating, and if the level is below 94dB SPL, subtract the difference from the EIA rating. Thus, a mic rated at –50dB EIA sensitivity, placed in a 104dB SPL environment, will yield about –40dBV (-37.8dB, or 10mV) output.

INPUT & OUTPUT CHARACTERISTICS

INPUT SPECIFICATIONS

Connection	IMPEDANCE Actual	IMPEDANCE Nominal Source	SENSITIVITY* (At Max. Gain)	INPUT LEVEL Nominal*	INPUT LEVEL Max. before Clip.	Connector In Console
Inputs (1-16)	1700 Ω, balanced	150 Ω Mics & 600 Ω Lines	0.25mV (-70dB)	0.8mV (-60dB)	12.3V (+24dB)	XLR-3-31
PB & Sub In (1-4)	2300 Ω, floating	600 Ω Lines & Instruments	40mV (-26dB)	80mV (-20dB)	3.2V (+12dB)	XLR-3-31
Master In (1-4)	9000 Ω, unbal.	5k Ω Lines & Instruments	31mV (-28dB)	62mV (-22dB)	12.3V (+24dB)	Phone Jack
Talkback Mic	2300 Ω, floating	150 Ω Mics	8mV (-40dB)	2.5mV (-50dB)	1.23V (+4dB)	XLR-3-31

OUTPUT SPECIFICATIONS

Connection	IMPEDANCE Actual	IMPEDANCE Nominal Load	POWER OUTPUT LEVEL Nominal	POWER OUTPUT LEVEL Max. before Clipping	Connector In Console
Line A (1-4), Line B (1-4), Monitor (1-4), Echo 1 & 2, Talkback	100 Ω, floating	600 Ω	1.23V (+4dBm)	10.8V (+22½dBm)	XLR-3-32
Master Out (1-4)	200 Ω, unbal.	5,000 Ω	62mV (-22dB)	2.1V (+9dB)	Phone Jack
Headphones	3.2 Ω, unbal.	8 Ω or greater	80mV (-20dB)	1.23V (+4dB)	Stereo Phone Jack (x 2)

* This is the level required to produce an output of +4dBm (1.23V).

GENERAL APPLICATIONS

THE ROLE OF AN AUDIO CONTROL CONSOLE

The mixing console is the heart of a sound system. The console receives all microphone and recorded sources, amplifies the sound, and processes it. Auxiliary devices (compressors, limiters, reverbs, tape machines and amplifiers) are connected to the sound system via the console. Within the console, some sources are combined, and others are assigned to more than one output. Level balancing, panning, filtering and equalization are all integral functions of the console. Ultimately the console feeds tape machines, and/or it drives power amplifiers that in turn drive loudspeakers. Often there are more than one set of loudspeakers, each with varying program feeds.

The PM-1000 is designed to satisfy a wide spectrum of audio control requirements. It has ample flexibility to serve as a recording studio console, and it will deliver excellent results in the largest of sound reinforcement applications. Ideal for reinforcement in large clubs, theatres, auditoriums, concert halls and amphitheatres, the PM-1000 is unsurpassed for its versatility, portability and economy.

PLACEMENT OF EQUIPMENT

The object of a sound system is to deliver an amplified version of some original audio source to the audience. Due to the significant and varied effects of room acoustics, the sound in different environments can never be identical. However, a skilled operator using sophisticated equipment can critically evaluate the sound quality and modify the audio to obtain a reasonably good approximation of the "ideal" sound.

The console operator must have continuous, accurate information. He must know what the audience is hearing. Because performances tend to be dynamic, changing constantly, the most effective means to guide the operator is to place the console in the midst of the audience. A preferred location is centered left-to-right, and slightly more than half way to the rear of the audience. In this position, the direct sound from the stage and the coloration from room acoustics jointly affect decisions at the console.

For convenience, tape machines and auxiliary signal processing gear are usually placed immediately adjacent to the console. The only electronics that are remotely located are the microphones and the power amplifiers which drive the loudspeakers.*

**Power amplifiers should be mounted as close as is practical to the loudspeaker(s) they drive. By keeping the amplifier-to-loudspeaker distance as short as possible, and by using a sufficiently large wire gauge, large power losses due to dissipation in the transmission cable are avoided. The practice of locating power amplifiers remotely from the console is suitable for stage monitor (foldback), main reinforcement and intercom loudspeakers, as well as for auxiliary headphone drive amps.*

If the console cannot be placed in the middle of the audience, an alternate position is at the rear of the listening area. In this location, the console operator needs an assistant who will stay in the middle of the audience. From there, the assistant can keep the console operator abreast of the sound quality, and can suggest necessary adjustments.

In recording studio applications, the PM-1000 should be placed in a control booth that is acoustically isolated from the studio. However, visual contact, via sound-trapped windows or closed circuit TV, should be maintained. Auxiliary equipment should be placed near the console, in the control booth. Due to the relatively short distance between the control room and the studio, all power amplifiers can be located in the control booth or in an adjacent room. Echo chambers and echo devices should be isolated from loud monitoring environments to reduce the likelihood of acoustic feedback or leakage. (Consult Section Four of this manual for some illustrated examples of typical console set-ups.)

THE DISTINCTION BETWEEN PROFESSIONAL AND HI-FI TYPE EQUIPMENT

In virtually any application, a variety of auxiliary equipment will be connected to the PM-1000. This equipment includes: tape machines, compressors, graphic equalizers, echo devices, reverb units, phasers, and just about any audio electronics imaginable. Regardless of the function of auxiliary equipment, it will undoubtedly fall into one of two general categories, professional type or hi-fi type (semi-pro).

The distinction between professional and hi-fi equipment is very important because it affects the way this equipment will be used with the PM-1000. Brand name, size, panel colors, durability, and subtleties in function *are not the significant differences.* What matters is that professional equipment and hi-fi equipment usually operate at different input and output levels, and require different source and load impedances to function correctly. The PM-1000 is designed to function at its peak with professional equipment, but it will yield excellent results with hi-fi type equipment so long as certain precautions are observed. These precautions are outlined in the Installation section of the manual, but the following paragraphs are intended to explain how the specific requirements of professional and hi-fi equipment differ.

IMPEDANCE

Professional equipment is generally designed to be driven from a 600-ohm source, and its output will drive 600-ohm or higher impedance loads. Often, professional input and output circuits are transformer isolated (balanced or floating); such circuits utilize dual-conductor shielded cables, with 3-pin XL type audio connectors (or Tip/Ring/Sleeve type).

Hi-fi equipment is designed to be driven from a 5,000-ohm or lower impedance source, and its output will drive 10,000-ohm or higher impedance loads. Hi-fi input and output circuits are usually unbalanced; such circuits utilize single-conductor shielded cables with 2-conductor audio connectors (standard phone type or RCA pin type).

The nature of balanced, floating and unbalanced circuitry is detailed in the Specifications section of this manual. For the purpose of this discussion, the most significant point is that an unbalanced circuit is somewhat more susceptible to hum and noise, especially if there is any irregularity in the grounding system. It should be noted that whether a circuit is unbalanced, balanced or floating has no direct correlation with its impedance. Low impedance and high impedance are relative terms. A-150 ohm microphone is considered low impedance, whereas a 2,000-ohm mic is considered high impedance. A 600-ohm line is considered low impedance, whereas 5,000-ohm, 50,000-ohm, or 100,000-ohm lines are all considered high impedance.

For audio transmission over long distances, low impedance lines provide the best results. High impedance lines attenuate the higher frequencies, however in short cable runs, under 10 feet, there is virtually no difference between the results obtained with low and high impedance lines. Also, provided the grounding is correct, there is little difference between unbalanced and balanced or floating circuitry. In any case, the specific requirements of the PM-1000's input and output circuits should be considered in setting up the system.

OPERATING LEVELS

Professional line level is nominally between 0dBm and +4dBm; that is, the average program level is approximately .775Vrms to 1.23Vrms terminated by a 600-ohm line. The peak level may extend up to about +24dBm or higher (12.3Vrms). The high level input (not mic input) of professional audio equipment is designed to accept levels in this power range without overdrive (clipping distortion); most professional equipment can be driven to full output by nominal +4dBm input levels.

Hi-fi type equipment operates at considerably lower levels than professional equipment, approximately -16dB nominal. This is equal to about 120mV across a 10,000-ohm or greater (high impedance) line. Peak program levels may reach or slightly exceed +4dB, 1.23V — *across a high impedance line.* Thus, hi-fi equipment is usually not capable of driving professional equipment to its full output, or at least not before the hi-fi output reaches a high level of distortion. Moreover, when the output of hi-fi equipment, which is meant to be terminated in a high impedance, is connected directly to professional equipment, the low impedance acts as a partial short-circuit. This serves to overload the hi-fi output.

The PM-1000's channel inputs are correctly matched to professional equipment; they will not overload low impedance microphones and lines. But the same inputs can overload some hi-fi equipment. To prevent possible overload, and to obtain the best frequency response with hi-fi equipment, a matching transformer may be installed. This converts a high impedance to a low impedance. Approximately 10dB to 20dB of voltage level is lost in the transformer, but this level can be recovered with the console's **Input** Level switches. Because the signal is being attenuated and then amplified again, some increase in the signal-to-noise ratio will occur, but it is usually not objectionable.

At the other end of the system, the PM-1000's outputs are best suited to driving professional equipment (with the exception of **Master Out**, as explained on pg. 3 of this section). When driving hi-fi equipment, care must be taken to avoid overdrive of the hi-fi input. An excellent method to avoid overdrive is to connect a 20dB or 24dB T-pad across the hi-fi input. The pad attenuates the console's nominal +4dBm output to about -16dB, a good match for average hi-fi input sensitivity. This allows both the console's faders and the auxiliary equipments' controls to be operated in the range that yields maximum headroom, lowest signal-to-noise characteristics, and a wide range of physical control. (Pad construction is illustrated in the Installation secton).

CHANNEL INPUTS

The PM-1000's channel inputs are transformer-isolated. A very broad range of level control, by means of attenuation and gain trim, is provided by the **Input** Level switches. Virtually any microphone or line source can be accommodated. Only very high level inputs, such as the speaker output from power amplifiers or electric instrument amplifiers, will require external attenuation pads.

The inputs are designed for low impedance sources, typically 150-ohm to 250-ohm microphones and 600-ohm lines. Low impedance cables yield superior transmission over long distances; many such cables can be run 2,000 feet or more without severe high frequency loss and without hum pick-up, although microphone manufacturer's specifications should be consulted.

When a high impedance microphone (or auxiliary electronic device) is used, a high-to-low impedance matching transformer should be installed at the remote (mic) end of the cable. Condenser microphones that contain integral preamplifiers can be phantom powered from the console's 48V supply. This power is transmitted through standard 2-conductor shielded cables, along with audio signal; the circuitry is designed so that non-condenser type microphones should not be damaged.

SUBMIXER INPUTS

Sub In provides a convenient means to expand the input capability of the PM-1000 by allowing another console or mixer to be "chained" to the PM-1000. When more input channels are required, an auxiliary mixer (or several for that matter) can accept the extra inputs, pre-mix them, and feed them to the console's mixing buses through the **Sub In** jacks. **Sub In** may also be used to accept feeds from remote studios or other remote sources, as well as from tape recorders and echo/reverb devices.

Since the four **Sub In** jacks apply audio directly to the program mixing buses, the level and impedance of the incoming audio must be correct. **Sub In** has a nominal input sensitivity of -20dB, so professional equipment with nominal +4dBm output levels will have to be padded 20 or 24dB. Some mixers, such as the Yamaha PM-400B, have a -20dB (medium level) output in addition to a +4dBm output, and can therefore be connected directly to **Sub In** without padding.

Many hi-fi type mixers, as well as hi-fi tape machines, can be connected directly to the **Sub In** jacks, despite the apparent impedance and level mismatch. The explanation for this capability lies in the fact that most hi-fi equipment, while rated at 10K-ohms output impedance, has an actual output source impedance of about 2K-ohms. **Sub In,** while rated at a nominal input impedance of 600-ohms, has an actual termination of about 2.5K-ohms. Thus, the hi-fi equipment will be loaded by **Sub In,** not severely, but enough to drop the level by about 6dB. Considering that the average hi-fi nominal output level is -16dB, a drop of 6dB brings the input to -22dB...very close to the nominal -20dB sensitivity of **Sub In.**

For that hi-fi type equipment which will not function satisfactorily when connected directly to **Sub In**, an external line amplifier must be installed. Matching transformers will not cure the problem because, to match a high impedance source to a low impedance input, something like 10 to 14dB of voltage loss occurs, and this would bring the hi-fi output level well below the nominal sensitivity of **Sub In.**

PLAYBACK INPUTS

The four Playback Inputs (**PB** In) are designed to accommodate the play output of a 4-channel professional tape machine. Alternately, two stereo machines, two monaural and one stereo machine, or any combination of sources that add up to no more than four channels may be connected to **PB** In. Due to the same factors explained under "Submixer Inputs", hi-fi tape machines cannot be connected directly to the **PB** In jacks; external line amplifiers to match impedance and level are required. **PB** In is available in each of the Master/Monitor modules by latching the **Direct/PB** switch.

Tape playback may be used to feed recorded programs through the console outputs. In sound reinforcement applications, this feature is handy for filling intermissions and breaks in the show. For studio applications, **PB** In allows the operator to listen to a recording that has just been made without having to alter the input channels' controls.

PB In provides no equalization, filtering, or mixing capability, so it is not suitable for playback of unmixed recordings (except for reference). Raw, unmixed tapes are better accommodated by the standard input channels. As explained under "Submixer Inputs", the -20dBm sensitivity of this input requires a 20dB or 24dB T-pad (or H-pad) to be connected between the professional machine's output and the **PB** In jacks. Locating the pad at the console permits the higher signal level to flow through the length of the transmission cable, reducing the likelihood of hum and noise pick-up.

TALKBACK INPUT

The talkback jack on the front panel of the console is provided for connection of a low impedance microphone. This mic, usually mounted on a flexible gooseneck, (available from your Yamaha dealer) is used by the console operator to identify tape recordings, make announcements, or for intercom with performers, lighting personnel, or other remote areas. When the console is not located in the middle of the audience, the talkback input can be fitted with a **Y**-adapter. It will then accept two mics, one for the console operator and one for his on-floor assistant. Both people can wear headphones connected to the console's talkback output via another **Y**-adapter. In this way, both can talk and listen when the talkback button is pressed.

The talkback input is preamplified in the Talkback/Oscillator module, which also has a level control and a push-to-talk switch. The talkback preamp output can be pushbutton-assigned to any combination of the four program mixing buses, and it is simultaneously available as a nominal +4dBm direct console output. The output will drive 600-ohm or higher impedance headphones, or a power amplifier. 8 ohm headphones will work, but with some loss of fidelity. For best results with 8-ohm headphones, and for additional listening volume, an external headphone amplifier should be used.

MASTER IN & MASTER OUT

The **Master In** and **Master Out** jacks are unbalanced, high impedance circuits designed for -20dB nominal levels. As such, they are ideal for most hi-fi type equipment. For example, **Master In** and **Master Out** facilitate the use of graphic equalizers and compressor/limiters. **Master Out** also may be used to make tape recordings, and/or to drive tape delay effects.

When no plugs are connected to the Master jacks, the program mix flows between the jacks via an internal jumper, then through the **Direct** side of the **Direct/Pb** switch, and on through the **Master Fader** to the console's **Line** and **Monitor** outputs. Inserting a plug in a **Master In** jack interrupts the internal jumper. Thus, when an equalizer or compressor/limiter is fed by **Master Out,** processess the program mix, and returns the processed audio to **Master In,** all audio reaching the **Master Fader** flows through the remote device.

Master Out is wired in a way that allows it to feed a remote device without interrupting the flow of the program mix to the console output; thus, a tape recorder (such as the TEAC 3340 or Pioneer RT-1040) can be fed from **Master Out.** Since the recorder feed comes before the **Master Faders,** program fades will not affect the level applied to the tape. The signal flow of the program to the **Master Faders** is not affected because, unlike **Master In,** inserting a plug in **Master Out** does not break the internal jumper between the jacks. If the recording is done in a studio situation, as opposed to sound reinforcement, then the play output of the recorder can be connected to **Master In.** While recording, leave the Source/Tape switch on the recorder in Source position; this allows the program mix to be monitored in real time via the console's **Line** or **Monitor** outputs. When the recording is completed, simply switch the Source/Tape switch to Tape mode; the tape playback can then be heard without any adjustment of console controls.

To obtain a tape delay effect, feed a 3-head tape recorder's input from **Master Out,** and return the Play output (the Off Tape output) to the PM-1000 channel input(s); then the delayed audio can be equalized with the channel equalizer and mixed back into the program using the **Channel Fader** and the **Output Assign** switches. Using **Master Out,** rather than **Echo Out** to drive the tape delay has twin advantages:

(1) attenuation is not needed at the tape machine input because **Master Out** is at the proper nominal level (-20dB, not +4dB), and (2) the Echo Mix buses remain available for foldback purposes.

TIME DELAY EQUIPMENT

Reverb, echo and delay devices fall under the general category of time delay units. They are most often used to add ambience or echo effects during the mixdown of tape recordings.

Natural reverberation (the sound of a large room) consists of rapid, multiple sound images. The images begin within a few milliseconds of the original sound, and they occur at closer intervals until the reverb decays completely. The overall decay time may range from 1/2-second to about 2-1/2 seconds, although higher frequencies fall off first. Artificial reverb devices are constructed with springs that have electro-mechanical transducers at either end.

Echo is a term often used to describe reverb. However, an echo differs in several respects from a reverb effect. Echo consists of one or more distinct, delayed sound images. These occur at finite intervals, beginning anywhere from approximately 40 milliseconds to 1/2 second or more, after the initial sound. Repeated echoes are obtained by feeding some proportion of the delayed output back to the echo input (regeneration). So-called "live echo" is obtained from an echo chamber, a room that contains a loudspeaker and a microphone. Artifical echo is usually obtained in either of two ways, with a tape recorder or a digital delay unit. In a tape recorder, the delay results from the time it takes the tape to travel from the record head to the playback head. In a digital delay unit, the audio is converted to a computer-like digital code, stored in shift registers, and then reconverted to audio. Other methods of obtaining time delay are available, including the use of a length of tubing with a microphone in one end and a speaker in the other end.

Professional reverb or echo units should be connected to the console's **Echo Out** (the Line & Monitor outputs are not usually suitable). **Master Out** will not drive most professional units, so it should be avoided. The return from the reverb or echo device can be brought to one or more input channels, or the **Sub In** jacks; this allows it to be blended with the dry, or "straight" sound and brought to the Line and Monitor outputs. (Refer to "Submixer Inputs" on page three-2.)

Master Out and **Master In** jacks are unsuitable for use with most echo and reverb units, even if the levels and impedances are compatible. This is because, to achieve a "proper" reverb or echo effect, the delayed audio must be blended with dry or "straight" audio. Since all audio reaching the Line and Monitor outputs would flow through the echo or reverb unit if it were patched between **Master Out** and **Master In**, there would be no such blend; only delayed audio would reach the console's Line and Monitor outputs.

GRAPHIC EQUALIZATION

"Graphic" equalization consists of a multi-frequency equalizer or filter. Unlike input equalizers, a graphic equalizer can simultaneously operate at 1-octave or 1/3-octave frequencies throughout the audible range. Some units provide only attenuation, most provide attenuation and boost.

Graphic equalization is used to eliminate resonant peaks and dips in the loudspeakers and the listening environment, thereby reducing the tendency to feedback. In many cases, the addition of graphic equalization will enable average sound levels to be elevated by over 10dB before feedback commences. Another use of graphic equalization is to contour the frequency response of the console's output to obtain the most pleasing sound quality.

Usually, each speaker feed (**Echo Out** and **Monitor Out**) requires its own channel of professional type graphic equalization installed between the console output and the power amplifier. Stage monitor feeds, for example, may require considerably more high frequency roll-off than audience feeds because the stage tends to bounce highs back at the microphones.

Hi-fi type graphic equalizers (such as the S.A.E. Mark 2700, the Soundcraftsmen RP-2212, etc.) can be installed between the **Master Out** and **Master In** jacks. However, the speakers will have to be fed from the **Line Out** rather than the **Mon Out** because the monitor mix would contain a very confused blend of different equalization that would be very difficult to set up properly.

COMPRESSION AND LIMITING

Compressors and limiters shrink the dynamic range of program material. In practice, a compressor is used to raise the average volume of low level program material, without overdriving the amplifiers during high volume passages. Compression increases average listening levels (apparent loudness) without actually requiring more powerful amplifiers. Limiting does not change the average program level; it does prevent peaks from exceeding a pre-set limit, or at least it becomes very difficult for peaks to rise very far beyond that limit. Limiting permits higher power settings to be used without risk of overdriving the loudspeakers or clipping the power amplifiers, tape machine inputs, etc. Limiting and compression utilize similar circuitry, so many manufacturers include both functions in the same package.

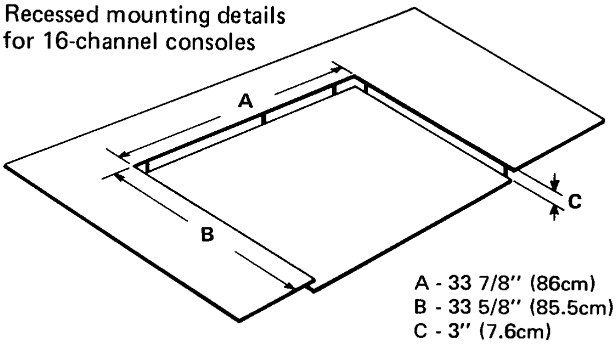
Compression and limiting may be used for tape recordings and for sound reinforcement. Professional type equipment can be connected between the console's **Monitor**, **Echo** or **Line Out** jacks and the power amplifier inputs or the tape machine record inputs. Hi-fi type units (such as the dbx 161) can be connected between the **Master In** and **Master Out** jacks. The record in and out of some compression/expansion type hi-fi noise reduction units can be used, in which case experimentation with a high-frequency roll-off filter (or equalizer attenuation) may yield a better sound quality; this is due to high-frequency pre-emphasis which is included in most noise reduction systems. Alternately, the pre-emphasis may be defeated (consult with the manufacturer of the device to determine whether this is feasible).

INSTALLATION

PLACEMENT OF EQUIPMENT

The PM-1000-16 is a fully portable, self-contained unit. It may be placed on a table top or shelf measuring at least 35" (90cm) wide by 35" (90cm) deep, and at any convenient working height. If desired, a shelf or table may be cut out to form a permanent, recessed mounting, as illustrated. This type of mounting has a lower profile, and is quite attractive since the rosewood cabinet remains visible. (In order to mount the console in the recess, the carrying handles must first be removed.) Whether recessed or table-top mounted, the console should be on a level surface, with sufficient rear-panel clearance to accommodate the input and output cables. Consult the "General Applications Section" and the diagrams further on in this section for suggested location of the console within the room.

Recessed mounting details for 16-channel consoles



A - 33 7/8" (86cm)
B - 33 5/8" (85.5cm)
C - 3" (7.6cm)

POWER MAINS

U.S.A. and Canadian models are designed to operate with 110-120V AC, 50-60Hz power mains. The console must be AC grounded for safety and for proper shielding; a 3-wire power cord is provided for this purpose. If a 3-wire outlet is not available, or if there is any chance the outlet may not be grounded, a separate jumper wire must be connected from the console chassis to an earth ground. Cold water pipes generally provide good grounds, but hot water pipes, pipes on the user side of a water meter, or PVC pipes should be avoided. When in doubt, drive a length of copper pipe into moist earth to obtain a ground point, burying at least 5' (1.5m) of pipe.

The console should be connected to the power mains ONLY IF THE VOLTAGE AND LINE FREQUENCY ARE CORRECT. The **Power** Switch on the meter panel should be shut off before connecting the console to the mains. As a precaution, the console should be disconnected from the mains while audio cables are being installed.

CAUTION. Severe over-voltage or under-voltage in the power mains can damage the console's circuitry. A safe practice is to first check the AC line before connecting the power cord. Using an RCA Power Line Monitor, or any suitable AC voltmeter, the power line must measure more than 100Vrms and less than 135Vrms.

If power line voltages do not fall within the 100V to 135V range, a variac or some other device must be used to correct the voltage. Failure to observe this precaution may damage the console, and will void the warranty.

A relatively minor power supply modification allows the use of 220-240V AC mains. Described in Section Seven of this manual,the modification is to be performed only by qualified service technicians.

THEORY OF GROUNDING

Careful grounding procedures are essential for proper operation, not only of the PM-1000, but of the entire audio system. Many grounding techniques exist, and certainly there are several ways to achieve a satisfactorily grounded audio system. Several books have been written on the subject. For further information (to complement what we are about to present) consult the following sources: THE AUDIO CYCLOPEDIA by Howard M. Tremaine (Pub. Howard W. Sams); the ALTEC ACOUSTA-VOICING MANUAL Ed. by John M. Eargle & Mark E. Engebretson (Pub. Altec Corp.); GROUNDING & SHIELDING TECHNIQUES IN INSTRUMENTATION by Ralph Morrison (Pub. Wiley).

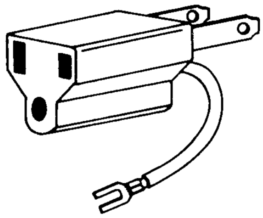
CAUTION. In any audio system installation, governmental and insurance underwriters' electrical codes must be observed. These codes are based on safety, and may vary in different localities; in all cases, local codes take precedence over any suggestions contained in this manual. As set forth in the PM-1000 Warranty, Yamaha International Corporation shall not be liable for incidental or consequential damages, including injury to persons or property, resulting from improper, unsafe or illegal installation of the PM-1000 or of any related equipment; neither shall the Corporation be liable for any such damages arising from defects or damage resulting from accident, neglect, misuse, modification, mistreatment, tampering or any act of nature.

The following grounding scheme is presented in the belief that it is one of the more simple, yet effective methods available, but by no means is it the only effective method.

Ground loops, multiple paths to the AC main ground (earth ground) often tend to induce hum and allow noise to develop. In severe instances, equipment may begin to oscillate due to ground loops; oscillation can cause distortion and damage amplifiers and loudspeakers. One way to avoid ground loops is to make certain that there is just one path to AC ground (earth ground) for the entire audio system.

The PM-1000 chassis provides a convenient point from which all other equipment in the system can derive its ground. The auxiliary equipment chassis is first isolated from the AC main ground, if necessary; then the auxiliary chassis are all grounded to the PM-1000 chassis via the shields of the interconnecting audio cables. To insure the success of this scheme, the PM-1000 chassis must be well grounded either through the ground lead in its AC cord, or through an earth ground attached to the console chassis.

Much of the auxiliary equipment sold today is equipped with 2-wire AC power cords, which implies that the equipment is isolated from the AC main ground. If the equipment has a 3-wire AC cord, its chassis is probably grounded to the power mains through the rounded, center prong of the power plug; a 3-prong to 2-prong adapter may be used to interrupt that ground.



WARNING: When a chassis is not grounded directly to the AC mains, it **must** be grounded to the PM-1000 chassis by the shields of interconnecting audio cables; the PM-1000 then links the remote equipment to the AC main (earth) ground. Should the shield of a cable break, or in the event a cable is disconnected, it is possible for dangerous potential differences (lethal AC voltages) to develop between the remote chassis and any other grounded device. Therefore, it is extremely important, especially with guitar amplifiers, that continuity between the remote chassis and the PM-1000 chassis be maintained at all times (even when power switches are OFF). As a precaution, every chassis and microphone case should be tested with an ohmmeter to assure it is grounded to the power main (or earth) ground. In the previously described grounding scheme, it should be sufficient to first establish the PM-1000's continuity to ground, and to then make certain there is virtually no resistance between the remote chassis and the PM-1000 chassis.

AUDIO CONNECTORS AND CABLE TYPES

The PM-1000 is fitted with only four types of audio connectors: 3-pin XLR male, 3-pin XLR female, 2-conductor standard phone jacks and 3-conductor (stereo) phone jacks. The circuits which are associated with these connectors, and the proper mating connectors, are listed on the following table.

In most installations, 2-conductor shielded cable is recommended for all XLR connections. Belden No. 8412 or its equivalent is an excellent cable due to its heavy construction, and this type of cable should be used for all portable applications. A lighter duty cable, Belden No. 8451 or its equivalent, is suitable for permanent installation only. "Snake" cables should be handled very carefully; cables containing multiple shielded pairs are not recommended because the leads tend to be fragile, and a broken conductor cannot be replaced.

In order to minimize crosstalk, low level (microphone)cables should be physically separated from high level (line) cables whenever possible. If low and high level cables must be run parallel and in close proximity to one another, they should be bundled separately.

INTERFACE WITH AUXILIARY EQUIPMENT

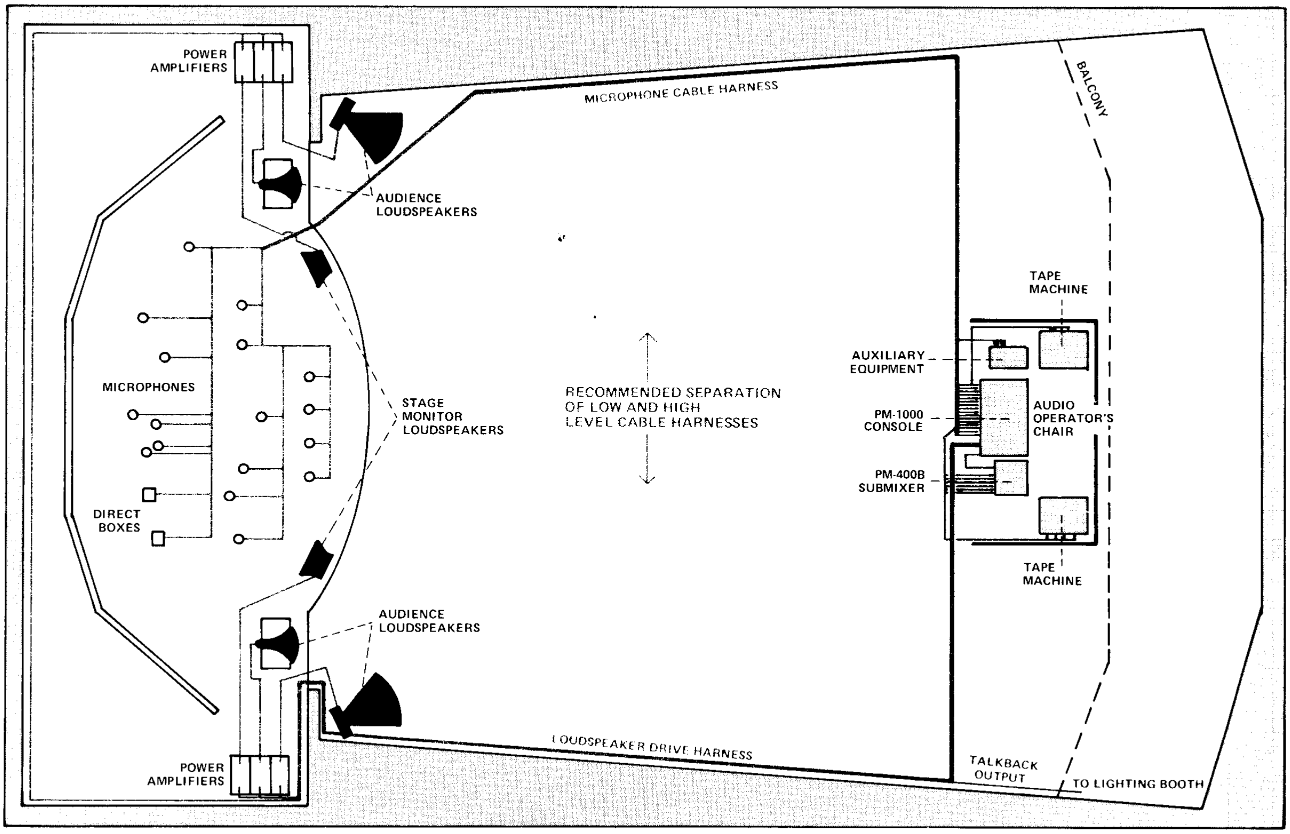
Auxiliary equipment, such as power amplifiers, compressors, equalizers, and tape machines, may be fitted with the same type of XLR or phone connectors used in the PM-1000. However, several other types of connectors are commonly used, including: Tip/Ring/Sleeve phone jacks, RCA-type pin jacks, and terminal strips. The cable preparation for interface of the console to all standard connector types is graphically illustrated further on in this section.

Typical audio systems for reinforcement and recording are discussed in the "General Applications" Section of this manual. The layout of these systems, and the block diagrams of the actual equipment connections, are illustrated in the following diagrams. Many variations are likely to be used, and these illustrations are intended only as guides.

AUDIO CONNECTORS USED TO MATE WITH PM-1000.

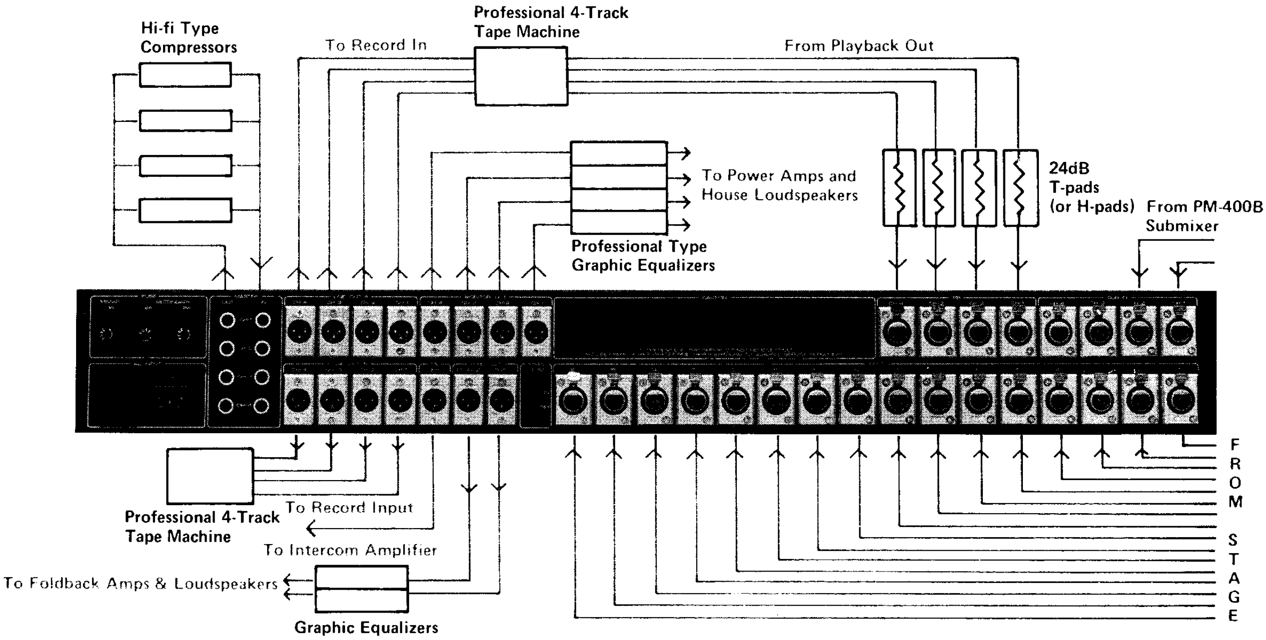
Circuit	Cannon Mfgr. No.	Switchcraft Mfgr. No.	Connector Description
Channel In	XLR-3-12C	A-3-M	3-pin male professional audio connector.
PB In	XLR-3-12C	A-3-M	3-pin male professional audio connector.
Sub In	XLR-3-12C	A-3-M	3-pin male professional audio connector.
Talkback In	XLR-3-12C	A-3-M	3-pin male professional audio connector.
Line A Out	XLR-3-11C	A-3-F	3-pin female professional audio connector.
Line B Out	XLR-3-11C	A-3-F	3-pin female professional audio connector.
Monitor Out	XLR-3-11C	A-3-F	3-pin female professional audio connector.
Echo Out	XLR-3-11C	A-3-F	3-pin female professional audio connector.
TB Out	XLR-3-11C	A-3-F	3-pin female professional audio connector.
Master Out	-----------	280	2-conductor standard 1/4" phone plug; shielded case and integral cable clamp.
Master In	-----------	280	2-conductor standard 1/4" phone plug; shielded case and integral cable clamp.
Headphone Out	-----------	290	3-conductor (TRS) 1/4" phone plug; shielded case and integral cable clamp.

Floor Plan of Theatre, illustrating sound system arrangement with PM-1000 in the midst of the audience. (Typical of many portable installations.)



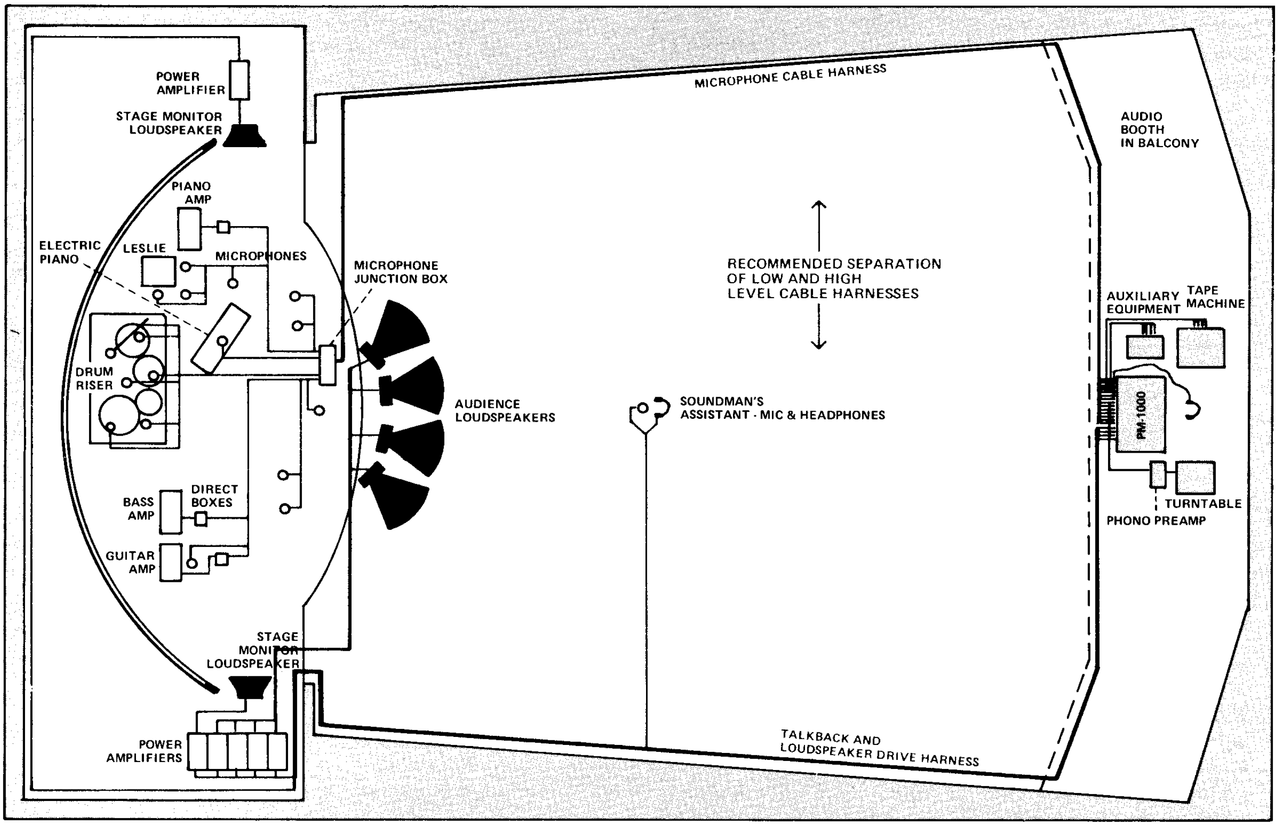
Note: Console is sometimes located closer to the stage.

PM-1000 Connections for Above Installation.

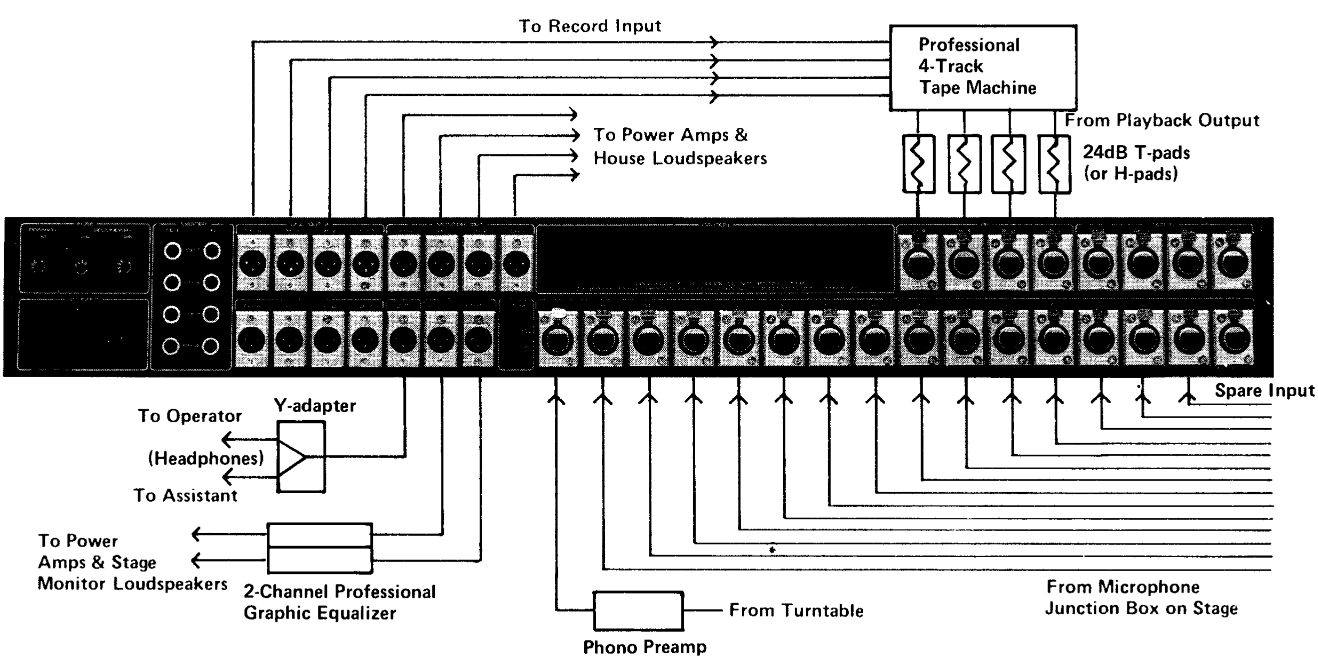


The use of two tape machines assures continuity of recording when a reel ends, or in the event of a problem. Note that the playback input has attenuation pads to reduce the 4-track output to the proper level. All power amplifiers are remotely situated, near the loudspeakers they drive.

Floor Plan of Theatre, illustrating sound system arrangement with PM-1000 in the balcony. (Typical of many fixed installations.)

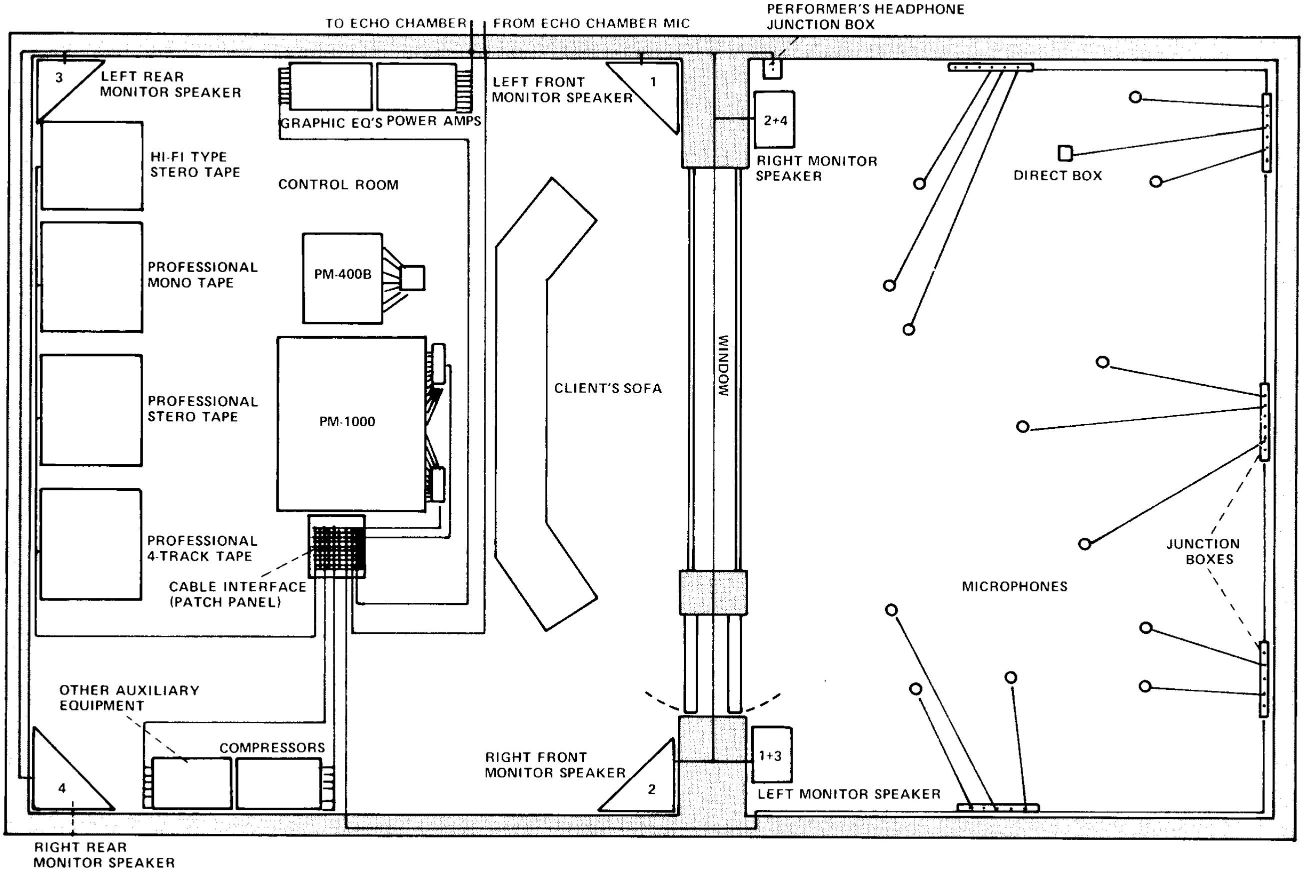


PM-1000 Connections for Above Installation.



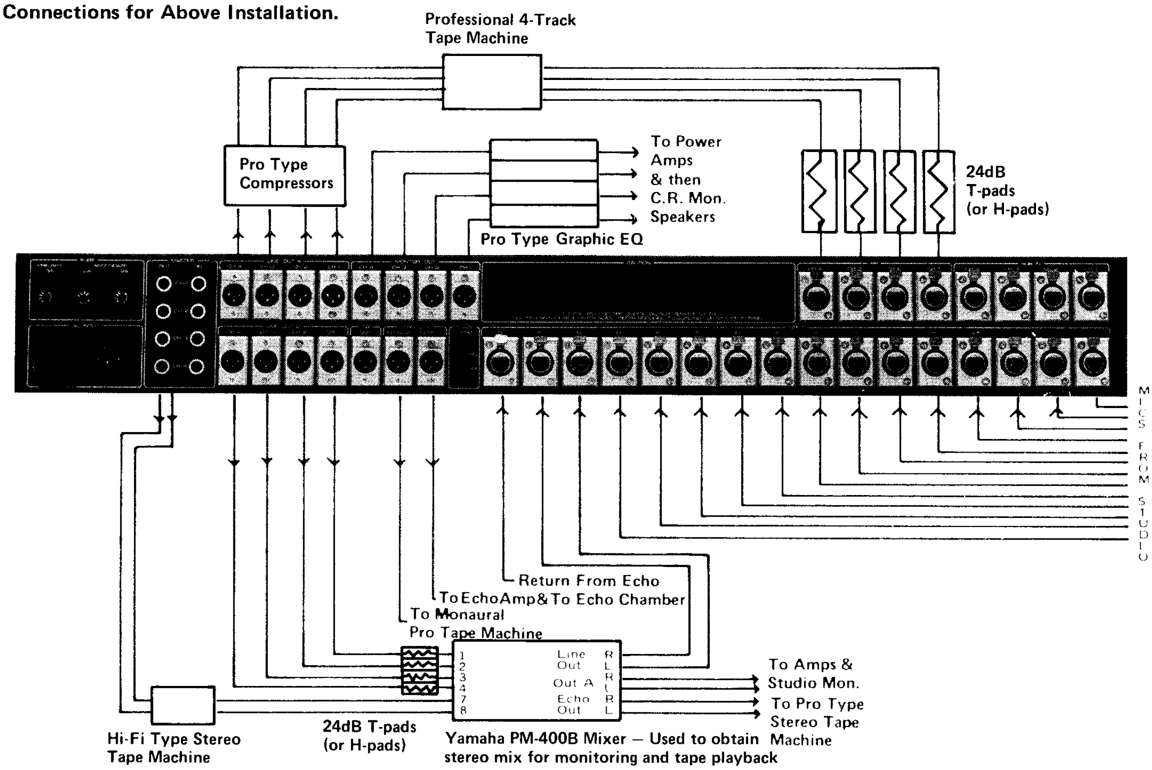
The turntable may be used to provide background music during breaks, or for special effects and performance back-up music. Note the attenuation pads at the playback input. All power amplifiers are remotely situated, near the loudspeakers they drive. MASTER jacks, LINE B OUT jacks, and SUB IN jacks are not needed in this set-up.

Floor Plan of recording studio illustrating typical PM-1000 installation.



NOTE: While shown in close proximity in this diagram, input and output cable harnesses should actually be physically separated and run at right angles to the greatest practical extent.

PM-1000 Connections for Above Installation.



The PM-400B serves as an auxiliary monitor mixing facility in this installation. Not shown are the headphone outputs. The PM-400B headphone output can be used for studio phones.

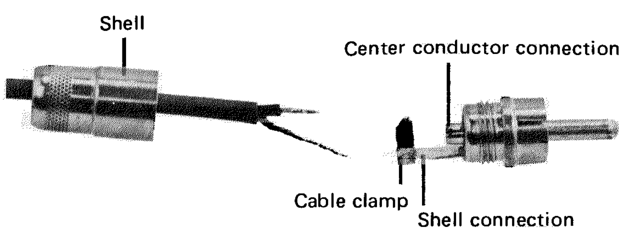
CABLE AND CONNECTOR WIRING CONFIGURATIONS

The preparation of complete cables, with connectors properly installed, is the key to reliable and trouble-free operation of any sound system. For this reason, the following illustrations are included. Experienced audio technicians may wish to review these illustrations, even if they already know how to wire connectors. A few moments of extra care here can save hours of trouble-shooting later on.

As a rule, the amount of insulation removed and the length of exposed cable should be minimized. This reduces the likelihood of short circuits and improves the ability of the clamp to grip the cable firmly. Enough heat should be used to obtain a free flow of solder, but allow leads to cool quickly after solder flows to avoid melting insulation. After each connector has been completely wired, the cable should be tested with an ohmmeter or a cable tester. Continuity between the various conductors and their associated connector pins must be established, and there should be infinite resistance (an open circuit) between all connector pins. In most cases, especially in portable installations, XLR connectors should not conduct at all between the shell and pin 1. This avoids grounding problems from inadvertent touching of the shell to other devices.

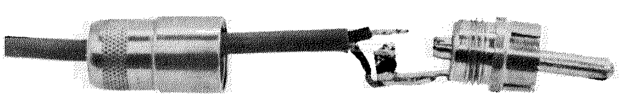
Cables to be connected to terminal strips should be prepared by stripping the ends and installing crimp-on or preferably, solder type lugs. If there is any chance the cable will be strained, use a cable that is constructed with internal strain relief cord, such as Belden No. 8412. Crimp a lug onto the cord, and secure the lug to an unused terminal. (The cord should be drawn slightly tighter than the wire leads in order to take the strain first.)

WIRING AN RCA-TYPE PIN PLUG*

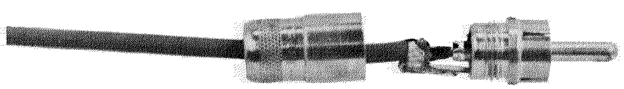


Parts identification and cable preparation.

Strip approximately 1/2" of outer insulation. Unwrap or unbraid the shield and form a lead. Strip approximately 5/16" of insulation from the center conductor. Tin both leads.



Solder the shield to the outer surface of the shell connection, allowing enough free shield to wrap the cable around to the center of the connector. Cool the connection immediately with pliers.

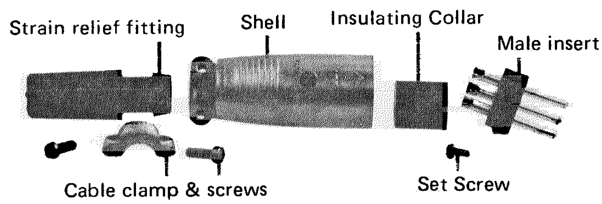


Insert the center conductor in the hollow pin, and fill that end with solder. Cool the connection immediately with pliers. Clean any solder splashes and inspect for burned insulation. Pinch the clamp around the outer insulation with pliers, firmly, but not so tight as to cut the insulation.

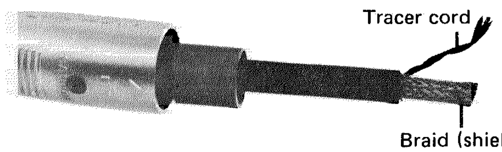
Slide the shell forward and screw it tightly to the threaded plug.

**Switchcraft No. 3502 connector illustrated. Many large diameter cables are more easily wired to "simple" RCA type pin plugs without a shell (Switchcraft No. 3501M, or equivalent). The braid can then be soldered directly to the shell of the plug.*

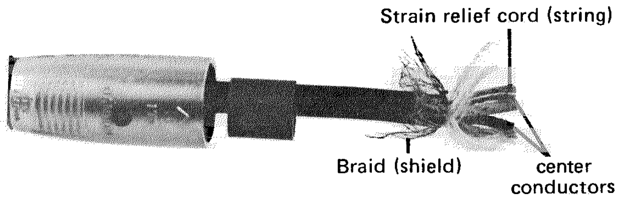
WIRING A MALE XLR CONNECTOR



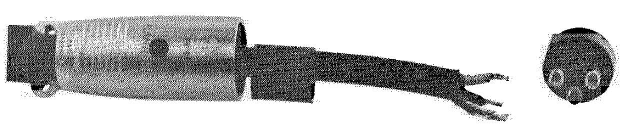
Parts identification (as the connector is usually packaged).



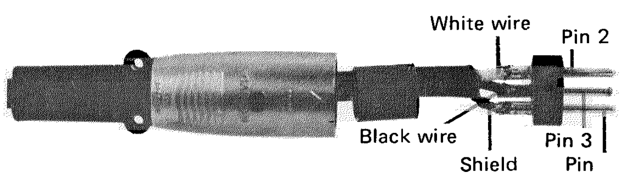
Insert strain relief in rear of shell. Then slip shell onto cable end, followed by insulating collar. Strip outer insulation 1/2". (No. 8412 cable illustrated here.)



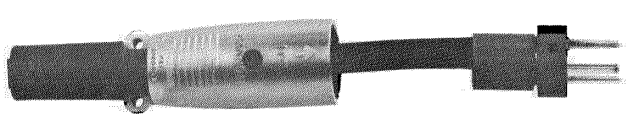
Cut tracer cord, unbraid shield, cut cotton strain relief cords.



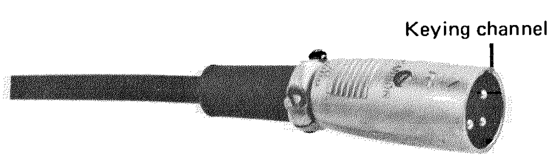
Strip approximately 1/4" of insulation from center conductors, tin, and trim to approximately 1/8" exposed wire. Then twist shield, positioning it in the correct orientation to mate with the insert. After tinning the shield, cut it to the same length as the center conductors.



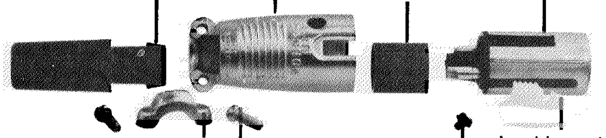
Solder the center conductors to their respective pins, using just enough solder to fill the end of the pins. Yamaha's wiring standard dictates that the black lead mates with pin 3 and the white (or red) with pin 2 (see footnote on page 10 of this section). Then solder the shield to pin 1. Clean any solder splashes and inspect for burned insulation.



Slide the insulating collar forward, up to the flange of the male insert. The outer cable insulation must be flush with, or covered by the end of the insert. If any of the center conductors are visible, the cable clamp may not be able to firmly grip the cable. Then slide the collar back into the shell.

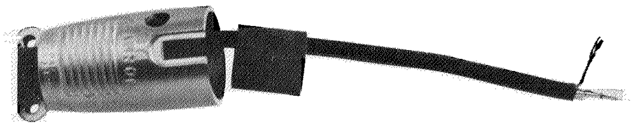


Slide the shell forward, orienting its internal keying channel with the raised lip (key) on the insert. Secure the insert in the shell with the set screw. Place the cable clamp over the rear of the shell, with careful attention to the clamp's orientation; a raised lip inside the clamp should be aligned immediately over a lip in the shell for thinner cable (No. 8451). The clamp should be turned around for heavier cable (No. 8412) to provide clearance. Insert the clamp screws and tighten fully.

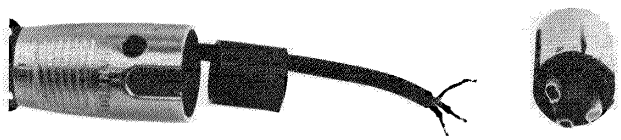


WIRING A FEMALE XLR CONNECTOR

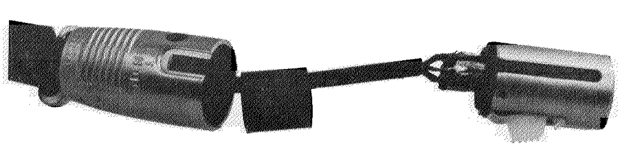
Parts identification (as the connector is usually packaged).



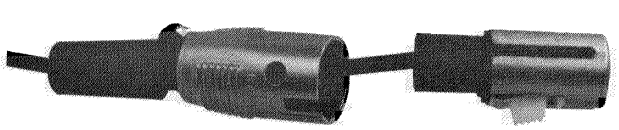
Insert strain relief in rear of shell. Then slip shell onto cable end, followed by insulating collar. Strip outer insulation approximately 9/16". (No. 8451 cable illustrated here)



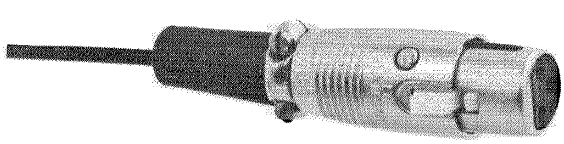
Pull off foil wrap. Strip approximately 5/16" of insulation from the center conductors, leaving approximately 1/4" of insulation between the bare wire and the outer insulation. Tin the center conductors, and trim so that about 1/8" bare wire remains. Then tin the shield conductor, orienting it with the center conductors so they are aligned with the proper pins of the insert. Cut the end of the shield so that it extends 1/16" beyond the center conductors.



Solder the center conductors to their respective pins, using just enough solder to fill the end of the pin. Yamaha's wiring standard dictates that the black lead mates with pin 3, the white (or red) lead with 2 (see footnote on page 10 of this section). Then solder the shield to pin 1. Clean off any solder splashes, and inspect for burned insulation. Insert the locking tab in the female insert, as illustrated, with small nib facing front of connector.

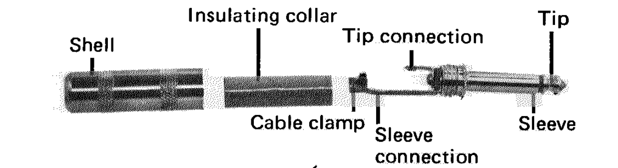


Slide insulating collar foward, up to rear edge of female insert. The outer insulation of the cable must be flush with, or covered by the end of the insert. If any of the center conductors are visible, the cable clamp may not be able to grip the cable firmly, and the connector leads will soon fatigue. Then slide the collar back into the shell.

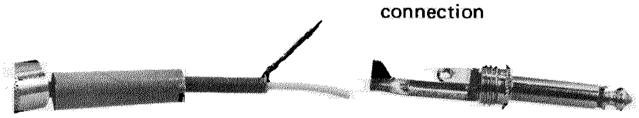


Slide the shell forward, orienting the notch in the shell with the locking tab in the insert. Secure the insert in the shell with the set screw. Place the cable clamp over the rear of the shell, with careful attention to the clamp's orientation; a raised lip inside the clamp should be aligned immediately over a lip in the shell for thinner cables (No. 8451). For heavier cables (No. 8412), the clamp should be turned around to offset the lips and provide more clearance for the cable. Insert the clamp screws and tighten fully.

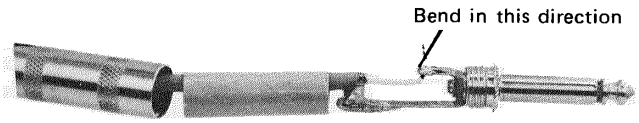
WIRING A STANDARD PHONE PLUG (2-conductor)



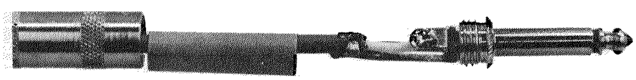
Parts identification.



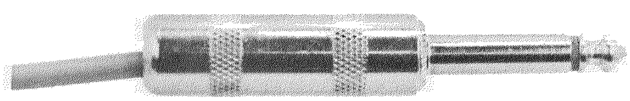
Slide shell, then insulating collar over cable end. Strip outer insulation for length equal to length of sleeve connection. Unwrap or unbraid shield, twist to form lead.



Position outer insulation just ahead of cable clamp, strip center conductor from point just behind tip connection. Tin center conductor and shield. Bend shield as illustrated, solder to outer surface of sleeve connection. (Cool immediately with pliers.) Insert center conductor in tip connection, solder, cut end flush. Bend the end of the tip connector (slightly) toward the sleeve connection to help prevent the burr (from the cut wire) from cutting through the insulating collar.

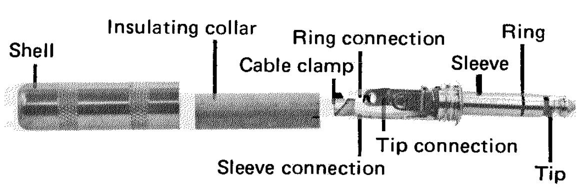


Using pliers, bend cable clamp around outer insulation. Clamp should be firm, but not so tight as to cut insulation.



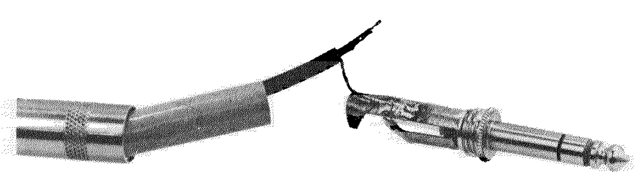
Slide insulating collar forward, until flush with rear of threads. Slide shell forward, screw tight to plug assembly.

WIRING A TIP, RING & SLEEVE PHONE PLUG (3-conductor)

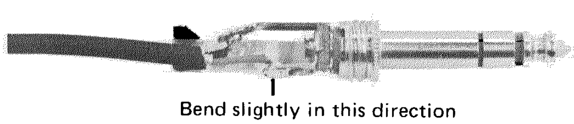


Parts identification.

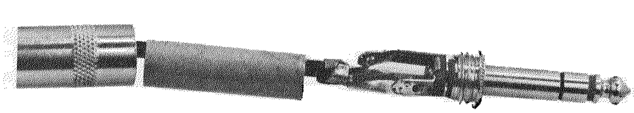
Slide shell and insulating collar over cable end. Strip outer insulation for length equal to length of sleeve connection. Remove any tracer cords and strain relief cords. Form lead from shield. Hold cable with outer insulation just ahead of cable clamp, and strip the red (or white) conductor just behind the tip connection. Then strip the black conductor just behind the ring connection. Tin all leads, and cut the center conductors so approximately 1/8" of bare wire remains.



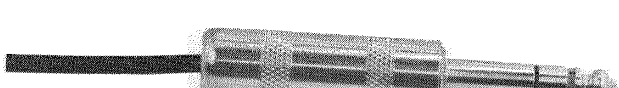
Solder the shield to the outer surface of the sleeve connection, allowing enough free shield to bend around to the other side of the cable clamp. Cool the connection immediately with pliers.



Insert the center conductor leads in their respective connection points, and solder in place. Trim the leads flush. Bend the end of the tip connection (slightly) toward the ring connection to help prevent the burr (from the cut wire) from cutting through the insulating collar.



Using pliers, bend the cable clamp around the outer insulation. The clamp should be firm, but not so tight as to cut the insulation.



Slide the insulating collar forward, until flush with rear of threads. Slide the shell forward, and screw tightly onto plug.

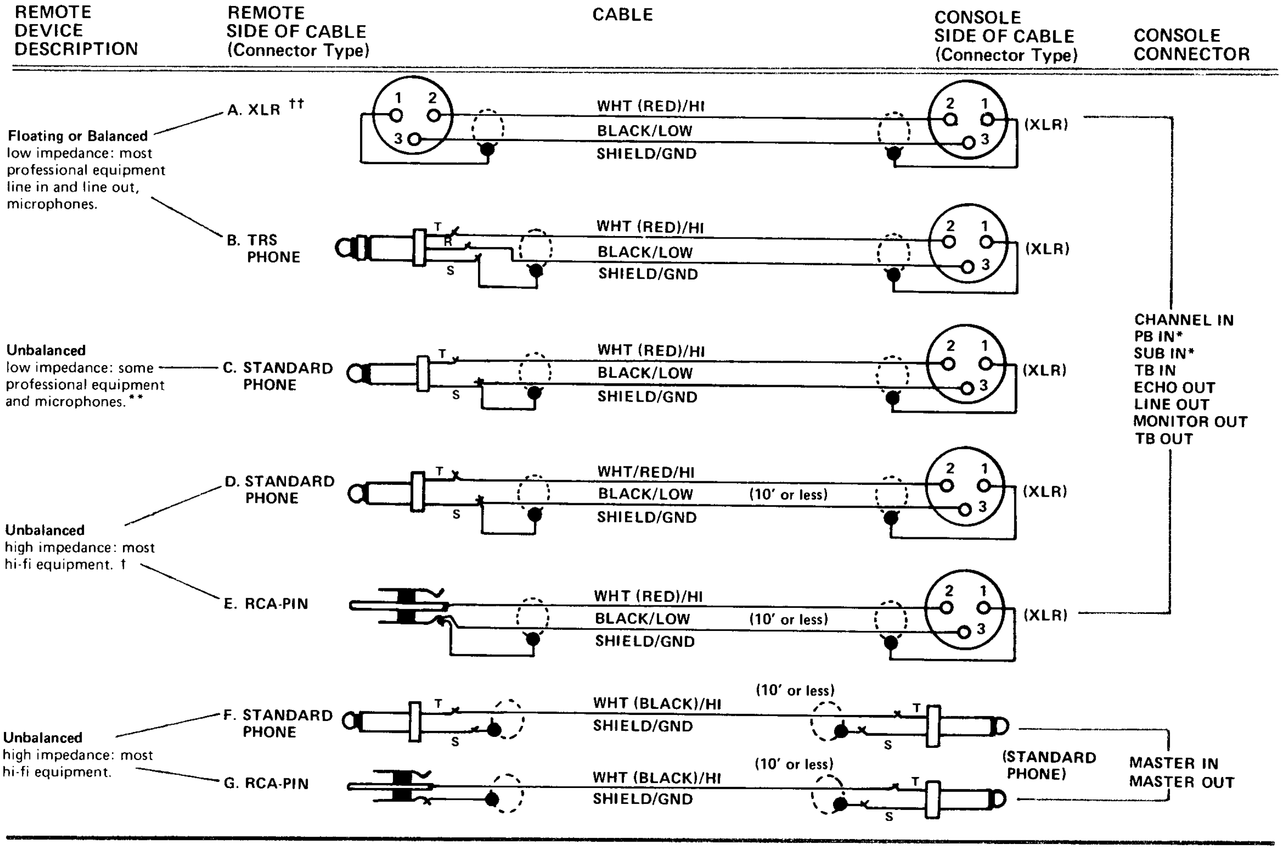
INTERCONNECTIONS

The PM-1000 channel inputs accept balanced or floating input sources directly, with no need for auxiliary transformers. Most professional microphones, and most professional (low impedance) audio equipment falls in this category. The PM-1000's phantom power may be switched on or off when using any of this equipment, and no difference in performance will be observed.

Unbalanced equipment may also be connected directly to the console's channel inputs. However, the phantom power must remain off. If switched on, the unbalanced line places a load on the phantom supply, inducing hum in that input (and, while unlikely, possibly damaging the auxiliary equipment). In order to utilize an unbalanced device at the same time condenser microphones are connected to other inputs, with the console's phantom power turned on the device should be connected to the channel inputs through a matching transformer. This transformer prevents the console from loading a high impedance output, and it prevents the unbalanced line (high or low impedance) from loading the phantom supply.

Attenuation pads, when used in conjunction with auxiliary transformers, should always be installed on the high level side of the transformer. This reduces the signal level passing through the transformer, which tends to improve the performance of the transformer.

Connector and cable configurations recommended for use with the PM-1000. These cables are based on the use of auxiliary equipment that is isolated from the AC power mains.



†† This wiring configuration (Pin 2 high, Pin 3 low) matches the PM-1000's wiring and DIN standards. Much of the equipment in the U.S.A. is wired with Pin 3 high and Pin 2 low (shield is still Pin 1). In most cases involving the PM-1000, this makes no difference. However, interconnections between other manufacturer's equipment may require that Pins 2 and 3 be reversed; consult the manufacturer's literature.

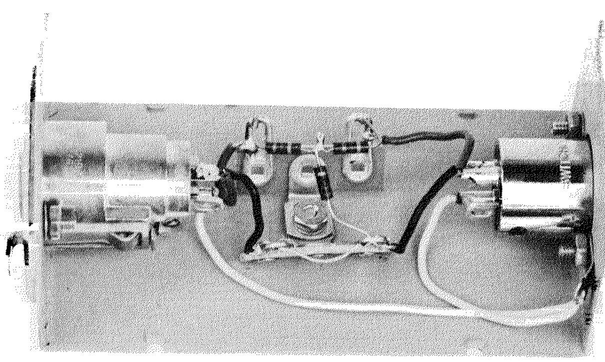
* Install 24dB pad between cable and console input.

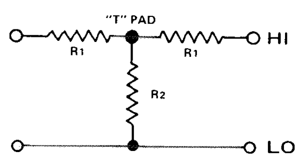
** Transformer between cable and channel input recommended if phantom power is to be used.

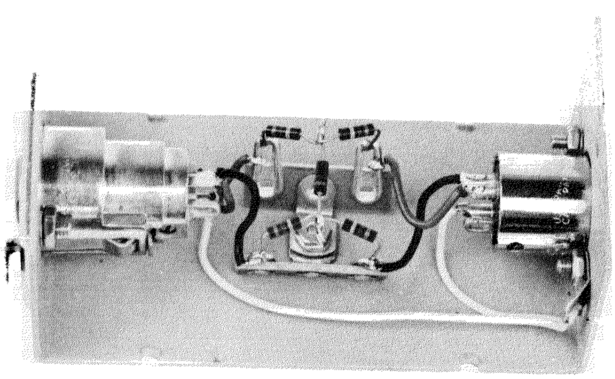
† Use this cable at remote equipment, and install matching transformer with high-Z side toward remote equipment. Then use cable A to join the low-Z side of the transformer to the console. Use of the transformer at the high-Z location allows long cable runs to the low-Z connection.

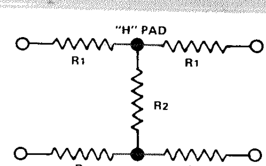
ATTENUATION PAD RESISTOR VALUES

db LOSS	R1 T (ohms)	R1 H (ohms)	R2
.5	16	8.2	10k
1	33	18	5.1k
2	68	33	2.7k
3	100	51	1.6k
4	130	68	1.2k
5	160	82	1k
6	200	100	820
7	220	110	680
8	270	130	560
9	270	150	470
10	300	150	430
12	360	180	360
14	390	200	240
16	430	220	200
18	470	220	150
20	510	240	120
22	510	240	91
24	510	270	75
26	560	270	62
28	560	270	47
30	560	270	36
32	560	300	30
34	560	300	22
36	560	300	18
38	560	300	15
40	560	300	12
50	620	300	3.6









PADS, TRANSFORMERS AND DIRECT BOXES

ATTENUATION PADS

The simplest pad that can be successfully used with the PM-1000 is the T-pad. T-pads unbalance true balanced lines, so they should not be used with equipment having center tapped, balanced transformer outputs or with condenser microphones. (For condenser microphones, H-pads provide better balance of the dc voltage in the cable. However, the PM-1000's **Input** Level switches include sufficient padding for virtually any condenser microphone, so H-pads should not be required.) Although other types of pads, such as "O" pads, will work satisfactorily, they generally yield no better performance.

T-pads should always be installed near the input of the device they are feeding, with a short length of cable on the low-level side of the pad. This maintains a high signal level in the longer transmission cable, reducing the effect of any induced hum or noise.

The pads illustrated are designed to attenuate the specified levels in 600-ohm lines. Commercially manufactured pads are available (consult your Yamaha dealer). When connected between a 600 ohm and a high impedance termination, pad attenuation values will remain fairly accurate. However, because pad attenuation is a function of the line impedance, pads should be tested in a given circuit to ascertain the actual attenuation. (T-pads do not retain their accuracy when inserted between high impedance terminations, but this condition should not occur with the PM-1000.)

Pads are normally constructed with 1/4Watt, 10% tolerance resistors. Higher power or more precision components may be used, but they cost more and are usually unnecessary.* It is possible to construct a pad within an XLR connector, but the extremely tight fit can adversely affect reliability. A mini box fitted with male and female XLR connectors is an easier to build, more rugged housing for a pad. Stranded wire is recommended.

Illustrated are two typical pad construction techniques. For most applications, involving the PM-1000 it will be sufficient to construct only a few types of pads; 20, 24 and 40dB pads cover almost any requirement. (Consult table for schematic and resistor value information.)

**For outputs with CONTINUOUS sine wave levels above +24dBm, use 1/2W resistors; for CONTINUOUS sine wave levels above +30dBm, use 1W resistors.*

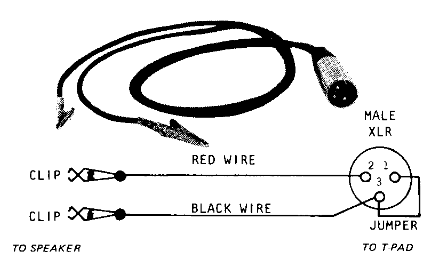
DIRECT BOXES

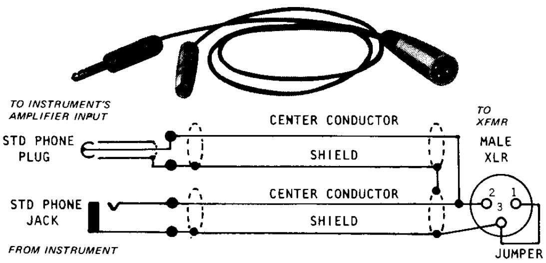
The term "direct box" refers to an adapter which permits a power amplifier to drive a relatively lower level input. Direct boxes are most often connected between the speaker output of electric instrument amplifiers and the input of a mixer, such as the PM-1000 channel inputs. By using the amplifier's speaker output, the reverb, tremolo, brightness, and any other sound characteristics are conveyed to the mix. The standard direct box consists of an attenuation pad that reduces the power, and an impedance matching transformer to correctly terminate the mixer's input.

A variation of the direct box just described is not meant for connection to the amplifier output. Instead, it contains a "Y"-adapter that enables it to be inserted between the instrument and the amplifier. The "Y"-adapter taps the instrument output and feeds it through an impedance matching transformer into the console.

Rather than building the standard direct box, we suggest using a combination of a T-pad and a matching transformer box (described in the following pages). The T-pad should be placed between the transformer's high impedance input and the output of the amplifier. Approximately 20-40dB of padding is needed to prevent transformer saturation (the pad value depends on amplifier power). Then, another 20dB (approximately) of voltage level is lost in the transformer due to the impedance matching. As a rough guide, the 40dB T-pad should be used with amplifiers rated from 50 to 200 Watts (continuous sine wave(rms) power), and the 20dB pad should be used for smaller amplifiers. Small level variations are corrected with the console's input level switches. If the instrument amplifier does not have a phone jack speaker output, prepare a cable similar to Figure D shown on "Connector and Cable" Chart (under "Connections" in this section), but substitute a pair of clips for the phone plug. These clips are then attached to the speaker terminals, and the XLR connector is brought to the T-pad. If hum occurs, try reversing the clip leads on the speaker terminals.

The direct box variation can be assembled by utilizing the impedance matching transformer box, and preparing a special "Y"-adapter cable.



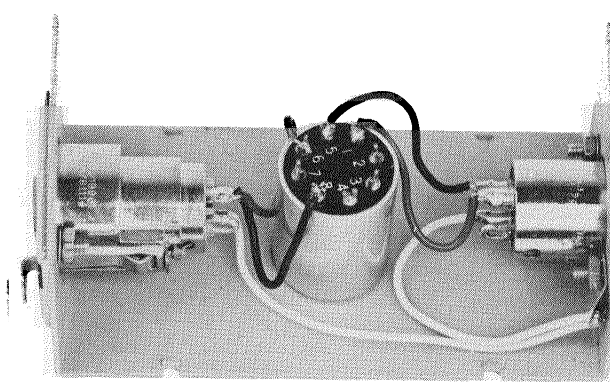


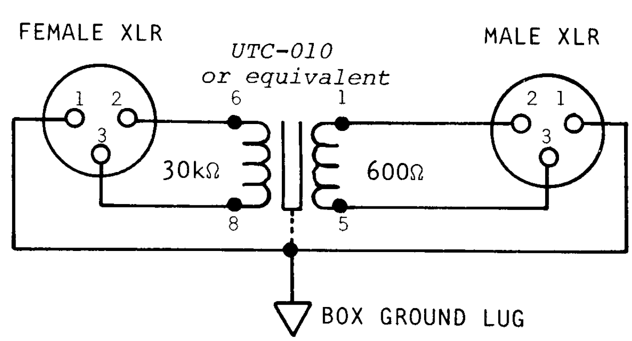
MATCHING TRANSFORMER BOX

Impedance matching transformers are manufactured by several firms. A transformer capable of handling nominal +4dB inputs (with at least +24dB peaks) should be used. Because there is an apparent loss of about 20dB in the transformer, due to voltage stepdown, the +4dB input becomes about -16dBm at the secondary, and the PM-1000's **Input** Level switch is then set at -20dB. The -20 setting is ideal because it bypasses the console's pre-amplifier, avoids any input attenuation within the console, and results in the best overall signal-to-noise ratio and the best headroom.

The transformer should have a primary impedance of approximately 30,000 ohms, with a secondary impedance of 600 ohms (for high-z microphones, a primary of 50K ohms and a secondary of 150 ohms is preferable). A UTC-010 transformer, with the UTC-019 shield, is ideal for most applications; equivalent transformers should have similar level handling and impedance characteristics. In-line transformers, such as those manufactured by Shure Brothers, may be used (although suitable cable adapters will be needed).

The transformer should be mounted in a mini box, wired to XLR connectors with standard cable, and connected to the auxiliary equipment with one of the cables previously illustrated.





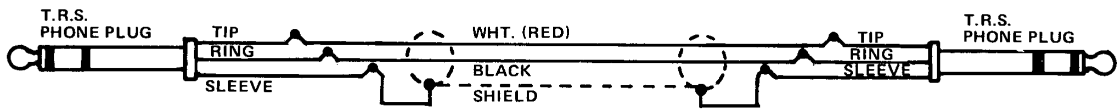
STEP-UP TRANSFORMER BOX

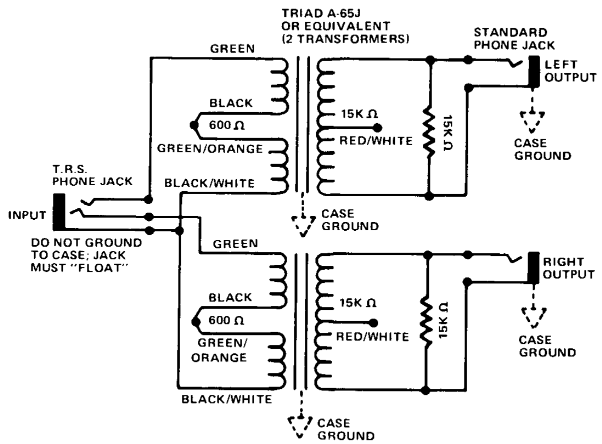
The headphone output of the PM-1000 may be used to drive power amplifiers, thereby providing additional monitoring flexibility. However, some solid-state power amplifiers having high input sensitivity (such as Crown and BGW amplifiers), may not always be driven to full output by the PHONES output. The use of a step-up transformer will avoid this potential difficulty and, at the same time, provide ground isolation between the console and the power amplifier(s).

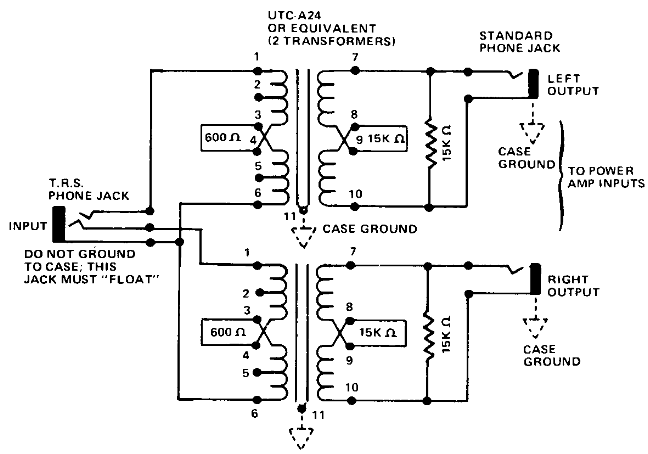
The step-up transformer box illustrated here is equipped with two standard (2-wire) phone jacks for connection to the "left" and "right" power amplifier inputs. An interconnecting cable consisting of a length of dual-conductor shielded cable and two T.R.S. phone plugs can be fashioned and used to connect the transformer box's input to the console output. The step-up transformer box should be located no closer than 5' to the power amplifier to avoid hum pickup from the amplifier's power supply. However, the cables from the transformer box to the amplifier should be no longer than 10'; since this is a high impedance circuit, use low capacity, coaxial, hi-fi type cable.

Two similar schematic diagrams are shown, one for the Triad A-65J transformer, the other for the UTC A-24 transformer. Either transformer will perform the required function, as will many similar 600-ohm to 15K-ohm transformers. The 1/4W, 10%, 15K-ohm termination resistors are installed in the box because, by terminating the transformers, they provide lower distortion and improved frequency response.

INTERCONNECTING CABLE FOR
PHONES OUTPUT TO STEP-UP
TRANSFORMER UNIT







TRANSFORMER AVAILABILITY

The matching and step-up transformers mentioned in the preceding subsections are available from many electronic parts dealers. Yamaha does not endorse specific products by citing them herein; rather, these transformers are mentioned for convenience only. If you are unable to locate the transformers from your local electronic parts dealer, contact the manufacturer at the address shown below.

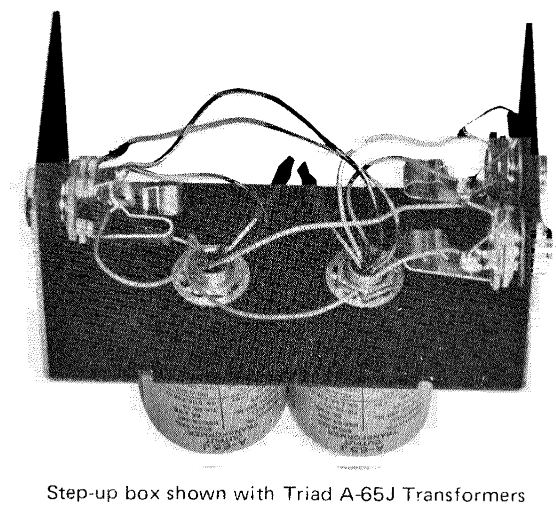
Triad
305 N. Briant St., Huntington, Indiana 46750
Phone (219) 356-6500 TWX 816-333-1532

UTC
150 Varick St., New York, NY 10013
Phone (212) 255-3500 TWX 710-581-2722

Shure Brothers, Inc.
222 Hartrey Ave., Evanston, Illinois 60204
Phone (312) 328-9000 Cable: SHUREMICRO

A line of very high quality transformers, suitable for the most critical applications, is available directly from:

Jensen Transformer Company
1617 N. Fuller Ave., Hollywood, Ca. 90046
Phone (213) 876-0059



Step-up box shown with Triad A-65J Transformers