

HOW TO USE THE CONSOLE

NOTE: The information presented in the following section of the manual complements the basic feature callouts in Section 1. In some cases, those callouts provide sufficient information, so the feature has not been included in this section.

INPUT MODULE

PHASE SWITCH

When two or more mics are receiving the same program source (or one mic is receiving leakage from the source intended for another mic) an *out-of-phase* condition may cause acoustic phasing cancellations that will be most noticeable in the low frequencies. To check inputs for correct phase, begin with all channels in the **N** mode. Assign mics 1 and 2 to the cue bus, placing the microphones close together and near a sound source that contains appreciable bass. Using **Input Level**, match the level of the two microphones. Then, listening to the blend of the two inputs in the cue headphones, reverse the phase of mic 2 by moving its phase switch to **R**. If the bass increases, leave the switch in **R** mode, and if the bass decreases, return the switch to **N**. This establishes an in-phase condition between mics 1 and 2. Repeat the procedure for mics 1 and 3, 1 and 4, and so forth until all inputs have been compared to mic 1. Line inputs may be checked in a similar fashion, listening for a decrease or increase in the bass content of the program. The absolute phase of an input is not significant, so long as the inputs are in phase with each other. Any input can serve as the "normal" or reference input.

Once the inputs are set up in phase, it may be desirable to intentionally reverse the phase of certain channels. Where numerous microphones are in close proximity, especially with drum sets, reversing the phase of adjacent mics can significantly improve the sound. The potential improvement is due to the time delay in the sound reaching two mics, a delay which causes frequency-dependent phase cancellations. These cancella-

tions are heard as an overall muddy character, and reversing the phase of one microphone may improve the sound by altering the frequencies at which cancellation occurs. When time permits, each input may be examined to hear whether reversing the phase will improve the overall sound character of the mix. This should be done after the initial set up, during the sound check. Listening is the only valid means to determine the best phase setting, and meters are not likely to help. What sounds best is best, except when the program is being monitored with two or more channels and is being recorded in a format with fewer channels.

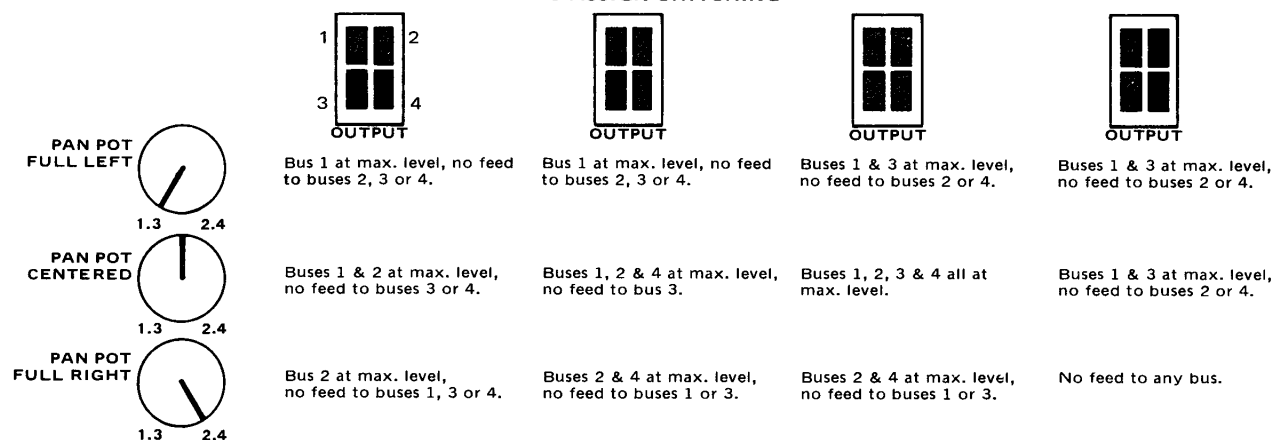
If, for example, a stereo live mix sounds good, but a monaural mix is being recorded, the monaural output should be examined to determine whether the reversal of one or more channels' phase switches will improve the mix. The stereo monitor can sound acceptable due to acoustic delays that tend to correct out-of-phase signals, whereas the same signals, when mixed together, cancel to a greater degree.

OUTPUT ASSIGN SWITCHES & PAN POT

The pan pot is designed so that when it is set a mid-rotation, it will feed the left buses (1 & 3) and the right buses (2 & 4) at maximum and equal level. This permits the pan pot to be moved off center during a performance without risk of overloading the channel(s) to which the signal is moved. In stereo sound reinforcement situations, as the pan pot is rotated to the mid-point, there is a 3dB acoustic build up at the center of the audience, making the audience aware of the intended motion. This unusually effective method of panning is an exclusive Yamaha feature.

The pan pot and output assign switches are operated together to achieve precisely the desired placement of the input source in the four program mixing buses. As the pan pot is rotated fully counterclockwise, the audio feed is gradually removed from output assign switches 2 & 4, while the feed to switches 1 & 3 remains constant. This effectively places the signal to the left. Conversely, panning clockwise gradually restores the audio feed to switches 2 & 4, reaching full level when the pot is at the mid-point. Continuing to pan clockwise, the audio feed is gradually removed from switches 1 & 3, placing the signal to the right. This panning operates in the same fashion, regardless of which bus assign switches may actually be latched; signal is only applied to the buses with latched switches.

BUS ASSIGN SWITCHING



(Black buttons indicate Bus Off, Lighter Buttons indicated Bus Assigned.)

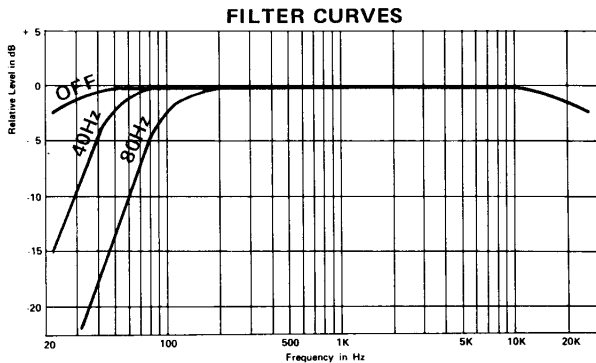
ECHO 1 & 2 MIX CONTROLS

The audio level applied to the echo mixing buses is assigned with these controls. In sound reinforcement, where the echo outputs usually drive stage monitor (foldback) amplifiers, these controls become foldback mix controls. Since the buses are fed with pre-fader, post-equalizer audio, performers are not distracted by program level adjustments.

HIGH PASS FILTER

The filter should nearly always be switched on at 40Hz. If no effect is noticed, it should be left on. 40Hz reduces subsonic signals from turn-on transients (and from warped records). It is also useful in reducing low frequency standing waves. These are acoustic resonances that occur in small rooms, like studios and smaller clubs. Standing waves will crosstalk into all microphones, creating a severe problem that baffles cannot cure. (In large rooms, the waves are sufficiently low in frequency that they are not a problem, and outdoors they do not occur.)

The 80Hz setting reduces vocal P-pops. It also reduces 60Hz hum from electric instruments without noticeably changing the sound (except perhaps on the lowest organ or bass notes.) If switching the 80Hz filter it has no audible effect, it should be left on; it is generally desirable to attenuate unused frequency response. This reduces noise. What is more, the lowest frequencies are not optimum for speech reinforcement purposes; they interfere with speech intelligibility, cause boominess to be accentuated, and waste amplifier power. The 80Hz filter cuts down on these deleterious effects without substantially altering the character of the mix.



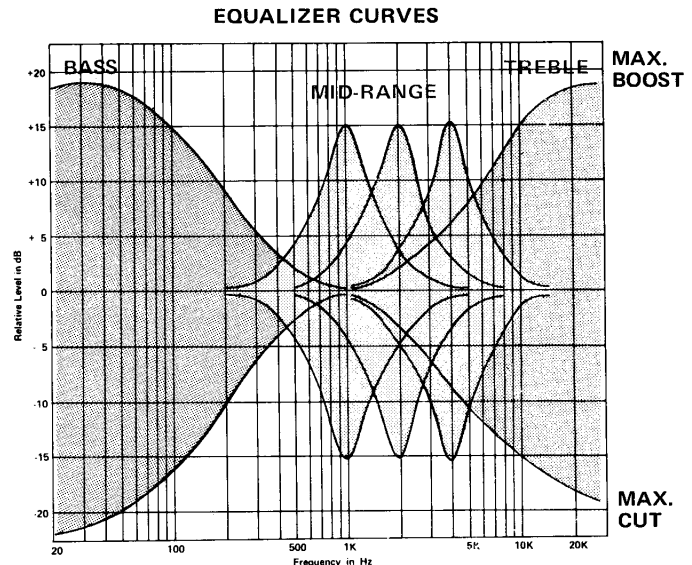
EQUALIZER

Equalizers are similar to "tone controls", but they serve a somewhat different purpose. Certainly, an equalizer can be used to enhance the sound of a given instrument or vocal, as well as an entire mix...often this is all that the equalizer does. However, equalizers can also be used to create a unique "acoustic space" for each type of instrument when instruments and vocals are mixed together. This is especially important when the sound sources share similar frequency ranges, thereby partially masking one another. By carefully tuning each input's equalizer to emphasize the dominant frequency range of the individual sound sources, there is less "competition" between the sources, hence less masking. This use of equalization not only enhances the sound of each instrument, it also tends to increase the apparent separation between channels, thereby improving the mix.

As an example of the concept just stated, consider two similar instruments, miked on separate inputs, and mixed onto the same program bus. A guitar and a bass, both boosted at 2kHz, will mix together in such a way that many of the notes will be indistinct; listeners may not be sure where one instrument leaves off and the other begins. If instead, the guitar is peaked at 4kHz, and the bass at 1kHz, then the two instruments will be easier to hear. The apparent loudness of the music may actually increase, although there is really no change in the overall power levels.

A related use of equalization is for the reduction of acoustic crosstalk (leakage). This is done by rolling off portions of the frequency spectrum that are not utilized by a particular input source. For instance, a bass drum and trumpet may be leaking into each other's microphones. The two instruments are played at loud levels, and physical separation cannot be made sufficient to cure the crosstalk. In this case, the Bass on the trumpet's equalizer should be reduced, and the Treble on the drum's equalizer should be reduced until the best sound is obtained. This type of equalization filters the leakage components without significantly affecting the character of each instrument in its own channel.

The PM-1000 channel equalizers provide low and high frequency shelving and mid-range peaking...both boost and cut. The distinction between the **Mid-Range** peaking and the **Bass** and **Treble** shelving curves can be understood by examining the diagram below. Observe the peaking curve for a 1kHz setting of the **Mid-Range Select** switch; as the frequency rises toward 1kHz, the amount of equalization increases. Then, as the frequency continues to rise past 1kHz, the amount of equalization decreases, but at 1kHz, the effect is maximum (boost or cut, depending on the rotation of the **Mid-Range** control). If the **Mid-Range** switch is moved to 2kHz, then the maximum effect occurs at 2kHz, and a 4kHz setting of the switch provides maximum boost or cut at 4kHz. Observe, too, that while the mid-range peak frequency may be a 1, 2 or 4kHz, a wide envelope of frequencies are actually equalized (light shading on diagram). This is peaking equalization.



The **Bass** and **Treble** controls provide low and high frequency shelving curves. Unlike the peaking characteristic of the mid-range, **Bass** and **Treble** curves both begin at 1kHz and the effect increases at a steep rate as the frequency decreases (**Bass**) or increases (**Treble**). The term "shelving" refers to the portion of the curve near the end of the frequency spectrum, where the rate at which equalization increases "levels off". For example, when the **Bass** control is set at maximum boost, the equalization rises rapidly from no boost at 1kHz to +15dB of boost at 100Hz. This boost continues to rise, at a reduced rate, reaching a maximum of about +19dB at 30Hz, beyond which no further increase occurs. Similarly, the **Treble** control provides a steep rise to +15dB of boost or cut at 10kHz, but continues to increase the effect somewhat less rapidly to a maximum of about +19dB at 20kHz.

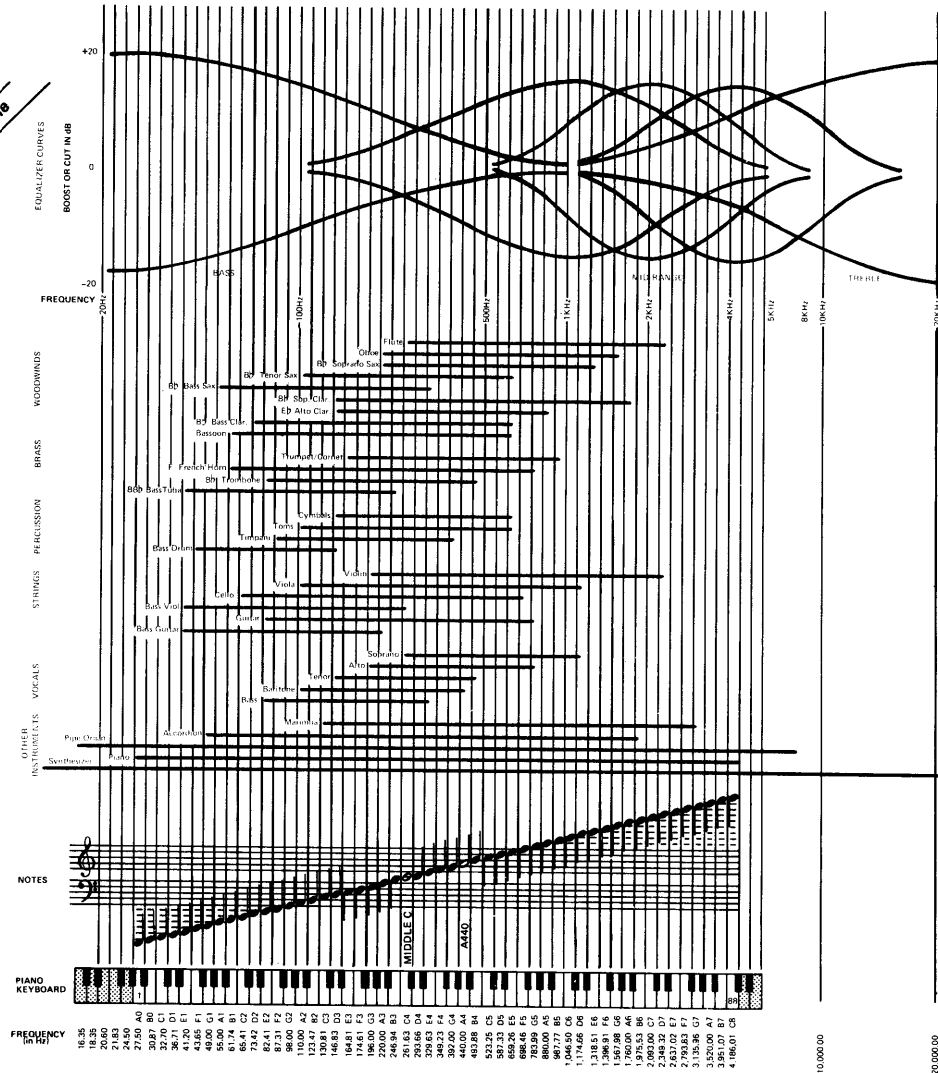
For **Bass**, **Mid-Range** and **Treble** controls, a rotational setting of less than maximum boost or cut results in a reduced degree of equalization (fewer dB of boost

or cut all along the curve); the shape of the curve "flattens out", but the frequencies affected do not change. The use of the **Mid-Range** control at 1 or 2kHz with the **Bass** control (or the **Mid-Range** control at 2 or 4kHz with the **Treble** control) does change the shape of the overall equalization curve due to overlapping regions of coverage in the two controls.

The preceding explanation of equalization is very basic, as is the diagram relating the frequency range of various instruments to the piano keyboard and the equalizer curves. This information is intended primarily as a reference point, which is all any written explanation of equalization can be. Equalizers can only be set meaningfully by listening to the sound as you make adjustments. Another very important factor in the ultimate sound is the choice of microphone and how the microphone is placed with regard to the sound source. If the correct microphone is not used, or placement is unsuitable, no amount of sophisticated equalization can make the instrument sound "right".

Channel Equalizer Controls Affecting The Characteristic

Instrument	Characteristic	Channel Equalizer Controls Affecting The Characteristic			
		Bass	Mid-Range	Freq.	Treble
Acoustic Guitar	Body/bass string Clarity Fret noise Pick noise	X	X X	2K/4K 4K	X
Electric Guitar	Fullness Bite Twang Brightness	X	X X	2K 4K	X
Bass Guitar	Bottom Pluck Twang (string) Bright	X	X X X	1K 2K 4K	
Strings	Scratchy Fullness	X			X
Horns	Fullness Shrillness	X	X	4K	
Woodwinds	Mellow Clarity Sharp Windy	X	X X	1K/2K 4K	X
Cymbals/Hi-Hat	Gong/Clanky Shimmer	X			X
Snare	Fat Attack Shhhh-Shhhh	X	X	4K	X
Tom-Toms & Floor Toms	Fullness Attack	X	X	4K	
Conga	Body Presence/Slap	X	X	4K	
Bass Drum	Bottom Slap	X	X	2K	
Harmonica	Fat Electric Acoustic	X	X X	2K 4K	
Piano & Organ	Bottom/Resonance Presence Honky-tonk piano	X	X X	2K/4K 2K	



A Guide To Basic PM-1000 Equalizer Utilization

INPUT LEVEL SWITCH & CHANNEL FADER

This precision, 11-step control alters both the pre-amplifier gain and the incoming signal level in order to obtain the best range of fader control, maximum amplifier headroom, and optimum signal-to-noise ratio for almost any input source. With the input fader set at its nominal position, the level switch should be adjusted to obtain the desired channel gain.

A suggested technique for initial console set-up, is to set all faders at "7". The individual input level switches are then adjusted to get the desired balance between channels, creating a rough mix. This program mix is trimmed with the input faders. Then the master faders are used to balance the four mixing buses and to achieve suitable overall program levels. (Refer to "VU Meters" in this section of the manual for further information concerning program levels.)

If an input fader must be set above No. 9 or below No. 5, the incoming signal may be too low or too high in level; the channel's input level switch may then be reset to bring the fader into the range between No. 5 and No. 9.

Input levels vary depending on the type of microphone, mic placement, and the nature of the musical material. Line inputs are also subject to a wide range of levels. Therefore, no single setting of an input level switch is ideal for a particular category of input sources. The setting which produces the correct mix level with the input fader set near No. 7 is the setting to use. For inputs of unknown level, it is a safe practice to begin with the input level switch at +4dBm (minimum sensitivity) and to rotate it clockwise until the desired sensitivity has been reached.

The settings listed in the chart are intended for use only as guides to finding the precisely correct setting of the level control.

One of the reasons that microphone levels vary is because they are proportional to the sound pressure level at the microphone diaphragm. A dynamic microphone rated at -50dB nominal output may indeed yield that output level when a vocalist holds the microphone

at arm's length, but the same microphone at a foot from a trumpet may average -20dB. Some instruments exhibit very high peak energy; this same microphone may peak above +1dB if placed inside a kick drum. Fortunately, Yamaha's sophisticated input level switch will provide excellent results in any of these situations.

The channel faders are linear-to-rotary attenuators. This design combines the advantages of easy to see and feel straight-line operation with the durability and noiseless performance of sealed, rotary units.

Once the input level switch is set for approximately the desired channel level, the input fader yields extremely fine gradations. Pulling a fader all the way down to the front of the console completely kills the output (to the program mix buses). The fader does not affect the cue or echo buses, although the input level switch does control these outputs.

CUE PUSHBUTTON

The cue button may be used during a performance to examine the input source of one or more channels without affecting the echo (foldback), line (tape) or monitor (house speaker) outputs of the console. Cue is heard only in the headphone output (when the cue buttons of the headphone mix section have been latched).

Cue is most useful when a portion of the program must be previewed. A tape recorder or turntable cue, for example, can be monitored in the headphones with the channel fader pulled all the way down. No audio will be fed to the program mixes until the fader is brought up, which can be done at the precise moment indicated by the headphone monitor.

Cue may also be used to locate a problem input. Consider the difficulty of killing exactly the problem channel during a live performance, say when a drummer hits a mic with his stick. Instead of pulling down the four or five drum faders, one at a time, until the bad mic is located, the drum channel cue buttons can be pressed sequentially while listening to the headphones. As soon as the bad mic is located, that channel is pulled down. Then, during a break, the mic can be repaired. This technique causes minimal disruption of the program.

TYPICAL INPUT LEVEL SETTINGS FOR VARIOUS SOURCES

INPUT SOURCE	INPUT LEVEL SETTING
LOW LEVEL MICROPHONES (dynamic & ribbon types)	-60, -50, -44
HIGH LEVEL MICROPHONES (condenser type)	-38, -32, -26
ELECTRIC INSTRUMENTS (direct, unamplified)	-38, -32, -26
LOW LEVEL LINES (hi-fi equipment)	-26, -20, -14
MEDIUM LEVEL LINES (hi-fi & some pro equipment)	-14, -8
HIGH LEVEL LINES (Professional equipment)	+4
ELECTRIC INSTRUMENTS (through direct box — pad & transformer)	-26, -20, -14, -8

MASTER & MONITOR MODULE

This module performs two distinctly separate functions. The linear Master Fader provides overall level control for the correspondingly numbered program mixing bus. The monitor portion of the module creates a monaural mixdown that is derived from the four Line A outputs (the program mixing buses). This mono mix is used to drive one of the house reinforcement amplifiers, a control room monitor amplifier, or (rarely) one channel of a tape machine.

MASTER FADER

The nominal fader setting, No. 7, produces the best combination of wide headroom and fine level control. For best results, when the initial program mix is set up, the master faders should be left at No. 7. Then, as the program mix is fully developed, or as channels are added and subtracted from the mix, the overall level of the four program buses can be trimmed with the master faders. If a master fader must be set above No. 9 or

below No. 5, the overall level on that program bus may be too high or too low; all input faders assigned to the bus may be brought up or down to correct the condition, thereby obtaining optimum results.

At the conclusion of a scene, a tune, or any segment of the program, overall fades are most easily performed by bringing down the master faders. This is much easier than bringing down all the input faders, and it does not disturb an established program mix. (It will be necessary to fade the echo send masters in order to kill the stage monitors; this fade may be done after the master fade to achieve an unusual effect.)

MONITOR MIX & MONITOR MASTER CONTROLS

The four monitor controls on each master module (labeled CH 1 through CH 4) derive a monaural blend from the console's master faders. Since the mix controls are duplicated in four adjacent modules, a 4 x 4 matrix is formed (not including the monitor master controls). Each vertical row of controls sets the mix and output level for one speaker feed (monitor output). Each horizontal row represents one program bus. The 4 x 4 matrix is an important new concept that allows far greater control of the sound than might be obtained by driving speaker amps directly from the bus outputs. Yamaha is the only console manufacturer to offer the matrix as standard equipment.

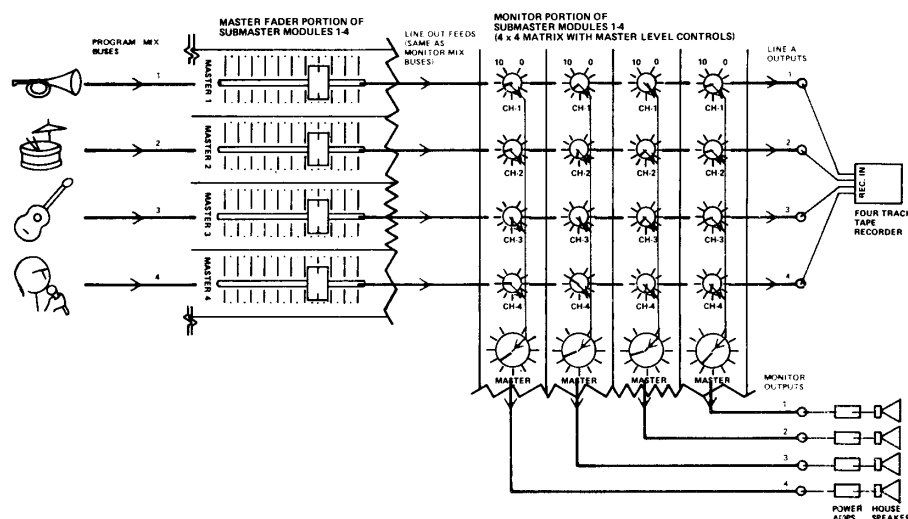
During a performance, stage monitor speakers and on-stage electric instrument amplifiers fill the near-stage audience with considerable amounts of instrumental sound. Therefore, reinforcement loudspeakers aimed at this portion of the audience require less instrumental and more vocal than those loudspeakers covering the middle and rear portions of the audience. The 4 x 4 matrix simplifies the job of obtaining this location-dependent speaker feed.

To take full advantage of the matrix, the four program mixing buses should be organized as group buses; vocal, rhythm, percussion, and lead instrument parts can be assigned to respective buses. The master faders then function as group masters. Within each master/monitor module, the mix controls set the balance between groups, and the monitor master control sets the overall mix level. In this way, each monitor output (speaker feed) can carry a distinctive blend of the four groups.

In practice, the 4 x 4 matrix is easy to use. For example, assume program bus No. 4 is the vocal group and there is too much vocal in the mix throughout the audience; pulling down master fader No. 4 simultaneously reduces the vocal content in all four speaker outputs. On the other hand, suppose that only the near-stage speakers contain too much vocal; the CH 4 monitor mix controls in the modules driving these speakers can be trimmed without affecting the vocal balance in the rear of the audience. As an added illustration of the matrix flexibility, assume the vocal balance which has just been corrected in the near-stage speakers has left those speakers with inadequate overall level; bringing up the monitor master controls for the same modules will increase the volume in the speakers without changing the balance of the vocal and instrumental groups — and without affecting the rear of the audience at all.

At the conclusion of a segment of the program, the house speakers may be faded in one of two ways: the master faders can be brought down or the monitor masters can be brought down. If the console's line output is driving a tape recording, the house speaker volume should be brought down with the monitor masters so that the recording continues at full level.

BLOCK DIAGRAM OF 4 x 4 MONITOR MATRIX AS USED TO ACHIEVE INDEPENDENT SPEAKER FEEDS



HEADPHONE & ECHO MODULE

ECHO SEND MASTER CONTROLS

These controls determine the overall level of the echo (stage monitor/foldback) outputs. The suggested technique for obtaining a stage monitor mix is to set the two Send Master controls at No. 7, and to then adjust the input channel Echo mix controls for the desired sound. The echo VU meters visually display the level of the echo outputs.

If one or both echo outputs are utilized to drive professional echo, reverb or tape delay units, the processed audio is normally returned to the program via channel inputs...No. 15 and No. 16 (or No. 31 and No. 32) are suggested for convenience.

By returning the audio via input channels, the echo (or reverb) may be equalized and filtered. Its level is controlled by the input channel Fader(s), and it may be assigned to any combination of the four program buses (using the bus assign switches). When a reverb spring or chamber is driven by the Echo 1 or Echo 2 output and returned through an input channel, set the channel's corresponding Echo mix control (E1 or E2) at "0" to prevent howling. However, where the echo output drives a tape delay, and the delay is returned through an input channel, advancing the channel's corresponding Echo mix control will create a slap-echo effect. A tape delay and a reverb may be "cascaded" for even more effects, as illustrated.

the ratio of dry program with delayed audio. If the reverb device or tape delay unit has output level controls, these may be adjusted to obtain the desired ratio. (The Master Faders control the echo and the "straight-through" audio, but not the ratio between them.) Using Sub In for echo return is advantageous because it frees input channels for use with microphones or other audio sources, but is limited in that there is no opportunity to equalize the return, nor is it possible to assign one return to more than one program mixing bus.

HEADPHONE CONTROLS

The headphone controls consist of two rows of push-buttons and a master level control. Each row of buttons selects the audio source(s) for one side of the stereo headphones. Selections include program mix buses 1 through 4 and the cue bus. When more than one push-button (per row) is latched, that side of the headphone output will consist of an equal blend of the selected sources. The headphone level control simultaneously adjusts both the left and right sides of the output.

Headphones are commonly used to monitor a mix prior to speaker feeding, especially to examine the cue bus. The headphone output is brought to two identical, stereo jacks (TRS phone jacks) in the front of the console, allowing more than one person to listen through phones. Either phone output may be utilized to drive a hi-fi tape recorder; since the output is not affected by the monitor master controls, the master faders, or the echo send masters, the tape can record the entire program at full level. The headphone output may also be used to drive power amplifiers for more monitoring flexibility. (See page four-13.)

TALKBACK & OSCILLATOR MODULE

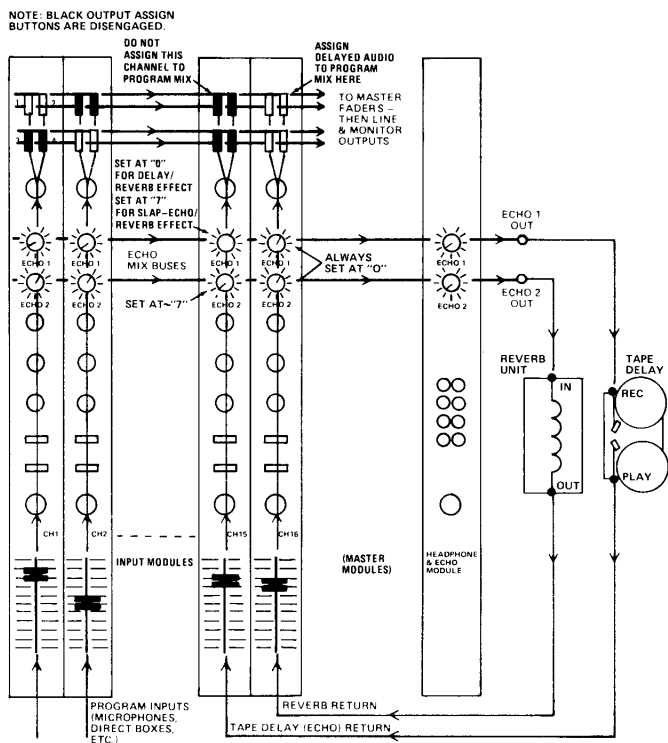
OSCILLATOR

The oscillator may be used to trace signal paths, calibrate levels, and to lay down reference tones on tape recordings. The oscillator output goes to two points, a transformer-isolated output jack on the rear panel and a set of bus assign switches in the TB & OSC module. The oscillator output level is fixed; it produces +4dBm at the TB out connector, and it produces a bus level that yields a +4dBm line output when the master fader is set at index No. 7.

Switching on the oscillator to 1kHz or 7kHz instantly applies a continuous sine wave to the TB Out jack. At the same time, this sine wave appears at the module's bus assign switches. Latching the switches applies the tone to the program mixing buses. * (Even when the switches are not latched, some leakage—crosstalk into the program—may occur. To prevent leakage, the oscillator must be shut off when not in use.)

For tape machine alignment, the 1kHz tone establishes the 0 VU reference level and the 7kHz tone is helpful for head azimuth adjustment. Commonly, the 1kHz tone is recorded for about 10 seconds, then the 7kHz tone for about 15 seconds. Both tones are recorded

*16-channel consoles manufactured after February, 1976 (Serial No. 1542 and 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.



Under certain circumstances, delayed audio from an echo or reverb unit may be returned to the Sub Input, although there is no level control within the signal path of the PM-1000 that can be used to balance

with the console's Master VU indication at 0 (+4dBm), and with the tape machine's record level adjusted for the proper nominal indication on the machine's meters. Tape boxes or reels should be clearly labeled to identify the reference tones (the frequency, level, and location on the tape).

The oscillator may be used for troubleshooting by connecting a jumper cable from TB Out to any channel input — or into auxiliary equipment. The master modules may be tested by latching the correspondingly numbered bus assign switches in the TB & OSC module.

NOTE: For precise level calibration, the oscillator output is +4dBm (1.23V) when terminated by a 600-ohm load. If terminated by a high impedance, the voltage may rise 1/2dB or more; to maintain correct calibration, a 604-ohm resistor should be connected from the high to the low side of the high impedance device's input.

TALKBACK

The console operator uses talkback to communicate with remote areas (intercom), to identify tapes (slating), or to make announcements through the house loudspeakers.

A low impedance microphone (or a high impedance mic with a matching transformer) may be connected to the **TB In** jack on the module's front panel. When the talkback button is pressed, the mic's preamplified audio appears at the **TB Out** connector and at the module's bus assign switches. Latching the switches applies audio to the program mixing buses, which may feed a tape machine, headphones, and/or the main reinforcement system. Unlatching the bus assign switches permits the talkback mic to be used for intercom with backstage areas, the lighting booth, dressing rooms, an assistant on the floor — or anywhere. The rear panel **TB Out** connector is designed to drive an external power amplifier or a pair of high impedance headphones (600 ohms or greater). 8-ohm headphones will cause some loss of high frequencies, but they are fine for voice communication.

For two-way communication, a pair of low impedance microphones can be connected to the TB mic input (with a Y-adaptor cable). Two high impedance headphones, one for the console operator and one for the remote station, would then be driven from the talkback output (also with a Y-adaptor). Both stations can talk and listen simultaneously when the talkback button is depressed.

THE METER PANEL

VU METERS

The two smaller VU meters display the Echo Out level. The four large VU meters display either **Line Out** level (Master mode) or **Monitor Out** level (Monitor mode), depending on the setting of a slide switch beneath each meter. Instead of switching the meters, 24 and 32 channel consoles house eight large VU meters; four for **Line Out** level and four for **Monitor Out** level. All meters are calibrated so that a 0 VU indication corresponds to a +4dBm console output (1.23Vrms into a 600-ohm load).

During initial console set up the large meters should be switched to master mode. The input faders are adjusted to obtain the desired program mix, and the master faders are then adjusted to obtain metered levels averaging from -10 VU to 0 VU, with occasional peaks reaching +3 VU. The echo mix and echo master controls are similarly adjusted for echo meter indications of from -10 to 0 VU. The large meters are switched to monitor mode, and the monitor mix (speaker mix) is then prepared; first the CH 1 to CH 4 mix controls are adjusted and then the monitor master control is adjusted to obtain metered levels of from -10 to 0 VU.

The choice of 0 VU (+4dBm) is not arbitrary. Rather, for the PM-1000 system, it represents the point of the highest signal-to-noise ratio that allows adequate headroom for undistorted transient peaks and maintains a sufficient output level to drive other equipment. To understand why 0 VU (+4dBm) is the nominal output level, it should be explained that VU meters are, by definition, averaging devices; they indicate approximate rms voltages that correspond to power output into a 600-ohm load. Often there are instantaneous peaks, transients that can exceed the average level by up to 20dB. The peaks are too brief to deflect the PM-1000 VU meter, but they can cause audible distortion by driving the amplifiers into clipping. Thus, a VU meter with peak deflections of 0 to +3 VU (+4 to +7dBm output) may represent actual peak output levels of up to +24dBm.

About 10dB of room (between -10 and 0 VU) is needed to establish the right level for different types of program material. This is because some sources have fairly constant energy levels, while others exhibit high peak-to-average energy ratios. For instance, a smoothly bowed violin may be mixed with average deflections of 0 VU because there is little likelihood of excessive transient peaks which don't show on the meter. On the other hand, a kick drum should probably be mixed closer to an average deflection of -10 VU with visible peaks reaching 0 VU; this will prevent sharp "attack" transients, which may exceed the 0 VU visible level by 20dB, from overdriving the console's amplifiers.

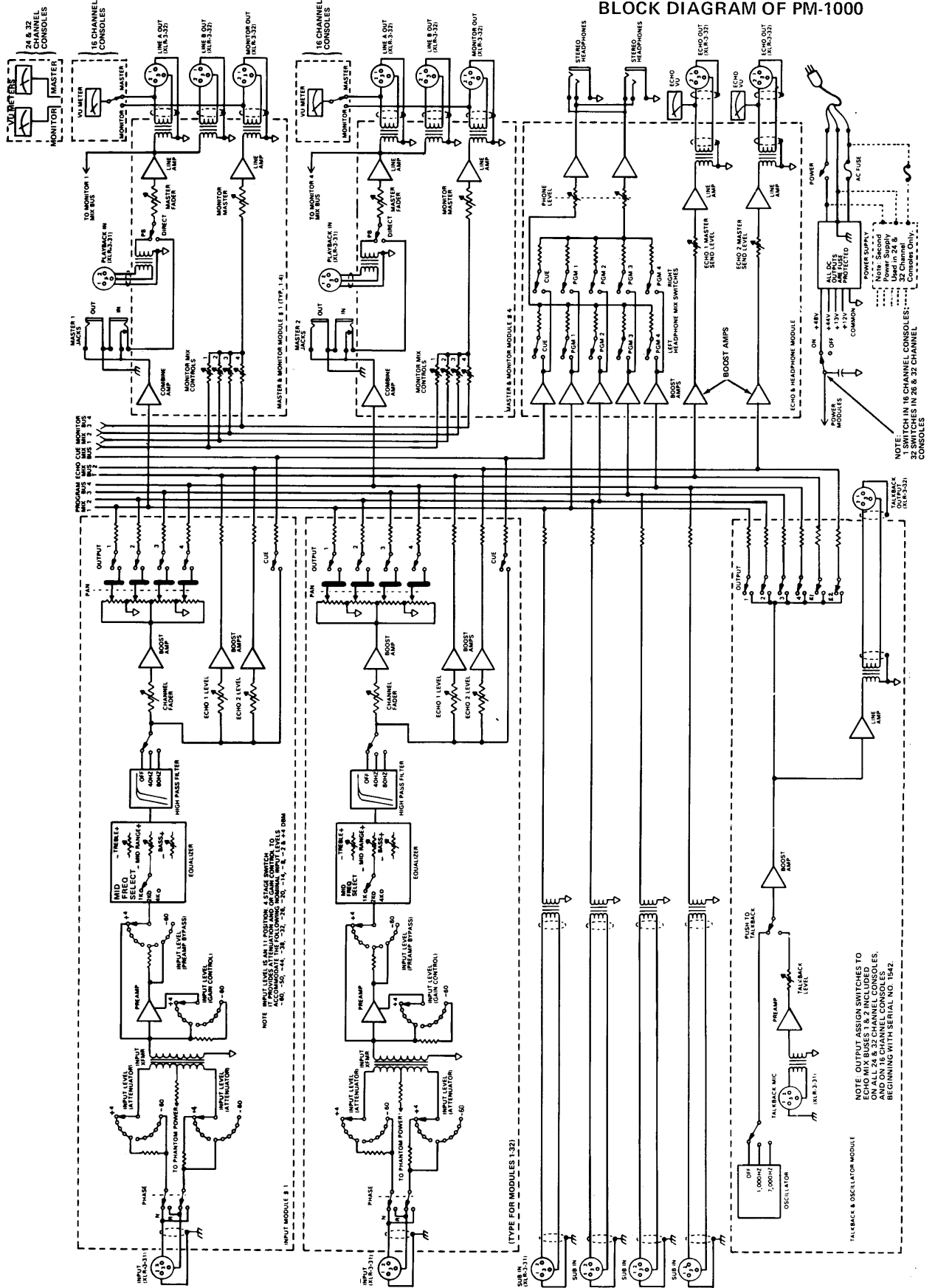
POWER SWITCH

Turning on the console can generate powerful low-frequency transients. In order to protect loudspeakers from damage, all power amplifiers should remain off until the console has been on for a few seconds. Switching on the console with the faders down does not prevent turn on transients.

As a further precaution, the console's faders should be down when auxiliary input electronics are switched on or off. The console power switch may be shut off at any time, and it is a safe practice to leave the switch off whenever cables are being connected or disconnected.

SECTION SIX

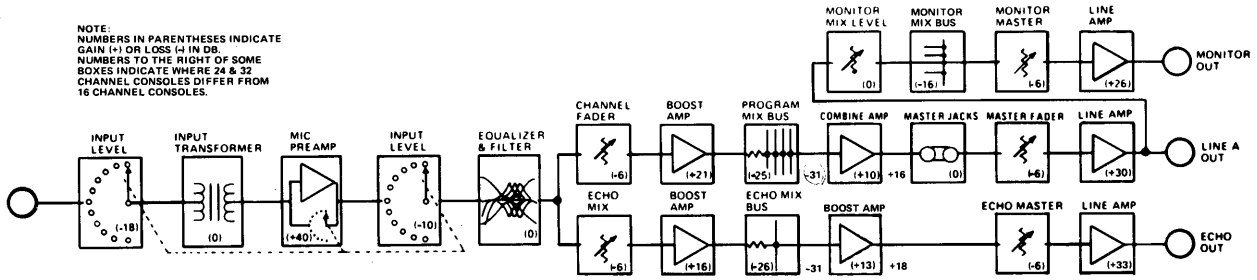
BLOCK & SCHEMATIC DIAGRAMS



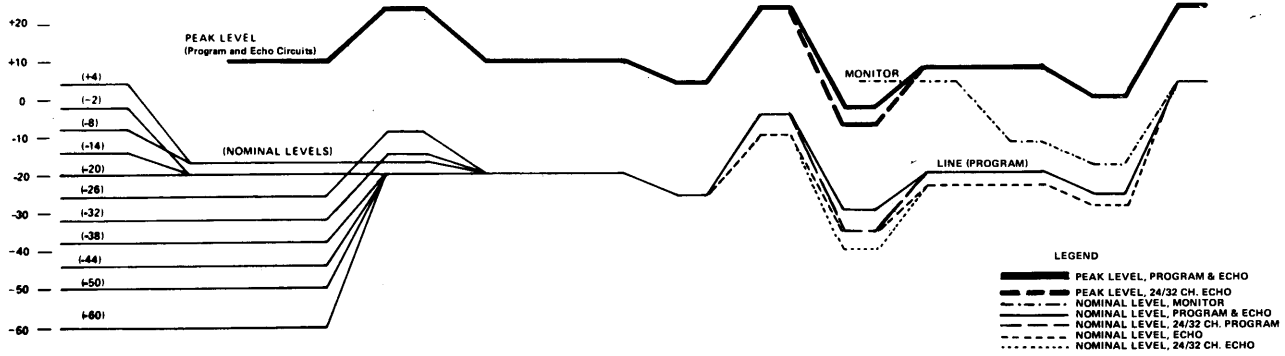
SIX2

PM-1000 SYSTEM HEADROOM

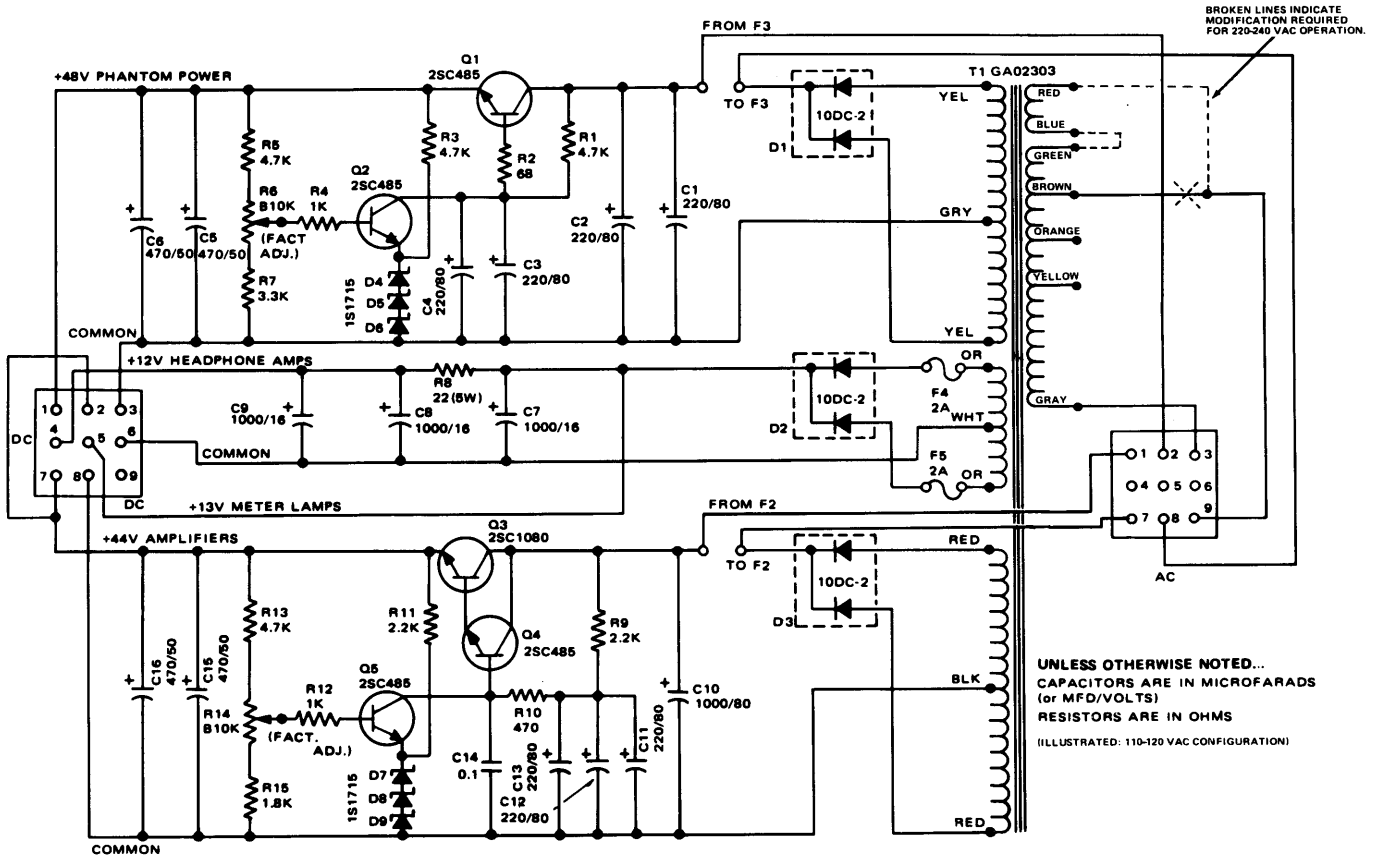
NOTE: NUMBERS IN PARENTHESES INDICATE GAIN (+) OR LOSS (-) IN DB. NUMBERS TO THE RIGHT OF SOME BOXES INDICATE WHERE 24 & 32 CHANNEL CONSOLES DIFFER FROM 16 CHANNEL CONSOLES.



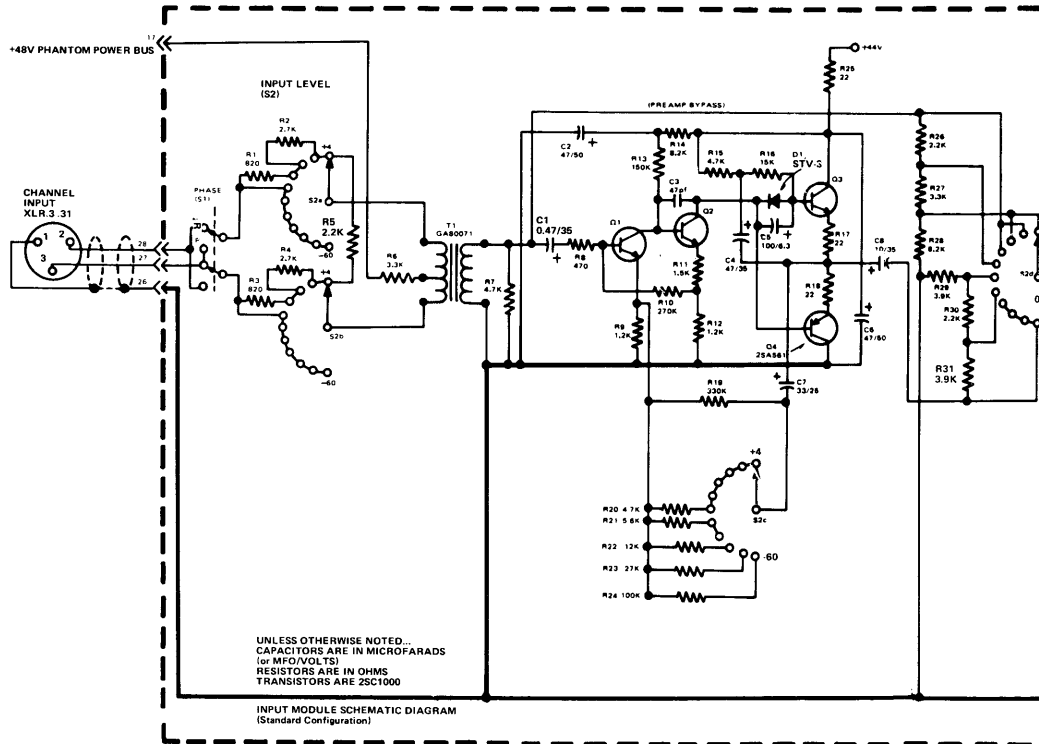
ACTUAL SIGNAL LEVEL (dBm) (INPUT LEVEL SETTING)



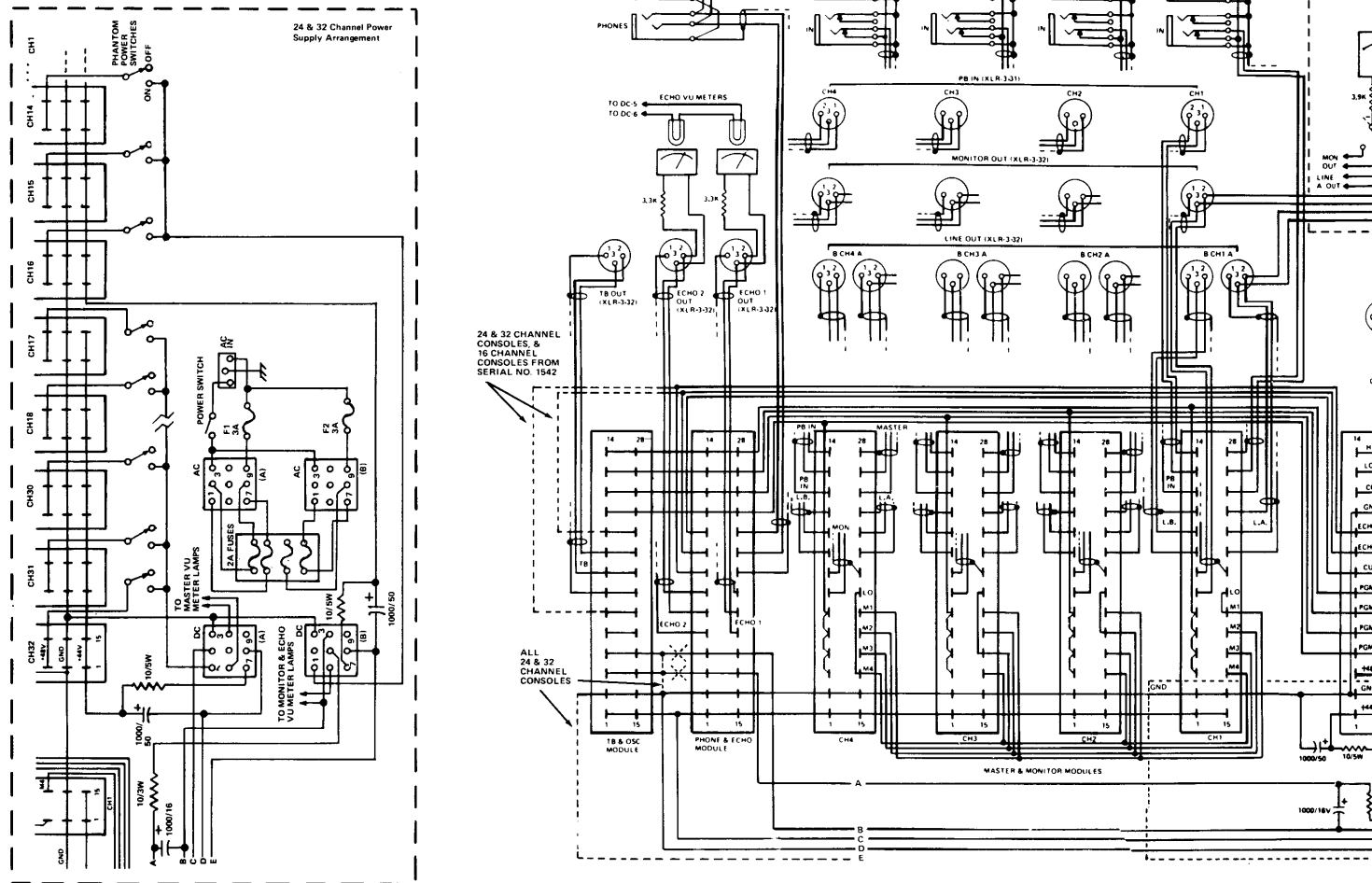
PM-1000 POWER SUPPLY SCHEMATIC

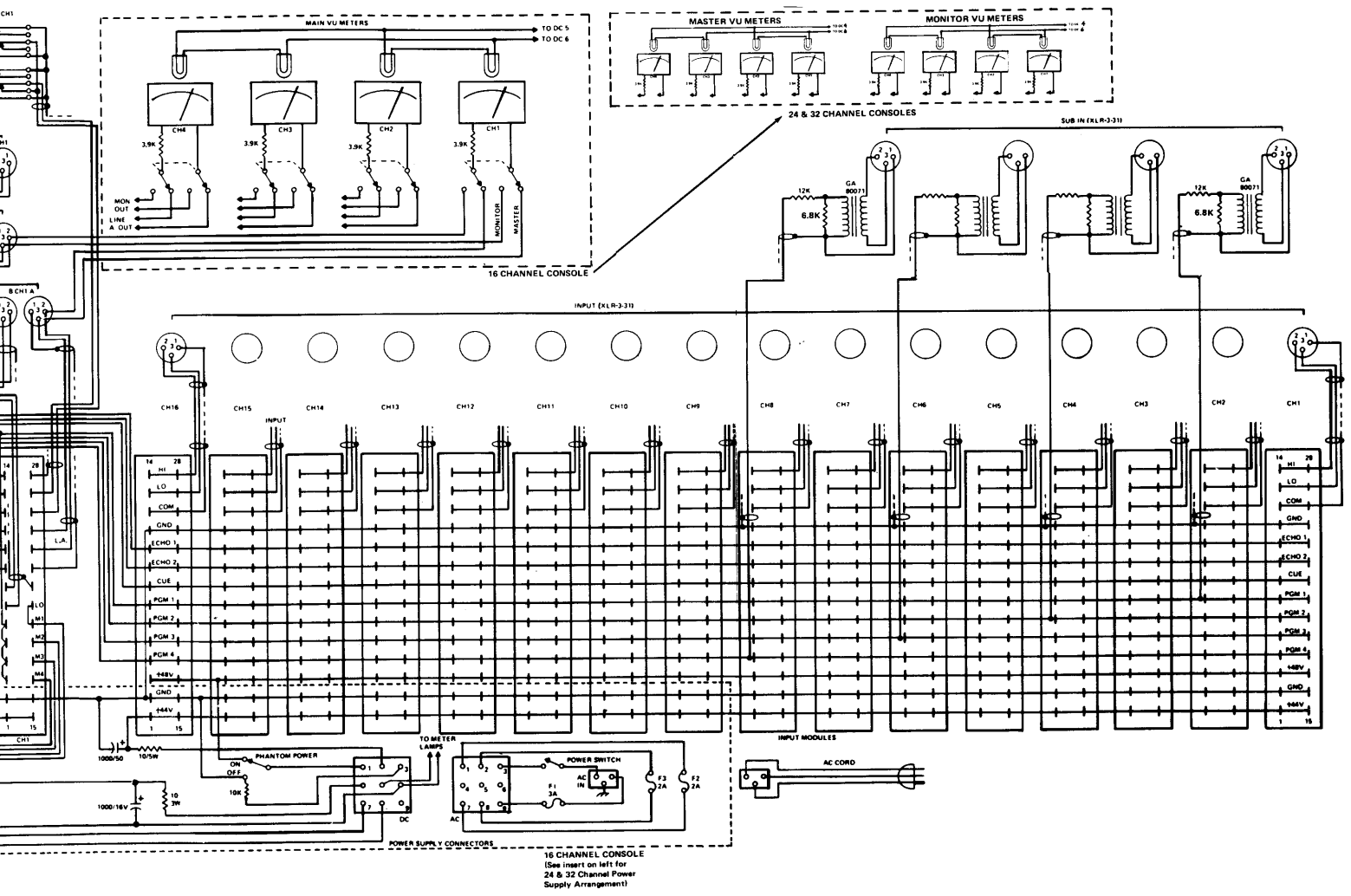
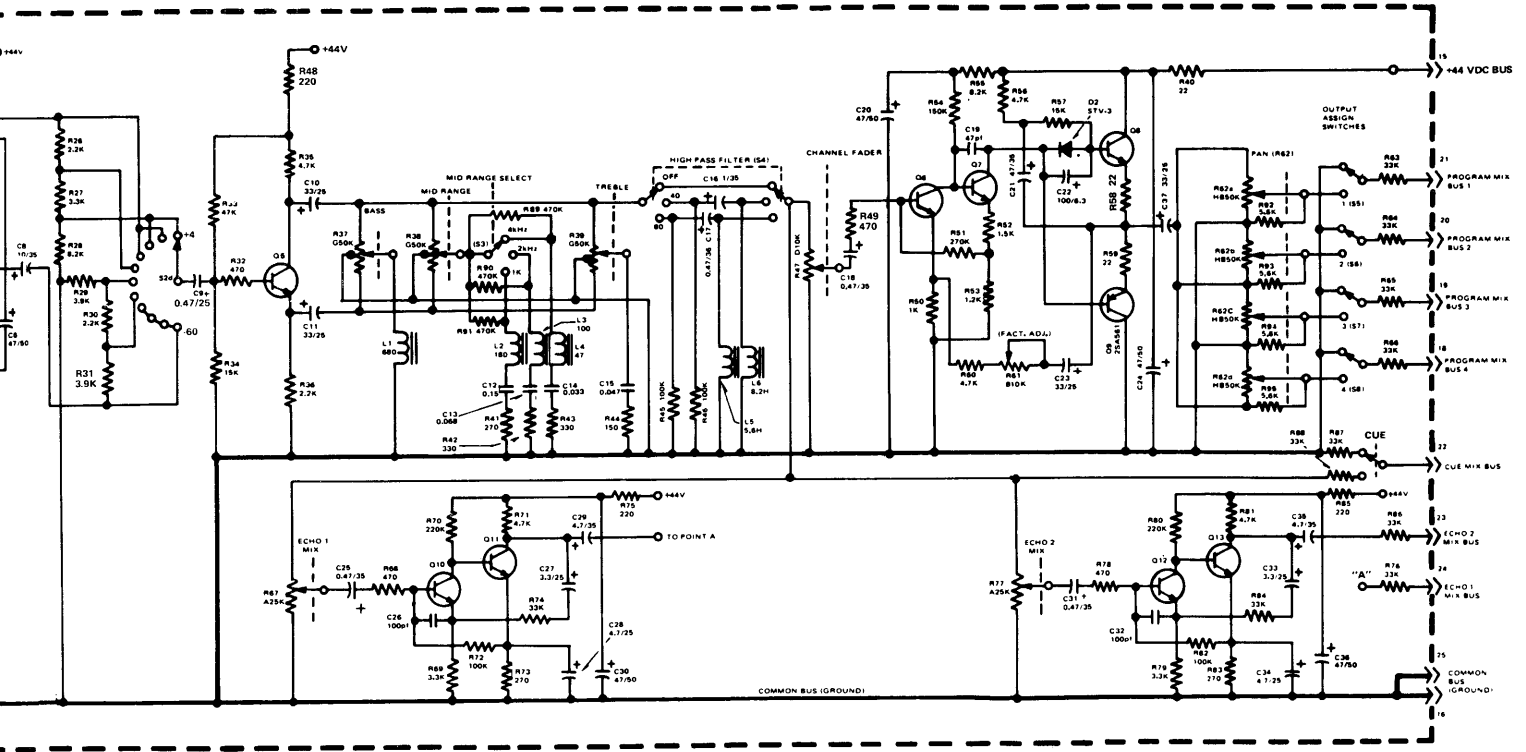


PM-1000 INPUT MODULE SCHEMATIC



PM-1000 FRAME SCHEMATIC

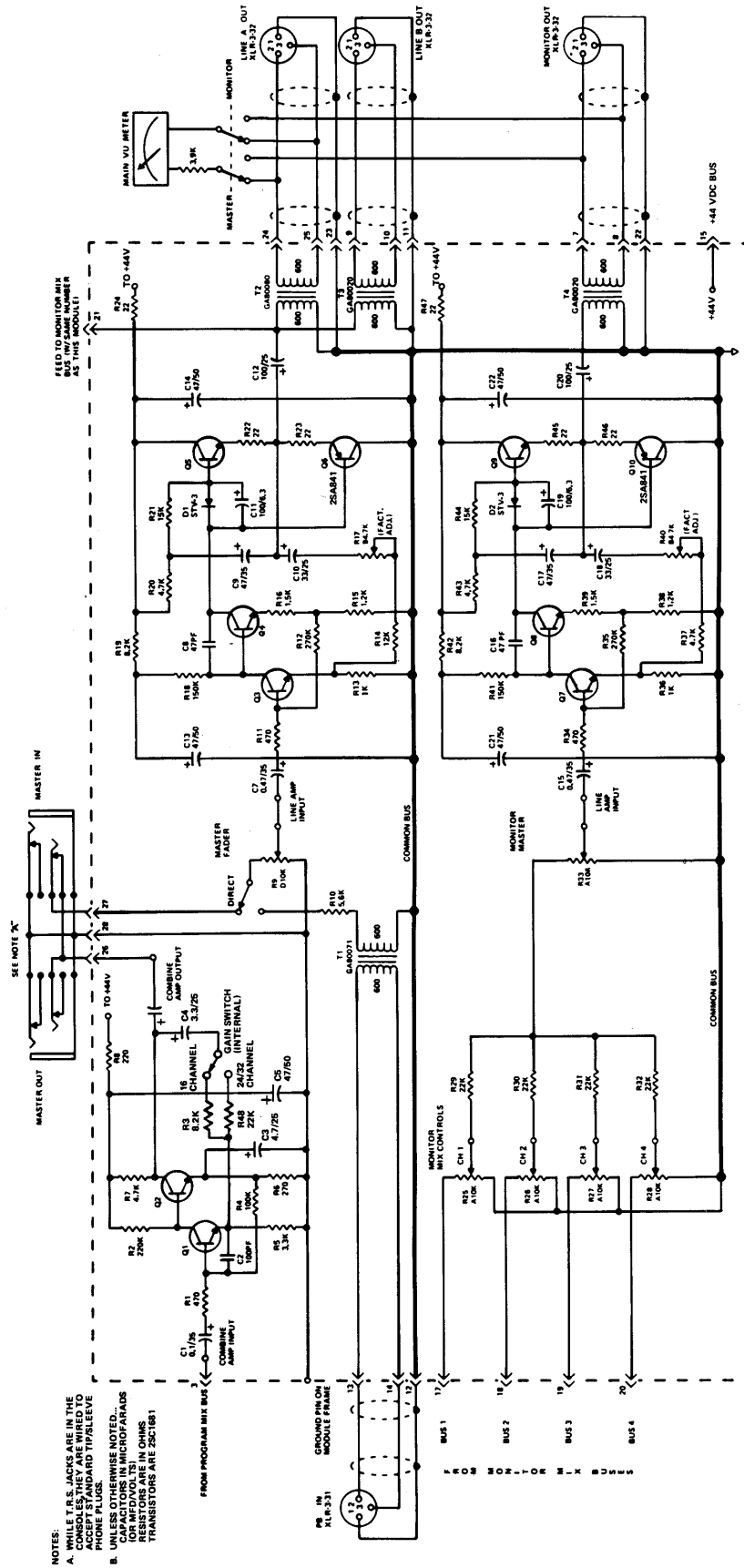




16 CHANNEL CONSOLE
(See insert on left for
24 & 32 Channel Power
Supply Arrangement)

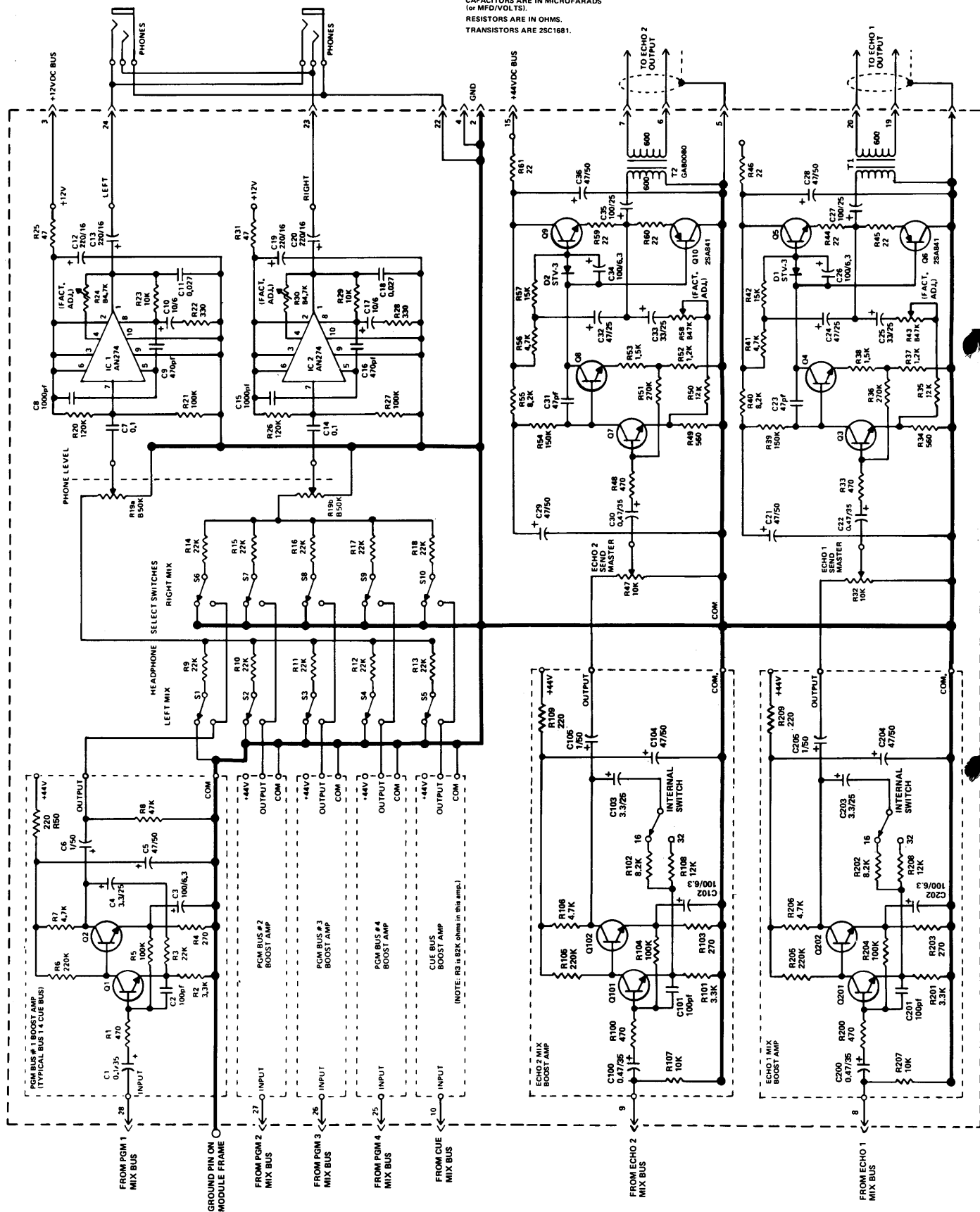
SIX4

MASTER & MONITOR MODULE

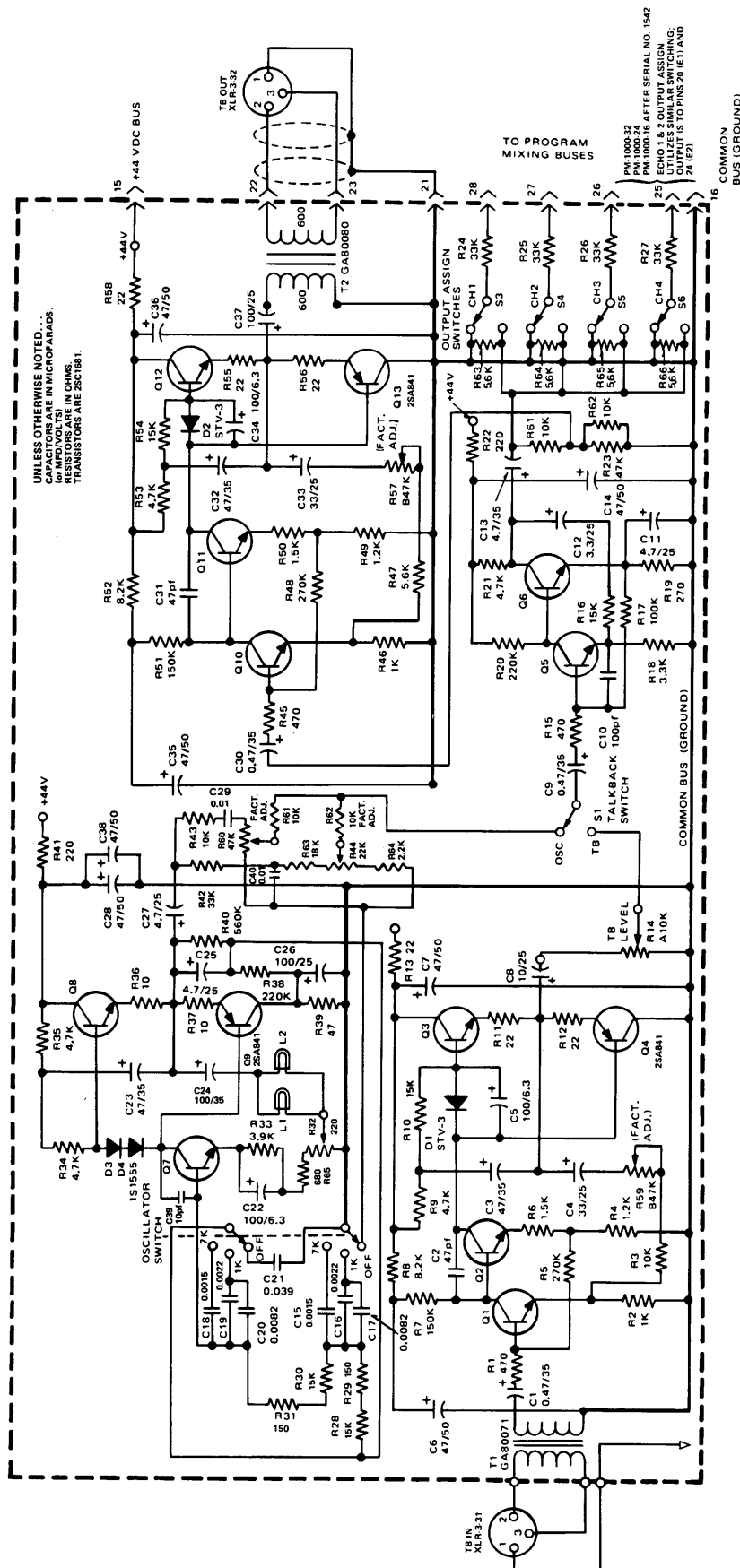


PM-1000 HEADPHONE AND ECHO MODULE

UNLESS OTHERWISE NOTED,
CAPACITORS ARE IN MICROFARADS
(or MFD/VOLTS).
RESISTORS ARE IN OHMS.
TRANSISTORS ARE 2SC1681.



TALKBACK & OSCILLATOR MODULE



SECTION SEVEN 1

MAINTENANCE, REPAIR & WARRANTY SERVICE

GENERAL MAINTENANCE NOTES

SUGGESTED SPARE PARTS

The PM-1000 is designed and built to exacting standards, and it should continue to operate properly without special maintenance. Still, accidents and failures may occur, and spare modules can save the day in a matter of minutes. A spare input module, master module, headphone and echo module, and power supply module should cover any eventuality.

PANEL AND CABINET CLEANING

The black anodized panels should be cleaned with a damp sponge. Stubborn soil can be removed with a mild detergent solution, such as dishwashing detergent. Strong detergents and chemical solvents may damage the plastic fittings.

The rosewood cabinet will retain its beautiful finish with very little care. When it looks dull or soiled, apply any liquid or paste furniture polish and buff with a soft cloth; aerosols should be avoided because the solvents may attack adjacent portions of the console.

FUSE REPLACEMENT

The rear panel is fitted with fuse holders for the AC line (and on 16-channel consoles for the DC side of the power supply). Fuses should be replaced with identical type and value fuses. If a fuse continues to fail, do not install a higher value fuse — find the cause of the failures and correct the problem. In the event the problem cannot be located, contact your Yamaha dealer or the Yamaha International Corporation service department (phone number listed at the end of this section).

REPAIRS

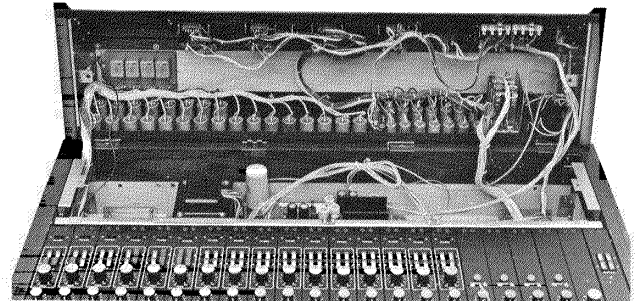
The modular construction of the PM-1000 enables almost any problem to be corrected within minutes. Defective or damaged modules should be returned to Yamaha for repair or replacement. Unauthorized service or modification of any part of the console may void the warranty.

ACCESS TO THE CONSOLE INTERIOR

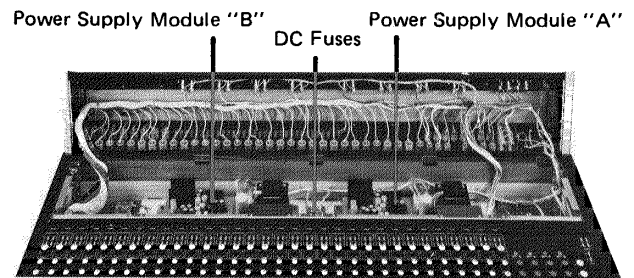
WARNING: There are no user-serviceable parts inside the PM-1000. Only qualified service personnel should attempt to open the meter panel, remove a module, or gain access to the console interior for any purpose. Lethal voltages are present inside the console, and the AC line cord should be disconnected prior to opening it.

The first step is to disconnect the AC line cord. If one of the front panel modules is to be removed, the meter pod need not be opened, but the armrest must first be lifted from the cabinet. Grasp both ends of the rest, holding it with fingers close to the front edge of the console. Lift the armrest straight up to disengage the coupling pins at each end, and set it aside. The binding screws and handles of the modules are now visible at the front of the console.

To open the meter pod, first remove the two screws from the top of the pod. The meter panel may then be tilted backward, exposing the power supply module(s). Exercise care not to strain the hinges on the meter pod, and avoid straining any cables connected to the rear panel.



16 CHANNEL CONSOLE

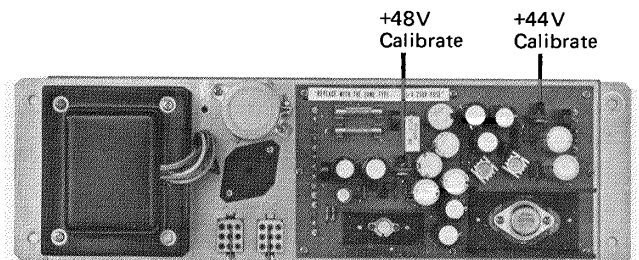


24 & 32 CHANNEL CONSOLES

POWER SUPPLY REMOVAL, REPLACEMENT & 220-240V AC CONVERSION

After opening the meter pod, the power supply will be visible on the bottom of the console. Grasp the locking tabs on either side of the multi-pin connectors on the top of the supply, and unplug both connectors. Then remove the four screws securing the perimeter of the power supply module to the bottom of the cabinet. Place a soft cloth on the front panel of the console, and after carefully removing the module place it onto the cloth; a ground wire is still connected to the module. Unscrew the ground wire and retain the hardware, observing the hole from which the wire is removed for future reference. This completes the removal of the supply module.

To replace the module, first reconnect the ground wire. Then lift the module into place and secure it to the cabinet with the four screws. Observe the two multi-pin plugs removed from the supply; one has fewer pins actually used than the other. Observe the power supply chassis connectors, and again one connector will utilize fewer pins than the other. **BE SURE TO INSTALL**



AC Connector DC Connector

SEVEN 2

THE TWO CABLES IN THE CHASSIS CONNECTORS WITH MATCHING PIN USAGE. A key at one end of the connector assures proper orientation of the connector, but it does not prevent installation in the wrong socket.

POWER SUPPLY CONVERSION TO 220-240V AC

The PM-1000, as delivered in the U.S. and Canada, is wired for operation with 110-120V AC power mains. The supply may be re-wired to operate with 220-240 mains. This modification is relatively simple, although it should be performed by qualified service personnel only. The procedure is outlined below. See the power supply schematic diagram on page SIX-2 for a circuit reference.

1. Remove the power supply module from the console, as detailed in the preceding subsection of this manual.
2. Invert the module so that the underside of the chassis becomes accessible, and locate the bundle of leads emerging from the bottom of the power transformer; cut the nylon tie-wrap that holds these leads together.
3. Cut the insulating caps from the ends of the blue, green and red leads; then strip ½-inch of insulation from each of these leads.
4. Twist the blue and green leads together, and fasten with a wire nut or with solder (use electrical tape or heat-shrinkable tubing to insulate the solder joint).
5. Locate the brown lead that runs from the transformer to pin 9 of the AC connector; cut this lead approximately half way between the connector and transformer.
6. Cap the portion of the brown lead that emerges from the transformer with a wire nut or electrical tape.
7. Strip ½-inch of insulation from the portion of the brown lead that comes from pin 9 of the AC connector, and twist this lead together with the red lead from the transformer. (Fasten the leads with a wire nut (or solder and insulate).
8. Re-tie the transformer leads, dressing them to the side of the transformer. This will prevent them from becoming pinched when the power supply module is placed in the console.
9. Return the power supply module to the console, as described in the preceding subsection.

(NOTE: Repeat steps 1-9 for each supply module in 24 and 32 channel consoles.)

10. Change the rear-panel AC fuse(s) from 3-ampere to 1½ ampere load rating. Also, change the AC power cord (or connector) to one suitable for use with 220-240V mains. This completes the modification procedure.

We strongly urge that a prominent label be affixed to the console rear panel, indicating that the power supply is now wired for 220-240V AC mains.

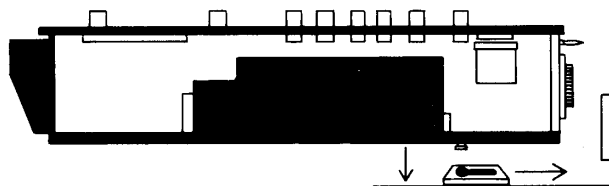
INPUT, MASTER, HEADPHONE AND TALKBACK MODULE REMOVAL AND REPLACEMENT

After removing the armrest, locate the binding screw

at the front edge of the module. Remove the screw from the module and set it aside. Grasp the handle at the front edge of the module, and pull the module toward the front edge of the console about 1" (2.5cm), until it stops. Gently lift the module straight up, grasping the front and rear edges; be particularly careful not to crush any wires between the module being removed and an adjacent module. Never try to lift a module up without first pulling it forward.

A module must be installed in the correct portion of the console frame; while it is possible to fit an input module in the space normally occupied by a master and monitor module, the console will not function under these conditions and damage may result. When in doubt, refer to the front panel illustration in section one of this manual.

To replace a module, align it over the correct space in the frame, with the screw head in the bottom of the module frame directly over the widest part of the key-hole-locking bracket on the bottom of the cabinet. Gently push the module down until the screw head slips through the keyhole. Then push the module forward, making certain the ground pin and audio connector on the rear of the module mate with the console-mounted connectors. When the module is pressed fully forward, thread the long binding screw through the front edge and into the bottom of the cabinet. Secure the screw firmly, but do not overtighten, as this will strip the threads.



MODULE REPLACEMENT

PERIODICALS OF INTEREST TO SOUNDMEN & ENGINEERS

dB Magazine
1120 Old County Road
Plainview, N. Y. 11803

Journal of the Audio Engineering Society
60 E. 42nd Street
New York, N. Y. 10017

Modern Recording Magazine
Recording Institute of America, Inc.
15 Columbus Circle
New York, N. Y. 10023

Recording engineer/producer
P. O. Box 2449
Hollywood, California 90028

WARRANTY AND FACTORY ASSISTANCE

The Warranty Registration Card delivered with the PM-1000 should be completed and returned to the factory at the time of purchase. Retain the Warranty ID card for your records. In the event warranty service

is required, please contact your Yamaha dealer. If you need assistance in locating a Yamaha dealer, call Yamaha Combo Product Service at (714) 522-9353. 24-hour emergency problem solving assistance is available at this same phone number.

YAMAHA PM-1000 PRODUCT WARRANTY

LIMITED ONE YEAR WARRANTY

The Yamaha PM-1000 mixing console is designed and manufactured to provide a high level of performance. Yamaha International Corp. believes in the craftsmanship that goes into Yamaha products. Please read this Warranty to familiarize yourself with the Warranty terms.

Yamaha desires that every new Yamaha product owner understand this Warranty; therefore, should you have any questions, ask your authorized dealer or write directly to Yamaha.

CONDITIONS OF WARRANTY

If, during the one (1) year period from the date of original purchase your product is found by Yamaha to have a defect in material or workmanship, Yamaha and/or its authorized service center will repair defects without charge for material or labor. In the event Yamaha determines that the defect cannot be repaired, then Yamaha will replace the defective product with an identical Product or one which is reasonably equivalent.

WHERE PERMITTED BY LAW, YAMAHA INTERNATIONAL CORPORATION'S LIABILITY SHALL BE LIMITED TO THAT SET FORTH IN THIS WARRANTY AND YAMAHA SHALL NOT BE LIABLE FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES, INCLUDING INJURY TO PERSONS OR PROPERTY.

YAMAHA INTERNATIONAL CORPORATION MAKES NO OTHER WARRANTY OF ANY KIND EXPRESSED OR IMPLIED. ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE WHICH EXCEED THE OBLIGATIONS AND TIME LIMITS STATED IN THIS WARRANTY ARE HEREBY DISCLAIMED BY YAMAHA INTERNATIONAL CORPORATION AND EXCLUDED FROM THIS WARRANTY.

OWNER'S RESPONSIBILITY

In order for Yamaha to provide proper warranty service, it is necessary that the purchaser of the product assume certain responsibilities.

1. Send to Yamaha the Warranty Registration Card within ten (10) days of purchase.
2. Be prepared to present the Warranty Card or proof of purchase to avoid undue difficulties in determining eligibility for warranty protection.
3. Notify your Authorized Yamaha Dealer of any defects within ten (10) days of discovery.
4. Return your Yamaha product to your Authorized Dealer in order that he may inspect and approve warranty service which shall be completed within a reasonable period of time.

WARRANTY SHALL NOT EXTEND TO:

1. Products subjected to abnormal strain, neglect, abuse, modification or accidental damage.
2. The result of normal wear.
3. Products whose trademark, name, or identification number has been changed or removed.
4. Products purchased from anyone other than an Authorized Yamaha Dealer.

This warranty is limited to the 50 states of the USA. If you plan to use the PM-1000 outside the USA, please contact Yamaha Combo Product Service at (714) 522-9353.

YAMAHA INTERNATIONAL CORPORATION
Box 6600
Buena Park, California 90620

PM-1000-32 Thirty-Two Input Channel Console

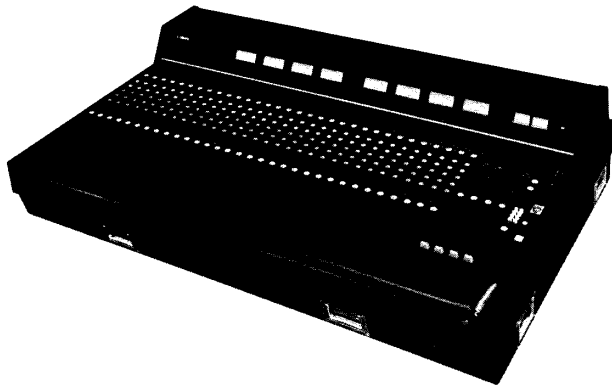
PM-1000-24 Twenty-Four Input Channel Console

GENERAL DESCRIPTION

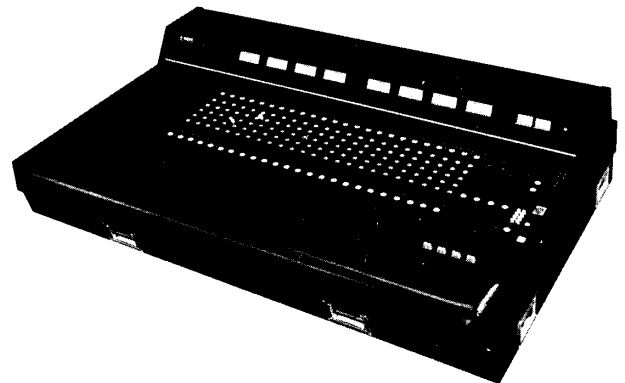
The original 16-input PM-1000 has become extremely popular; in fact, many users who need more than 16 input channels have purchased two or more PM-1000's and "chained" them together. In response to the demand for additional input capability, Yamaha has created the PM-1000-32 and the PM-1000-24, in essence "stretch" versions of the PM-1000-16. Built in the same tradition of high performance and rugged reliability, the larger consoles are every bit as roadable as the PM-1000-16.

The PM-1000-32 main frame (MF 10-32) is approximately twice as wide as the 16-channel main frame (MF 10-16), and is used for both the thirty-two and the twenty-four channel configurations. Whether you have the PM-1000-32, or the PM-1000-24, the frame contains all the connectors, transformers, wiring, and power supply capability necessary for 32-channel operation. This makes it a simple and inexpensive matter to expand the twenty-four channel console to thirty-two channels; just remove the 8 blank modules on the PM-1000-24, and drop in 8 input modules.

The width of the larger main frame makes it possible to provide separate VU meters for the **Master** (Line A Out) and **Monitor** outputs; therefore, the **Master/Monitor** switches have been deleted from the meter panel. The additional space on the rear panel has been utilized to provide 16 more XLR-type input connectors, and individual Phantom Power ON/OFF switches for each input channel. The individual switches allow some inputs to be connected to unbalanced equipment (without using isolation transformers) while other inputs are used simultaneously to power condenser microphones. Another difference between the 16-channel console and the 24 or 32 channel consoles is the power supply arrangement; whereas the 16-channel console has one power supply module. Two identical power supplies share the load in the larger consoles. Aside from these relatively minor differences, the PM-1000-24 and PM-1000-32 are virtually identical in features and performance to the original PM-1000-16. For installation, operation, and maintenance instructions, refer to sections one through seven of this manual.



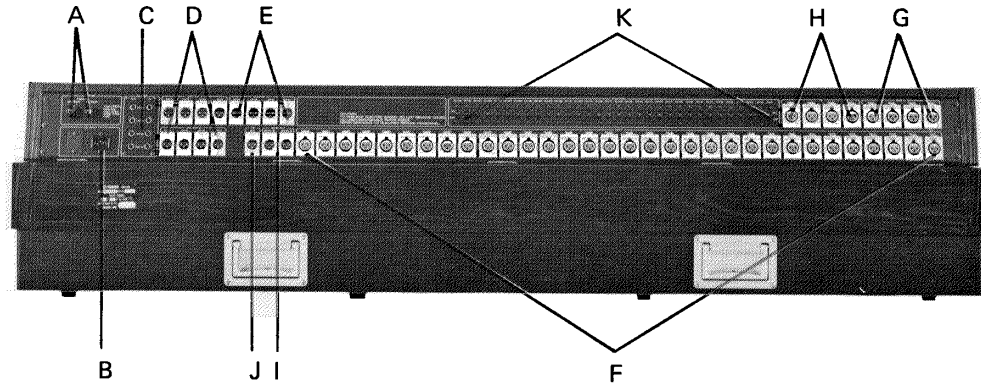
PM-1000-32



PM-1000-24

EIGHT²

REAR PANEL



- A. Power Fuses
- B. Power Cord Connector
- C. Master Out & Master In Jacks
- D. Line-A-Out & Line-B-Out Jacks (Program Mix Out)
- E. Monitor Out (Speaker Feed)
- F. Channel Input Jacks
- G. Submixer Input Jacks
- H. Playback Input Jacks
- I. Echo Output (Foldback Out) Jacks
- J. Talkback Output Jack
- K. Phantom Power Switches

SPECIFICATIONS

Most of the specifications for the PM-1000-24 and the PM-1000-32 are identical to those of the PM-1000-16, as listed in section two of this manual. Input and output characteristics, for example, are identical. Those specifications which do not correspond to the 16-channel data are listed below.

Hum and Noise* (20Hz-20kHz)	-121dBm Equivalent Input Noise. -66dB (70dB S/N) Line Out A & B; Master Fader at nominal level and all Input Faders down. -57dB (61dB S/N) Line Out A & B; Master Fader and one Input Fader at nominal level.
Maximum Voltage Gain	PROGRAM — 76±2dB from Channel In to Line Out A & B. 46±2dB from Channel In to Master Out. MONITOR — 78±2dB from Channel In to Monitor Out. ECHO — 76±2dB from Channel In to Echo Out. MASTER — 30±2dB from Master In to Line Out A & B.
Talkback	Microphone input jack, preamp, level control, and push-to-talk switch; to echo out and pgm. buses and/or direct out.
Inputs to Console	32 x Channel Inputs (microphone and line sources). 24-channel consoles are fitted with blank input modules for channels 1 through 8.
VU Meters (0 VU=+4dBm)	8 x large, illuminated meters; 4 for Master (Line Out) and 4 for Monitor (Monitor Out).
Phantom Power	48V dc applied to balanced channel input transformers for powering condenser microphones. Individual, rear-panel ON/OFF switches for each channel.
Power Supply	Self-contained modules inside the console, fused and fully regulated. Requires 110-120V AC, 50-60Hz, 90 Watts. May be converted for 220-240V AC operation.
Dimensions	58-1/8" (147.7cm) wide x 37" (94.2cm) deep x 12-7/8" (32.7cm) high.
Weight	Overall — 32-ch. 308 pounds (140kg) 24-ch. 292 pounds (133kg). Modules — Blank, 1.2 pounds (0.55kg) Input, 3.2 pounds (1.45kg).

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



YAMAHA

International Corp., Box 6600, Buena Park, Calif. 90620