

Installation Notes

4.1 PLANNING AN INSTALLATION

Before installing the PM1800A, it is worthwhile considering how it will be used, how it is going to be connected, and what is the best way to implement the installation.

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

Be sure to provide a location within 10 feet (3.6 meters) of the console for housing the PW1800 power supply. This supply may be rack mounted, or it may be placed on a shelf. For touring or critical fixed applications, it may be advisable to purchase a spare PW1800 supply and to keep it next to the main supply for rapid changeover in the rare event of a problem.

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

4.2 POWER MAINS

4.2.1 Verify the Correct Mains Voltage

PW1800 power supplies sold in the U.S.A. and Canada are designed to operate with 110 to 120 volt, 50 or 60 Hz AC power mains. The General Export model operates on 220 or 240 volt, 50 or 60 Hz AC mains. If you are traveling with this equipment, be sure to test the power mains, and to use the appropriate power supply. Consult your Yamaha PM1800A dealer for assistance.

4.2.2 Ensure There is a Good Earth Ground

The console must be grounded for safety and proper shielding. A 3-wire power cable is provided for this purpose. Use a special circuit tester to insure that the outlet is properly grounded, and that the "neutral" is not weak or floating. If a grounded, 3-wire outlet is not available, or if there is any chance the outlet may not be properly grounded, a separate jumper wire must be connected from the console chassis to an earth ground.

In the past, cold water pipes often were relied upon for an earth ground, although this is no longer the case in many localities. Modern building codes often specify

that the water meter be isolated from the water mains by a length of plastic (PVC) pipe; this protects water company personnel working on the water mains from being shocked. It also insulates the cold water pipes from the earth ground. While an electrical wire bypasses the water meter in some locations, this ground path should not be assumed. For similar reasons, avoid hot water pipes. Gas pipes should not be used because if there is a poor electrical connection between two sections of pipe, and if a ground current is being dissipated through the pipe, there exists the potential for a heat or spark-generated fire or explosion. The safest and most reliable approach is to provide your own ground. Drive at least 5 feet (1.5m) of copper pipe into moist, salted earth, and use that for a ground, or use one of the specially made chemical-type ground rods available for this purpose.

CAUTION:

Connect the PW1800 power supply to the power mains only after confirming that the voltage and line frequency are correct. At the least, use a voltmeter. It is also a good idea to use a special outlet tester that will also indicate reversed polarity, weak or missing neutral, and weak or missing ground connections in the outlet. Test the power supply before connecting the umbilical cables to the console.

Severe over voltage or under voltage in the power mains can damage your equipment. For U.S.A. and Canadian models, the power line must measure more than 105V and less than 130V RMS. The tolerance for General Export models is plus or minus 10%. Some lines are "soft," meaning that the voltage drops when the line is loaded due to excessive resistance in the power line, or too high a current load on the circuit. To be certain the voltage is adequate, check it again after turning on the PW1800 with the PM1800A connected, and with any power amplifiers turned on if they are connected to the same power mains.

If the power line voltages do not fall within the allowable range, do not connect the PW1800 to the mains. Instead, have a qualified electrician inspect and correct the condition. Failure to observe this precaution may damage the power supply and console, and will void the warranty.

NOTE:

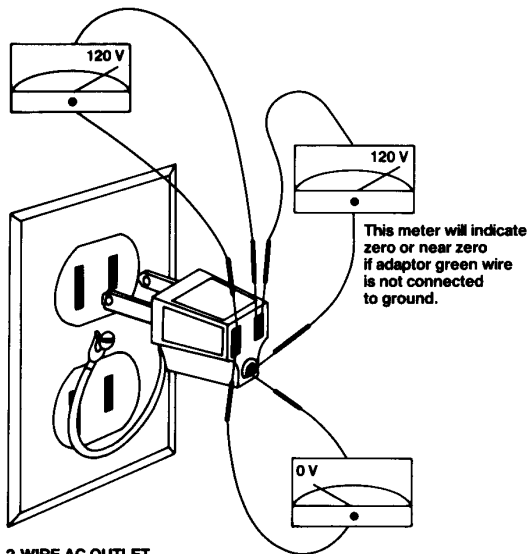
The following discussions of AC outlet wiring are written for U.S.A. and Canadian power systems, although the principles generally apply worldwide. In other areas, however, be sure to check local codes for specific wiring standards.

4.2.3 How to Obtain a Safety Ground when Using a 2-wire Outlet

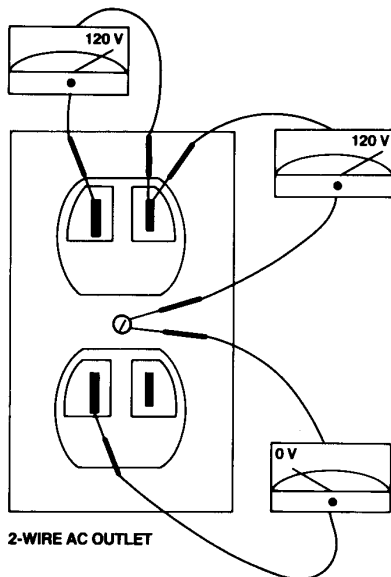
Two-wire AC outlets do not have a hole for the "safety ground" prong of a 3-wire power cord. A two-wire to three-wire AC adaptor is required if you want to use one of these two-wire outlets with the three-wire AC plug on your sound equipment. These adaptors can maintain a safe ground for the sound system if you connect the loose green wire on the adaptor to a grounded screw on the two-wire outlet. How do you know whether or not the screw is grounded?

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

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



2-WIRE AC OUTLET WITH A 3-WIRE ADAPTOR



2-WIRE AC OUTLET

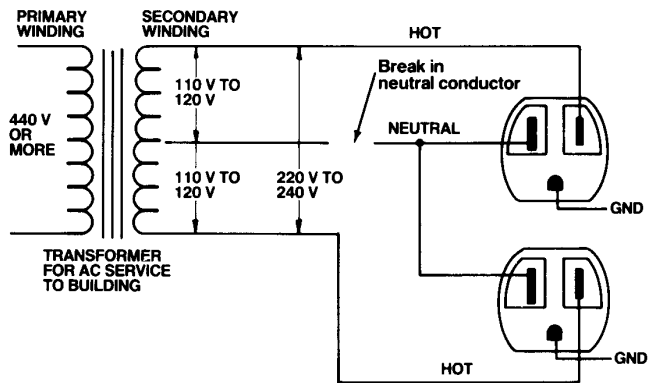
FIGURE 4-1. TESTING A 2-WIRE AC OUTLET.

4.2.4 Improperly Wired AC Outlets: Lifted Grounds

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

4.2.5 Improperly Wired AC Outlets: Lifted Neutral

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



Ordinary voltmeter or tester may not reveal this condition unless the two outlets have very different loads.

FIGURE 4-2. SCHEMATIC OF AN OUTLET WITH A LIFTED NEUTRAL.

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

If the PW1800 is plugged into one socket of the two outlets with lifted neutral, and a rack of signal processing equipment or power amplifiers is plugged into the other, fuses would probably blow upon turning on the system, and some of the sound equipment could be destroyed.

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

WARNING:

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

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

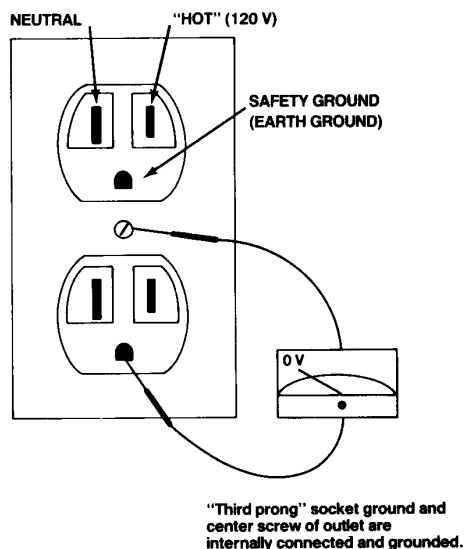
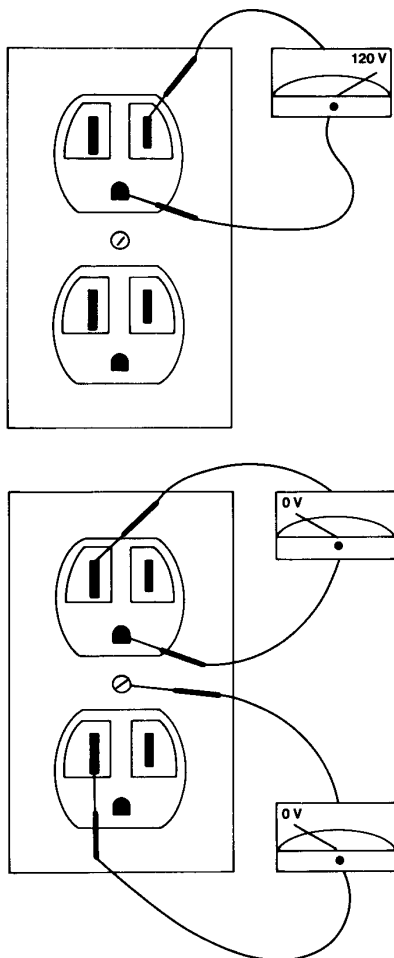


FIGURE 4-3. TESTING A 3-WIRE AC OUTLET.



4.2.6 AC Safety Tips

1. If you are going to verify the quality of AC wiring, there are two inexpensive items you should carry. One of these is a commercial outlet tester, the other is a neon lamp type AC voltage tester. These items are inexpensive and available at most hardware stores, electrical supply houses and some lighting stores. It is advisable to also have an RMS (or averaging) voltmeter to measure the exact AC line voltage.
2. The outlet tester should be used on all power outlets. The neon voltage tester should be used to check for voltage differences between microphone and guitar amps, microphones and electric keyboard chassis, and so forth.
3. If you're not sure whether an outlet is good, don't use it. Just in case, carry a long, heavy duty extension cord. A good extension should be made of #12-3 (12 gauge, 3 wires), and no longer than 15 meters (about 50 feet).
4. If there is no suitable power source at a venue, don't plug in your equipment. *Any fault* in the wiring of the AC outlet is potentially hazardous. Rather than take a chance with damage to equipment and possibly lethal shock, it is best to refuse to use a faulty outlet until it has been repaired by a licensed electrician. Don't take unnecessary risks.

4.3 THEORY OF GROUNDING

Grounding is an area of "black magic" for many sound technicians and engineers; and certainly for most casual users of sound systems. Everyone knows that grounding has something to do with safety, and something to do with hum and noise suppression, but few people know how to set up a proper AC power distribution system, and how to connect audio equip-

ment grounds so that noise is minimized. This subsection of the manual won't make anyone an expert, but it does point out a few of the principles and precautions with which everyone should be familiar. Whether you read this material or not, before you start cutting shields and lifting grounds, read this warning:

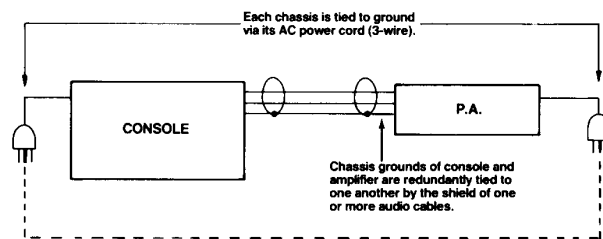
WARNING:

In any audio system installation, governmental and insurance underwriters' electrical codes must be observed. These codes are based on safety, and may vary in different localities; in all cases, local codes take precedence over any suggestions contained in this manual. Yamaha shall not be liable for incidental or consequential damages, including injury to any persons or property, resulting from improper, unsafe or illegal installation of a Yamaha mixing console or of any related equipment; neither shall Yamaha be liable for any such damages arising from defects or damage resulting from accident, neglect, misuse, modification, mistreatment, tampering or any act of nature. (IN PLAIN WORDS... IF YOU LIFT A GROUND, THE RESULTING POTENTIAL FOR ELECTRICAL SHOCK IS YOUR OWN RESPONSIBILITY!)

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

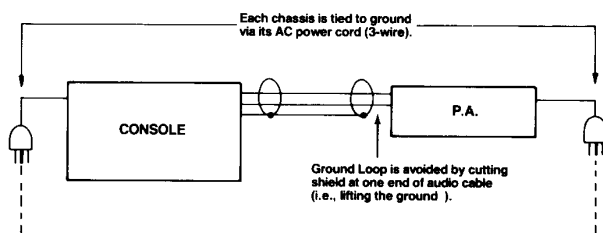
4.3.1 What is a Ground Loop, When is it Bad, and How is it Avoided?

The Ground Loop is perhaps the least understood, most widespread problem that turns up in one sound system after another. A ground loop is a multiple electrical path between two or more components — a path formed by the ground wiring, the chassis of the components themselves, or by combinations of these two main elements. Electrical noise current (induced RFI and power line hum) that flows through the shield, chassis, and/or AC power grounds can "loop" around from one piece of equipment to another. Normally, this should not cause any noise in the audio circuits if (a) the circuits are truly balanced or floating, and (b) the audio common is maintained separately from the chassis ground within the equipment. If one of these conditions is not met, then instead of going directly to earth ground and disappearing, these circulating "ground loop" noise currents (which act like signals) travel along paths that are not intended to carry signals. The currents, in turn, modulate the potential of the signal-carrying wiring (they are superimposed on the audio), producing hum and noise voltages that cannot easily be separated from program signals by the affected equipment. The noise is thus amplified along with the program material.



The ground path between the two AC plugs provides a redundant ground (ground loop) since the audio cable shield(s) already does the job.

A typical sound system ground loop caused by redundant audio shield and AC mains ground paths.



Dual ground path provided by AC cords does not create ground loop since the two chassis are not grounded redundantly via cable shield.

Elimination of the typical ground loop by cutting the shield of the audio cable retains AC safety.

FIGURE 4-4. TYPICAL GROUND LOOPS IN SOUND SYSTEMS.

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

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

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

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

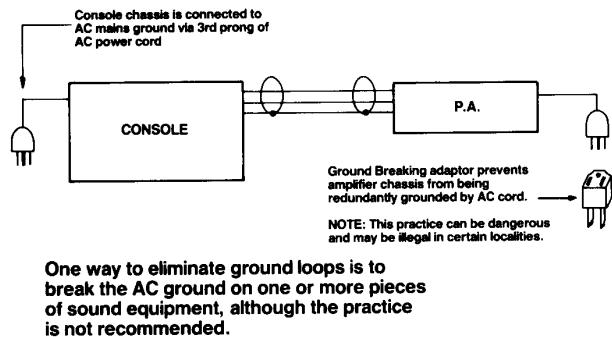


FIGURE 4-5. AVOID USE OF AC POWER CORD GROUND ADAPTOR TO "BREAK GROUND"; CONNECT GREEN WIRE TO OUTLET BOX.

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

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

4.3.2 Balanced Lines, Unbalanced Lines, and Ground Lift Switches

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

CAUTION:

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

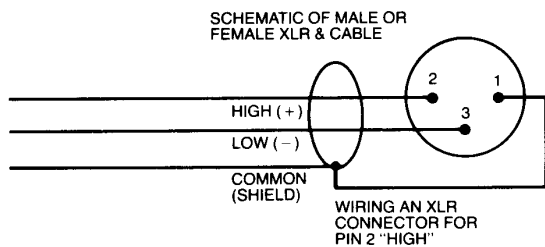
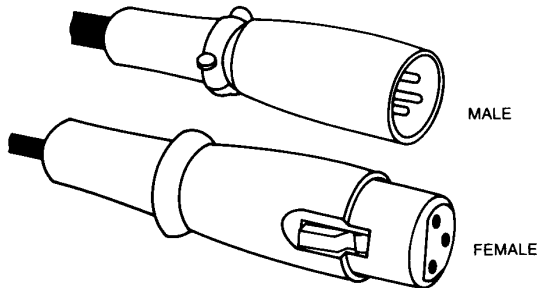
In those audio devices which anticipate ground loops by providing "ground lift" switches next to XLRs or three-wire phone jacks, the ground lift switch makes and breaks the connection between the connector's shield and the chassis of the particular device. Ground lift switches are usually found on "direct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

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

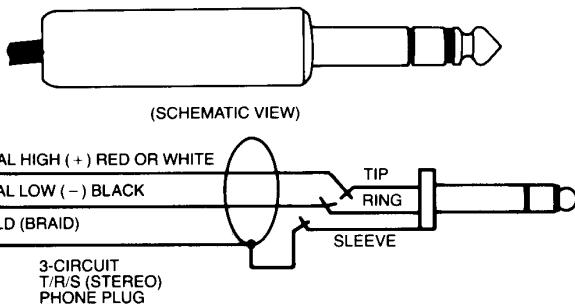
4.4 AUDIO CONNECTORS AND CABLES

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

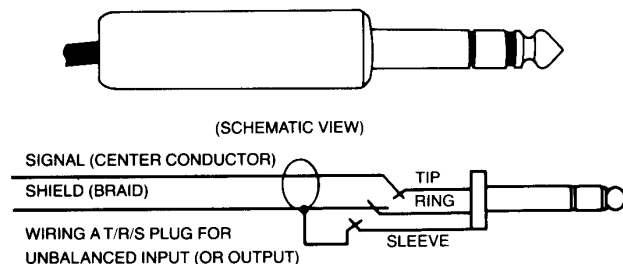
The PM1800A is fitted with only two types of audio connectors: 3-pin XLRs, both male and female, and 3-circuit (tip/ring/sleeve) 1/4" phone jacks (also known as stereo phone jacks, although their function is generally to carry an unbalanced mono signal rather than a stereo signal).



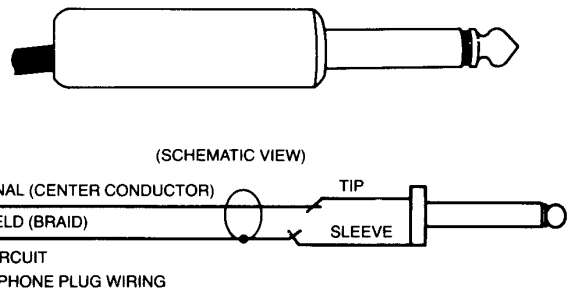
A) XLR-3 INPUT OR OUTPUT,



B) T/R/S PHONE PLUG FOR BALANCED AUX SUB IN (ONLY FOR 16CH MODEL).



C) T/R/S PHONE PLUG FOR UNBALANCED INSERT INPUT, INSERT OUTPUT, CUE OUT



D) T/S PHONE PLUG FOR INSERT INPUT, INSERT OUTPUT, CUE OUT

FIGURE 4-6. CONNECTOR WIRING FOR PM1800A:

4.4.1 Types of Cable to use

2-conductor (twisted pair) shielded cable is best for all XLR connections. Belden 8412, Canare L4E6S, or an equivalent are excellent choices due to their heavy duty construction, multiple strands that avoid breakage, good flexibility, and good shielding. Such cables are suitable for all portable applications, and for microphones. For permanent installation or for cables confined to portable racks or cases, a lighter duty cable such as Belden 8451, Canare L-2E5AT or an equivalent are suitable. "Snake" type multi-core cables containing multiple shielded pairs must be handled very carefully because the leads tend to be fragile, and a broken conductor cannot be repaired. If you are using a "snake," allow at least one or two spare channels that can be used in case of breakage in one of the channels in use.

4.4.2 Cable Layout

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

Similarly, avoid closely bundling the line-level outputs from the PM1800A with any mic-level inputs to the console. Specifically, avoid using a single multi-core "snake" cable for running mic lines from the stage and power amp feeds up to the stage. The close proximity of such cables promotes inductive and/or capacitive coupling of signals. If the stronger output signal from the console "leaks" into the lower-level mic or line feeding a console input, and that weaker signal is amplified within the console, a feedback loop can be established. This will not always be manifest as audible "howling," but instead may be manifest as very high frequency (ultrasonic) oscillation that indirectly causes distortion of the signal and that can lead to premature component failure. The best solution is to widely separate mic input cables from line-level output cables or, if not practical, to at least bundle them loosely.

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

4.4.3 Balanced Versus Unbalanced Wiring

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

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

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

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

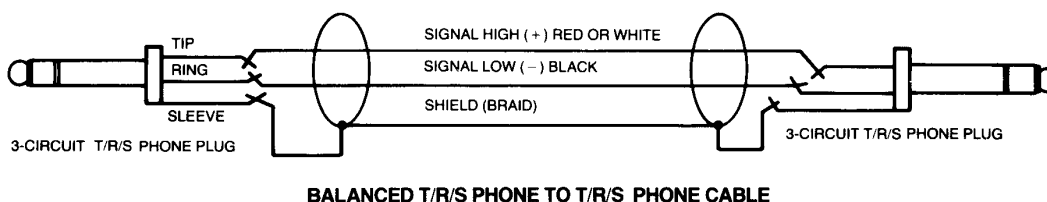
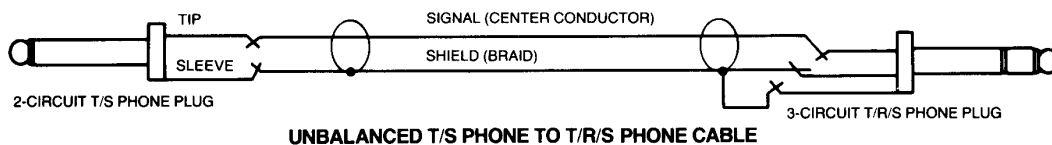
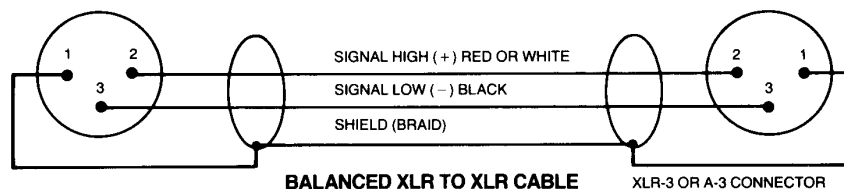
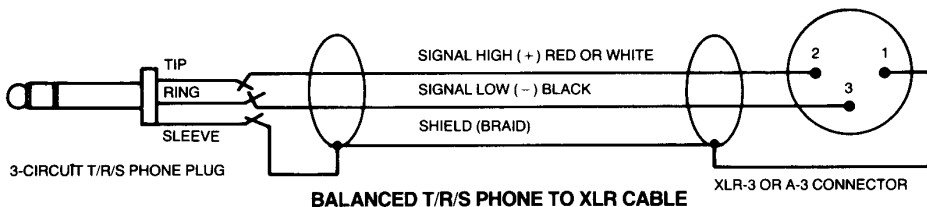
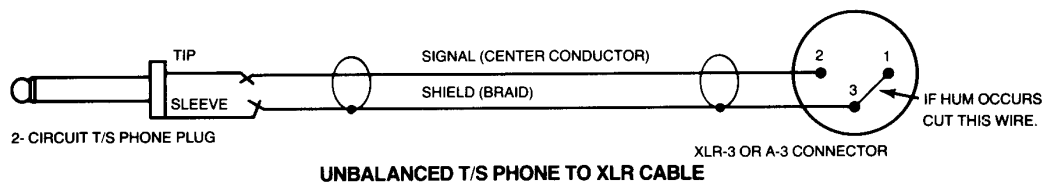


FIGURE 4-7. CABLES FOR UNBALANCED AND BALANCED LINES.

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

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

Balanced wiring helps eliminate some types of externally-generated noise. The two wires of the "balanced" cable carry the same signal, but each wire is opposite in signal polarity to the other. In a balanced input, both of the signal-carrying wires have the same potential difference with respect to ground (they are "balanced" with respect to ground), and the input is designed to recognize only the *difference* in voltage between the two wires, and (hence the term "*balanced*

differential input"). Should any electro-static interference or noise cut across a balanced cable, the noise voltage will appear equally — with the same polarity — on both signal-carrying wires. The noise is therefore ignored or "rejected" by the input circuit. (This is why the term "common mode rejection" applies; signals in *common* to the two center wires are *rejected*.)

Not all balanced wiring has a shield. In older telephone systems, many miles of cable were run with no shielding in order to save money (now fiber optic cables are replacing costly copper with inexpensive glass or plastics). Out in the open, wires are subjected to radio interference and to hum fields emitted by power lines. Balancing the two signal hot wires with respect to ground gives long lines immunity to external interference. Twisting two wires together theoretically subjects each wire to the same amount of electrostatic or electromagnetic noise. A balanced input will then cancel the unwanted noise signals common to both wires, while passing the desired audio signal, as illustrated in Figure 4-8.

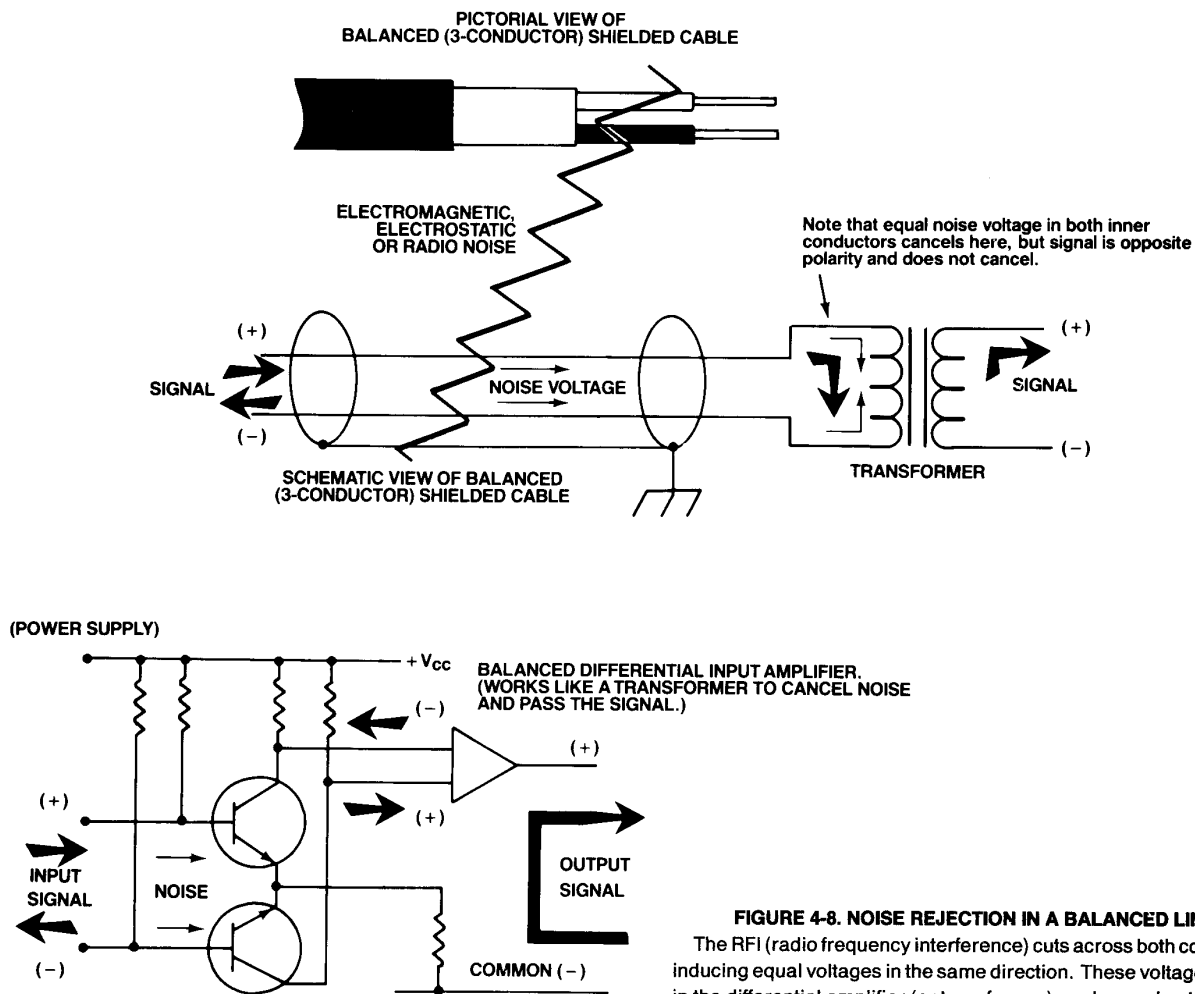


FIGURE 4-8. NOISE REJECTION IN A BALANCED LINE.

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

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

As illustrated, there are two means to achieving a balanced input; either with a transformer or with a differentially balanced amplifier (an "electronically balanced input"). The latter approach is used in the PM1800A, and was chosen for several reasons: (1) it is more "transparent" sounding than most transformer inputs, (2) it cannot be saturated by low frequency, high-level signals as can a transformer, (3) it is lighter in weight.

There are a number of reasons why input transformers are used in some installations. In the case of certain audio equipment which has an unbalanced input (not this console), a transformer converts the unbalanced input to a balanced input. Beyond that, there are cases where a transformer is desirable even if the input is electronically balanced. For example, where there is a significant amount of electrostatic or electromagnetically induced noise, particularly high-frequency high-energy noise (the spikes from SCR dimmers, for example), the common mode rejection voltage (CMV) of an electronically balanced input may be insufficient to handle the noise induced in the cable. In such cases, when the noise voltage exceeds the "rails" of the power supply in the electronic input, the common mode rejection ratio (CMRR) drops drastically, and noise comes right through the now unbalanced circuit. In these cases input transformers can be essential.

To a minor degree, safety is an issue with an electronically balanced input. For the ultimate in safety, there are instances when a transformer will isolate the console ground from the external source. Consider what happens, for example, when a performer is touching a mic and also touches an electrically "hot" item such as a guitar which is electrically "live" due to a fault in the guitar amp; if the mic is grounded, current will flow. The performer can be subjected to very high currents, and to consequently severe AC shock. If the mic is isolated from ground, via a transformer, then that low-resistance return path for the AC current is eliminated, and the performer has a better chance of surviving the shock. The truth, in reality, is that the transducer capsule in a microphone is generally isolated and insulated from the mic case, so an electronically balanced input still would not permit a current to flow through the mic... assuming everything is wired correctly in the microphone. If a transformer is used in this way, primarily for ground isolation and to obtain the benefits of a balanced line, it is said to be an "isolation" transformer.

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

In general, the PM1800A has no need for input transformers since it already has electronically balanced inputs. In the occasional instances where absolute isolation of the grounds between the console and the other equipment must be obtained, as cited above, there is no viable substitute for a transformer, and an optional input transformer kit (Model IT1800A) can be installed in individual input modules. Similarly, PM1800A outputs can be transformer isolated by purchasing one or more optional output transformer sets. The Model OT1800 output transformer set contains 4

transformers, and the Model OT3000 contains 8 transformers; each set includes XLR connectors in a compact package that is external to the PM1800A. In this way, those inputs or outputs which require a transformer can be so equipped, and it is not necessary to pay the price, carry the weight or incur the slight performance penalty that comes with the transformers.

NOTE:

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

4.4.5 Noise and Losses in Low and High Impedance Lines

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

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

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

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

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

4.5 DIRECT BOXES

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

NOTE:

If a preamplifier head is used (such as the Yamaha PG-1 or PB-1), a direct box is not necessary since the head provides a balanced, isolated output to a console.

One further application of the direct box is to isolate and pad the speaker-level output of an instrument amplifier so that signal can be fed to the console input. Normally, one would not connect a speaker-level signal to a console input. However, the reverb, tremolo, distortion, EQ, and other characteristics of many instrument amps are an integral part of the instrument's sound. If the amp head does not provide a line-level output for a console, then a suitably designed direct

box can "tap" the speaker output for feed to the console. Even where a line level output is provided, sometimes the coloration of the signal at the speaker output (due to intentional clipping of the power amp section of the guitar amplifier, and back EMF from the speaker) is desired, and can only be obtained at the speaker terminals.

There are two main variations of the direct box: the passive version, with only a transformer, and the active version, which employs a powered circuit in addition to the transformer and thus provides minimum pickup loading while boosting low level signals from the guitar pickup for maximum noise immunity. We present information here for constructing one of each of these types of direct boxes. Credit should be given to the designer of the boxes, Deane Jensen, of North Hollywood, CA. While these designs are believed to work well with the PM1800A, their inclusion in this manual does not represent an endorsement by Yamaha of the specific products mentioned. The specified transformers are available from Jensen Transformers, Inc., 10735 Burbank Blvd., North Hollywood, CA 91601. Phone (213) 876-0059, TELEX via WUI 6502919207 MCI UW.

4.5.1 Passive Guitar Direct Box

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

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

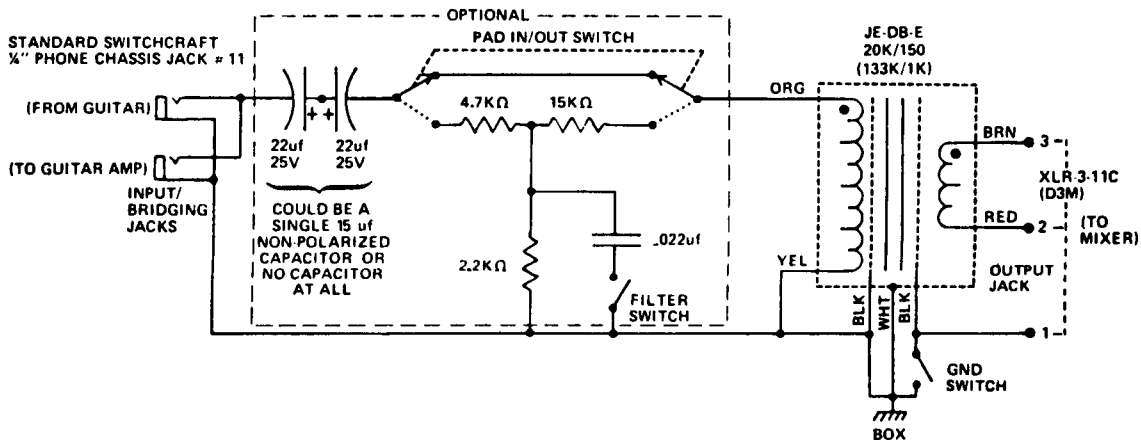


FIGURE 4-9. PASSIVE DIRECT BOX SCHEMATIC DIAGRAM.

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

The transformer was designed specifically for use in a guitar direct box. When connected to a typical electric guitar pickup, and an XLR channel input on a PM1800A, the transformer reflects the optimum load impedance to both the guitar pickup and the mic preamp input. This preserves optimum frequency response and transient response. The transformer has two Faraday shields to prevent grounding and shielding problems that could cause hum in the PM1800A or the guitar/instrument amplifier. Place the ground switch in whichever position works best.

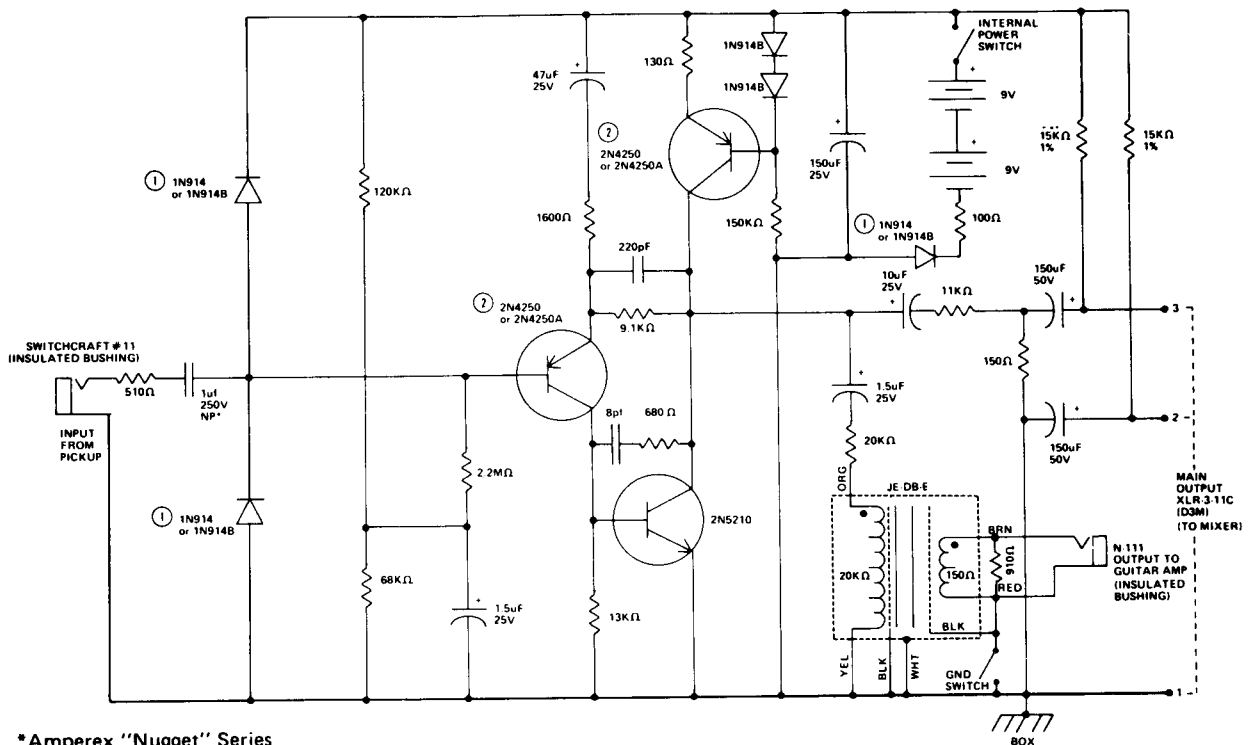
Assembly can be accomplished in a small metal box. Keep the phone jack electrically isolated from the chassis of the box. During operation, keep the chassis of the box away from the chassis of any guitar/instrument amp or any other grounded object. If you decide to use a transformer other than the Jensen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

proximate 133K ohm "load" for the guitar when connected to a nominal 1K ohm console input (the approximate actual load impedance of most mic inputs). The PM1800A's electronically balanced XLR inputs are rated at 3K ohms, so the load on the guitar pickup would be nearly 500K ohms, which is ideal. Each winding, each Faraday shield, and the transformer chassis shield should have separate leads.

4.5.2 Active Guitar Direct Box

The active direct box shown here can be used at the output of a standard electric guitar, with or without an amplifier. Because of its very high input impedance, it can be used with a piezoelectric instrument pickup, taking the place of the preamp that is normally included with such pickups. This box is *not* meant for use at the output of a guitar amplifier (see PASSIVE DIRECT BOX information). The active direct box can be powered by its own pair of standard 9V "transistor radio" type batteries, or by phantom power from the PM1800A or any condenser microphone power supply.

The circuit can be constructed on a piece of perf board, or on terminal strips, or on a printed circuit layout. It should be assembled into a shielded case, using isolated (insulated) phone jacks, as shown. When the direct box is used between the guitar and guitar amplifier, place the ground switch in the position that yields the minimum hum. As with the passive direct box, any part substitution should be carefully considered.



*Amperex "Nugget" Series

- ① 1N914B may be substituted for 1N914.
- ② 2N4250 may be substituted for 2N4250A.

FIGURE 4-10. ACTIVE DIRECT BOX SCHEMATIC DIAGRAM.

Gain Structure and Levels

5.1 STANDARD OPERATING LEVELS

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

A. Mic Level or Low Level

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

B. Line Level or Medium Level

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

C. Speaker Level and High Level

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

NOTE:

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

Let’s discuss these levels in the context of a sound system. The lowest power levels in a typical sound system are present at the output of microphones or phono cartridges. Normal speech at about one meter from the “average” dynamic microphone produces a

power output from the microphone of about one trillionth of a watt. Phono cartridges playing an average program selection produce as much as a thousand times this output — averaging a few billionths of a watt. These signals are very weak, and engineers know that they cannot be “run around” a chassis or down a long cable without extreme susceptibility to noise and frequency response errors. This is why microphone and phono preamps are used to boost these very low signal levels to an intermediate range called “line level.” Line levels are between 10 millionths of a watt and 250 thousandths of a watt (1/4 watt). These levels are related to the “dBm” unit of measurement as follows:

–20 dBm	=	10 microwatts	=	0.0001 watts
0 dBm	=	1 milliwatt	=	0.001 watts
+4 dBm	=	2.5 milliwatts	=	0.0025 watts
+24 dBm	=	250 milliwatts	=	0.025 watts
+30 dBm	=	1000 milliwatts	=	1.0 watts
+40 dBm			=	10.0 watts
+50 dBm			=	100.0 watts

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

5.2 DYNAMIC RANGE AND HEADROOM

5.2.1 What is Dynamic Range?

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

5.2.2 The Relationship between Sound Levels and Signal Levels

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

The program, now converted to electrical rather than acoustic signals, still has a dynamic range of 90 dB: +24 dBu – (–66 dBu) = 90 dB. This dB SPL to dBu or dBm correspondence is maintained throughout the sound system, from the original source at the microphone, through the electrical portion of the sound sys-

tem, to the speaker system output. A similar relationship exists for any type of sound reinforcement, recording studio, or broadcast system.

5.2.3 A Discussion of Headroom

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

5.2.4 What Happens When the Program Source has Wider Dynamics than the Sound Equipment?

If another mixing console were equipped with a noisier input circuit and a less capable output amplifier than the previous example, it might have an electronic noise floor of -56 dBu (1.23 millivolts), and a peak output level of +18 dBu (6.16 volts). The dynamic range of this system would only be 74 dB. Assuming the original program still has an acoustic dynamic range of 90 dB, it is apparent that 16 dB of the program will be "lost" in the sound system. How is it lost? There may be extreme clipping of program peaks, where the output does not rise higher in response to higher input levels. Quiet passages, corresponding to the lowest signal levels, may be buried in the noise. Typically, portions of that 16 dB difference in dynamic range between the sound system capability and the sound field at the microphone will be lost in both ways. A system with +24 dBu output capability and a -66 dBu or better noise floor, or +18 dBu output capability and -72 dBu noise floor, would be able to handle the full 90 dB dynamic range. Thus, for high quality sound reinforcement or music reproduction, it is necessary that the sound system be capable of low noise levels and high output capability.

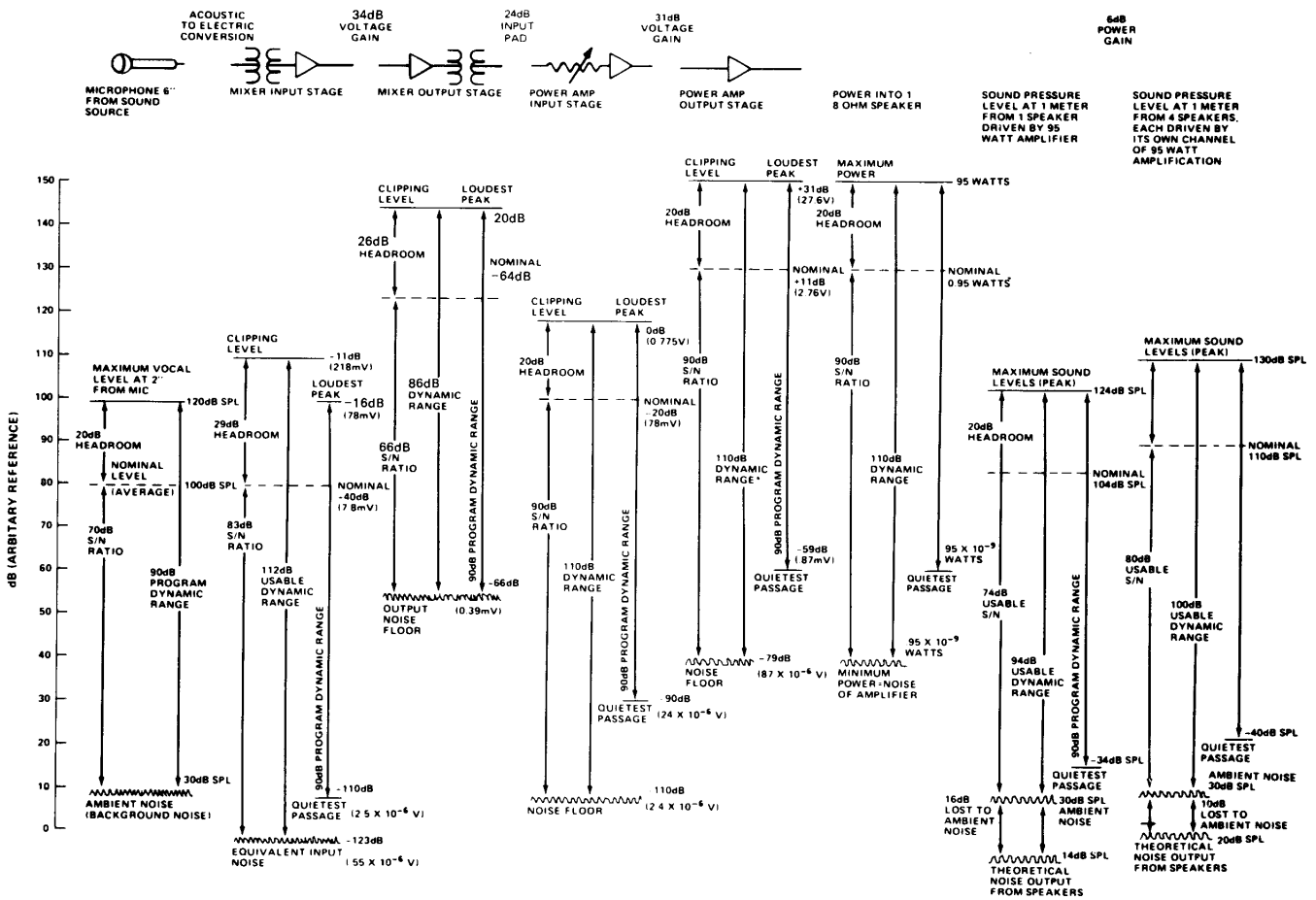


FIGURE 5-1. DYNAMIC RANGE AND HEADROOM IN SOUND SYSTEMS.

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

5.2.5 A General Approach to Setting Levels in a Sound System

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

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

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

Check the operating manual of each piece of equipment to determine the specified nominal and maximum input levels. An accurate AC voltmeter is often helpful for verifying levels. As a rule, keep signal levels as high as possible throughout the system, up to the input of the power amplifier(s); at that point, reduce the program level, as required to achieve a given headroom value, using the amplifier's input attenuators. Input attenua-

tors should be set so that maximum program levels from the source equipment won't drive the amplifiers to clipping (or at least, won't do it very often). This keeps overall system noise as low as possible.

5.2.6 How to Select a Headroom Value and Adjust Levels Accordingly

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

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

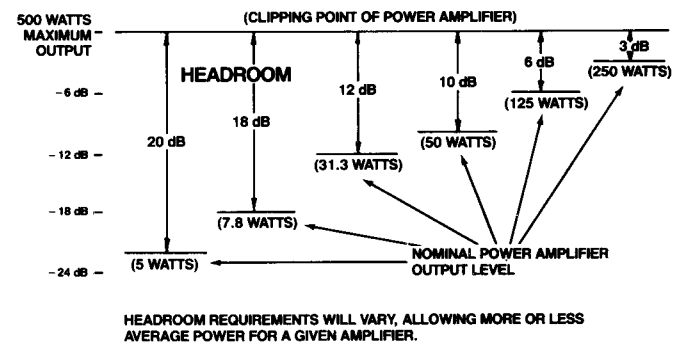


FIGURE 5-2. HEADROOM IN DIFFERENT APPLICATIONS

Let's go through an actual setup procedure for a high quality, music reproduction system. First choose a headroom figure. For maximum fidelity when reproducing music, it is desirable to allow 20 dB of headroom above the average system output. While some extreme musical peaks exceed 20 dB, the 20 dB figure is adequate for most programs, and allowing for greater headroom can be very costly. A 20 dB headroom figure represents a peak level that is one hundred times as powerful as the average program level. This corresponds to an average 0 VU indication on the PM1800 meters (0 VU = +4 dBu, which allows 20 dB headroom before the console reaches its maximum +24 dBu output level).

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

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

1. Initially, set the attenuators on the power amp at maximum attenuation (usually maximum counterclockwise rotation). Feed a sine wave signal at 1000 Hz to the console input at an expected average input level (approximately -50 dBu (2.45 mV) for a microphone, +4 dBu (1.23 volts) for a line level signal). The exact voltage is not critical, and 1000 Hz is a standard reference frequency, but any frequency from 400 Hz to about 4 kHz may be used.
2. Set the input channel level control on the console at its marked "nominal" setting, and adjust the master level control so that the output level is 20 dB below the rated maximum output level for the console. Suppose, for example, the maximum rated output level is +24 dBu (12.3 volts); in that case, the output level should be adjusted to +4 dBu (1.23 volts), as indicated by a "zero" reading on the console's VU meter (0 VU corresponds to +4 dBu output per factory calibration).
3. Assume that the rated maximum input level for the graphic equalizer to which the console output is connected is +14 dBu (3.88 volts). Subtracting +4 dBu from +14 dBu leaves only 10 dB of headroom, so in order to maintain the desired 20 dB of headroom, a 10 dB resistive pad should be inserted between the console output and the equalizer input. The signal level at the input to the equalizer should now be -6 dBu (388 mV), which can be confirmed with a voltmeter.

NOTE:

If the graphic equalizer is inserted in the console's group or stereo INSERT IN/OUT loop, that signal level is already a nominal -6 dBu when the VU meters are at 0 VU, so no pad would be required.

4. Assume that the maximum rated output level of the equalizer in this example is +18 dBu (6.16 volts). Adjust the master level control on the equalizer so that its output level is 20 dB below the rated maximum, or -2 dBu (616 mV). If the equalizer has no built-in VU meter, use an external voltmeter to confirm this level.

NOTE:

If the graphic equalizer is placed in the console's group or stereo INSERT IN/OUT loop, the nominal sensitivity of the input is -6 dB, so the equalizer output can be reduced to that level, providing another 4 dB

of headroom, which is a good idea anyway since it will allow for more EQ boost without overdriving the equalizer output.

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

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

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

5.3 GAIN OVERLAP AND HEADROOM

As explained previously, the PM1800A can deliver +24 dBu output level, a level which exceeds the input sensitivity of most other equipment. A power amplifier's sensitivity, for example, is that input level which drives the amplifier to maximum output (to the point of clipping). Hence, a power amplifier with a +4 dBu sensitivity rating will be driven 20 dB into clipping if driven with the full output capability of the PM1800A. It would appear, then, that the console has "too much" output capability, but this is not really true.

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

In other cases, the PM1800A may be driving a passive device such as a passive filter set, graphic equalizer or low-level crossover network. Such devices will attenuate some of the signal, often 6 dB or more. Here, the extra output capability of the console offsets the loss of the passive signal processor so that adequate signal

can be delivered to the power amplifiers, tape machine inputs, etc.

Consider those instances where the PM1800A outputs are connected to a tape machine. Many professional tape machines are subject to tape saturation at input levels above +15 dBu. Why would one want +24 dBu output from a console? Well, it turns out that analog tape has what is considered a "soft" saturation characteristic, whereby the distortion is not terribly harsh in comparison to the clipping of the typical solid state line amplifier. If the mixing console were to clip at +18 dBu, for example, that clipping would overlay a very harsh distortion on the 3 dB of "soft" saturation on the tape. Because the PM1800A does not clip until its output reaches +24 dBu, there is less chance of applying harsh distortion to the tape. Today, however, there is another consideration: digital recording technology. Here, the available dynamic range of the tape recorders is so great that all the headroom a console can provide is advantageous.

SECTION 6

Optional Functions

The PM1800A is factory wired to suit what Yamaha engineers believe to be the greatest number of applications. Yamaha recognizes, however, that there are certain functions which must be altered for certain specific applications. In designing the PM1800A, a number of optional functions have been built in, and can be selected by moving jumper wires within certain modules.

WARNING:

- 1) Underwriter's Laboratories (UL) requires that we inform you there are no user-serviceable parts inside the PM1800A. Only qualified service personnel should attempt to open the meter bridge, to remove a module, or to gain access to the inside of the console or power supply for any purpose. Lethal voltages are present inside the power supply, and the AC line cord and console umbilical cord should be disconnected prior to opening the console.
- 2) We at Yamaha additionally caution you never to open the console and remove or install a module for the purpose of inspection, replacement or changing the internal jumpers unless the power has first been *turned off*. If a module is removed or installed with power on, the circuitry may be damaged. Unless you are a qualified service technician, do not plug in the AC cord while the interior of the power supply is exposed; dangerous voltages may exist within the chassis, and lethal shock is possible. Yamaha neither authorizes nor encourages unqualified personnel to service modules or console internal wiring.

Damage to the console, the individual, and other equipment in the sound system can result from improper service or alterations, and any such work may void the warranty.

6.1 REMOVING AND INSTALLING A MODULE

The modules in the PM1800A are designed for easy removal.

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

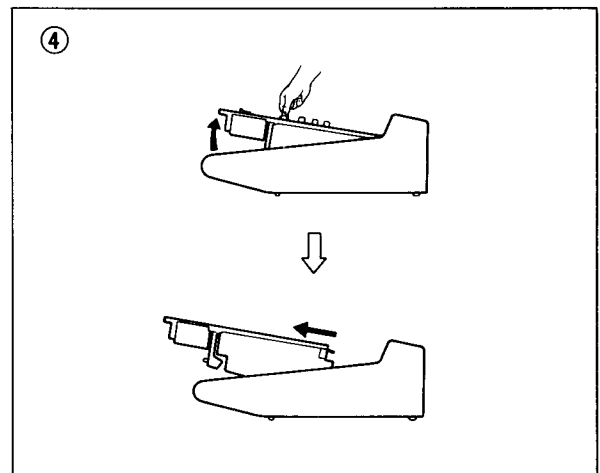
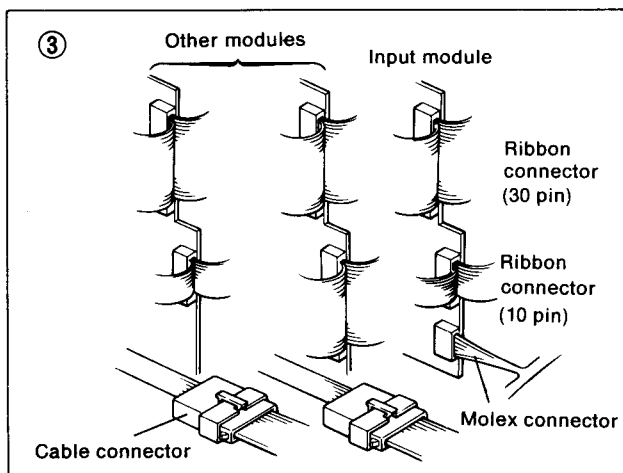
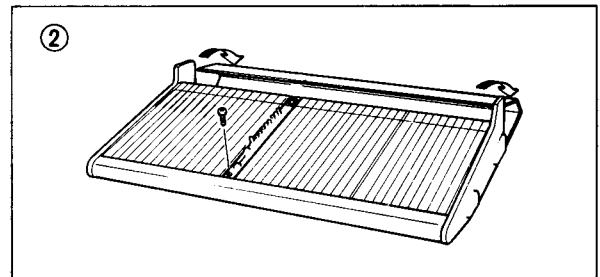
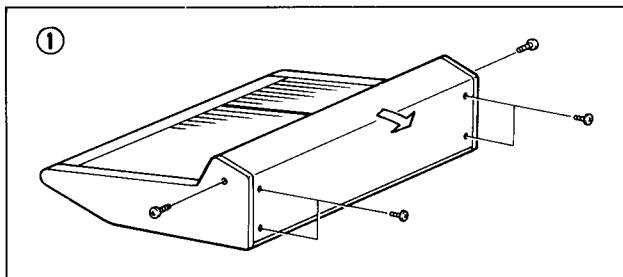


FIGURE 6-1. REMOVAL OF MODULE FROM PM1800A.

6.2 INPUT CHANNEL AUX 3 THROUGH 6 SENDS: PRE OR POST FADER

As shipped, the console is wired so that the AUX 3, AUX 4, AUX 5 and AUX 6 send controls in each input module derive signal ahead of the fader, equalizer and high pass filter. If desired, individual sends can be al-

tered, by moving an internal jumper, so they are derived after the fader, EQ and filter.

Note that the factory "Pre" position supplied can be altered to be post EQ & filter, but still pre fader, as described in Section 6.3.

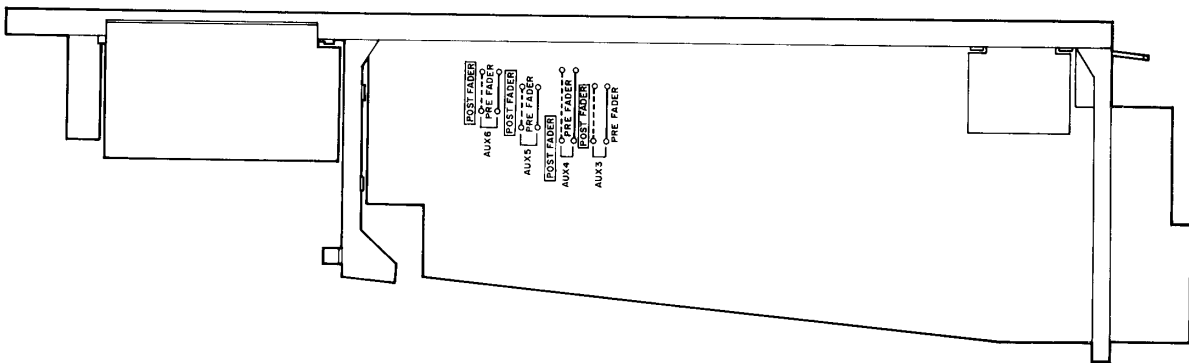
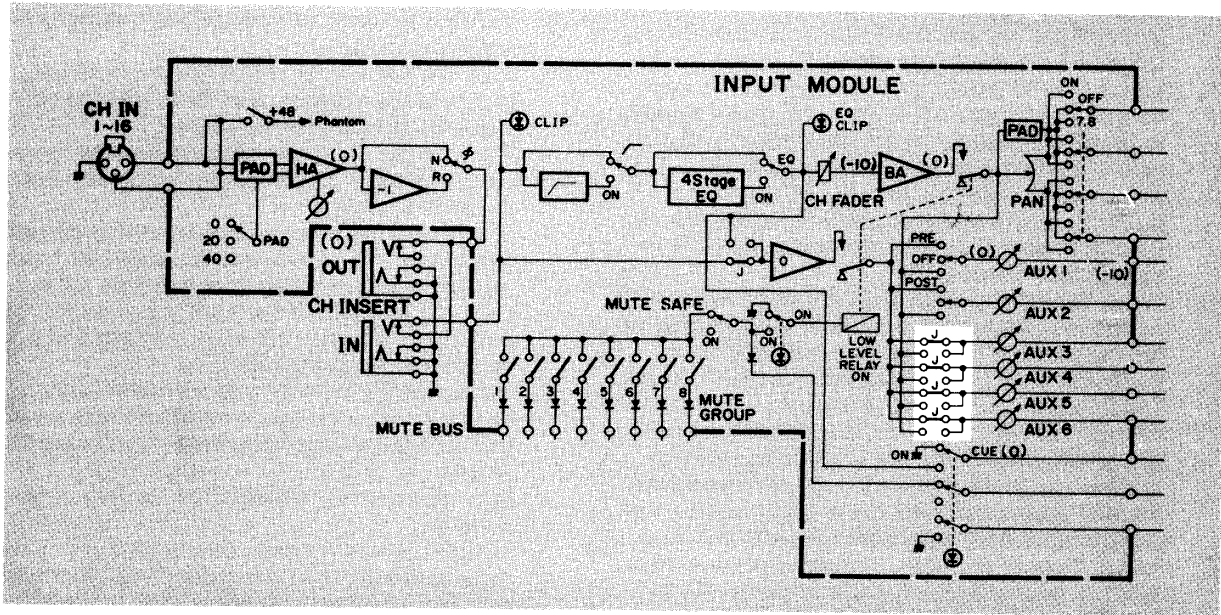


FIGURE 6-2. INTERNAL JUMPER WIRING FOR PRE OR POST FADER (& EQ) SENDS ON AUX 3 THROUGH AUX 6

6.3 INPUT CHANNEL AUX SEND PRE POSITION: PRE FADER & EQ OR PRE FADER/POST EQ

A jumper wire in each input module permits all six auxiliary sends to be altered. As shipped, the console is wired so that if front-panel PRE/OFF/POST switch on the AUX 1 or AUX 2 send is set to PRE position, the aux send is derived ahead of the fader, equalizer and high pass filter. This is useful for stage monitor work, for example, where the channel EQ for the house may not be desired for the monitors. On the other hand, suppose that one aux mix is used for a pre-fader effects send. In this case, it may be desirable to apply channel

EQ and HP filter effects to the send, yet the POST position would also cause the channel fader to affect the send. To solve the problem, the jumper can be moved so that the PRE position remains pre-fader, but is taken after the EQ and HP filter.

Note that, as factory wired, the AUX 3, AUX 4, AUX 5 and AUX 6 sends are also pre-fader and pre-EQ. If you move this jumper, it will also change those four sends so they are pre-Fader, Post EQ. (It has no effect, however, if you alter the individual jumpers on AUX 3, 4, 5 or 6 to make them Post-Fader & EQ sends).

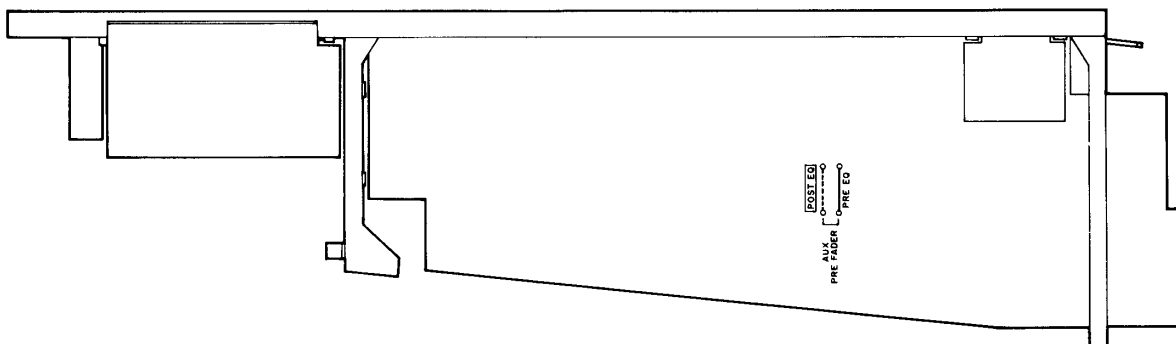
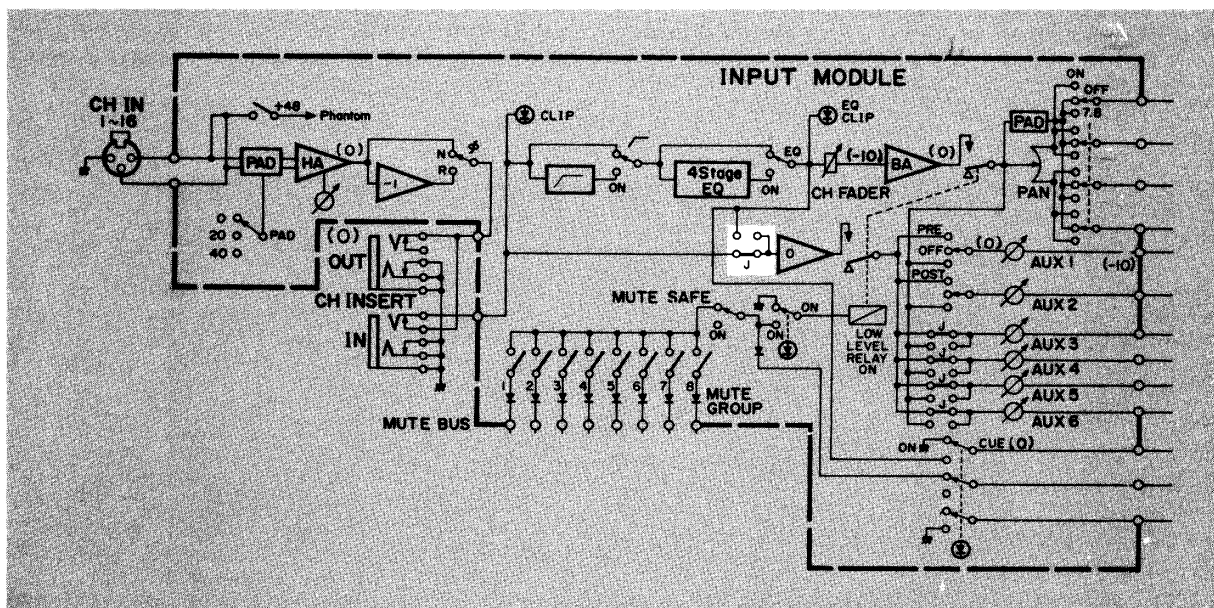


FIGURE 6-3. INTERNAL JUMPER WIRING FOR PRE-EQ AND POST-EQ AUX SENDS (WHEN AUX 1 OR AUX 2 PRE/OFF/POST SWITCH IS SET TO "PRE," OR AUX 3 THROUGH 6 JUMPERS ARE IN FACTORY-WIRED "PRE" POSITION).

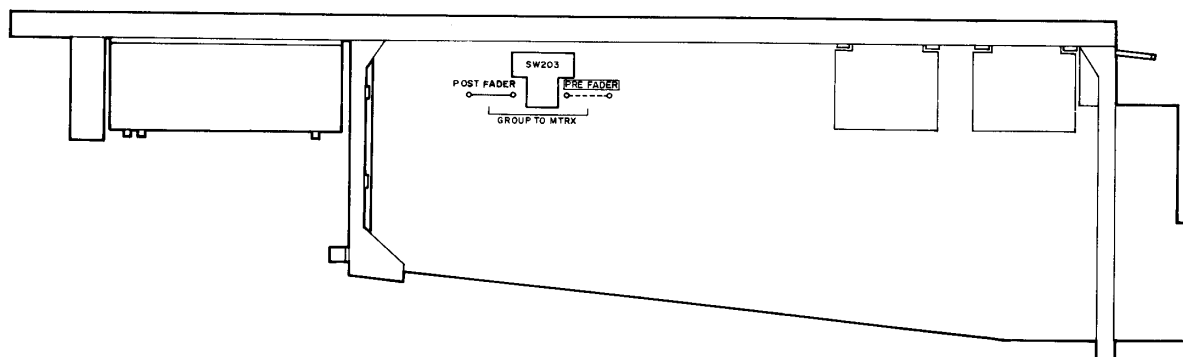
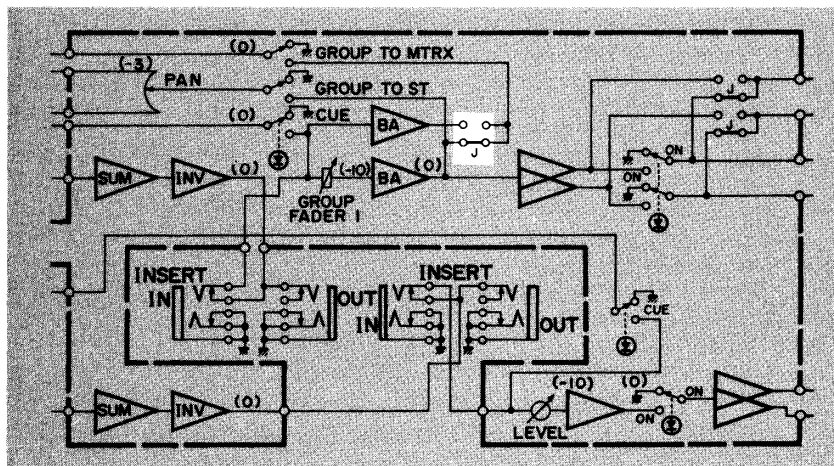
6.4 GROUP-TO-MATRIX: ASSIGNED PRE OR POST GROUP MASTER FADER

A jumper wire in each master module permits the eight group sends to the mix matrix to be altered. As shipped, the console is preset so that when the GROUP-TO-MATRIX switch is on, the matrix is fed signal after the Group Master Fader (but before the GROUP ON/off switch). The internal jumper in each of these modules can be repositioned so that the matrix is fed before the Group Master Fader.

In the factory preset configuration, the matrix follows the group mix. If one group, for example, is used for vocals, another for keyboards, etc., then all vocals going to all matrix outputs can be adjusted with one Group Master Fader... all Keyboards going to all matrix outputs can be adjusted with another Group Master Fader, etc. Suppose, however, that you plan to feed a stereo

house mix from the eight subgroups, yet you need as many as eight additional mono or five stereo mixes.

The mix matrix alone allows for only one stereo and two mono mixes, or a total of two stereo mixes. A greater number of mixes can be obtained by selecting the alternate (pre-Group Master Fader) jumper positions. In that case, you can assign the Group Outputs to the stereo bus via the ST switch [38] and the adjacent PAN pot [37]; the Group Master Faders will serve as submasters for this stereo mix, and the Stereo Master Fader will control the mixed output. At the same time, the matrix controls on each master module will provide an 8:1 mix of the same groups; that matrix channel's #1 - #8 mix controls will serve as submasters, and the MATRIX MASTER will control the mixed output. In this way, you can obtain one stereo and four mono mixes, three stereo mixes, or some combination thereof all with independent submaster and master controls.



(RTRN)

FIGURE 6-4-A. INTERNAL JUMPER WIRING FOR PRE- AND POST- GROUP MASTER FADER FEEDS TO MIX MATRIX.

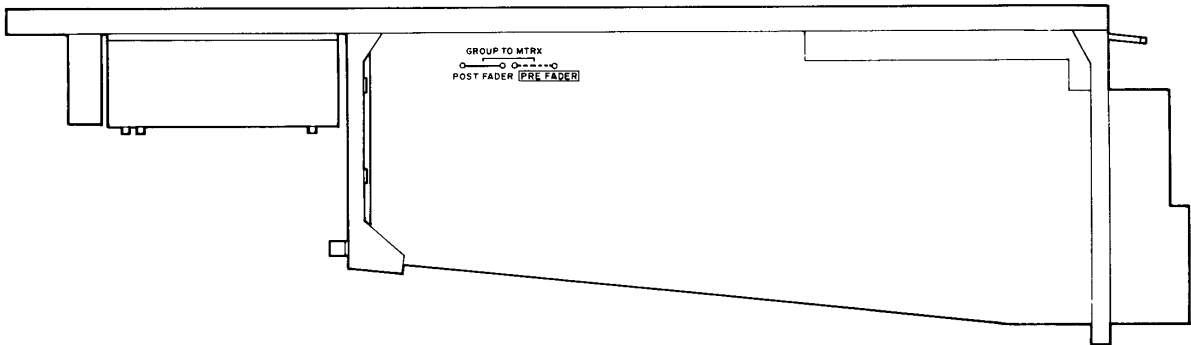
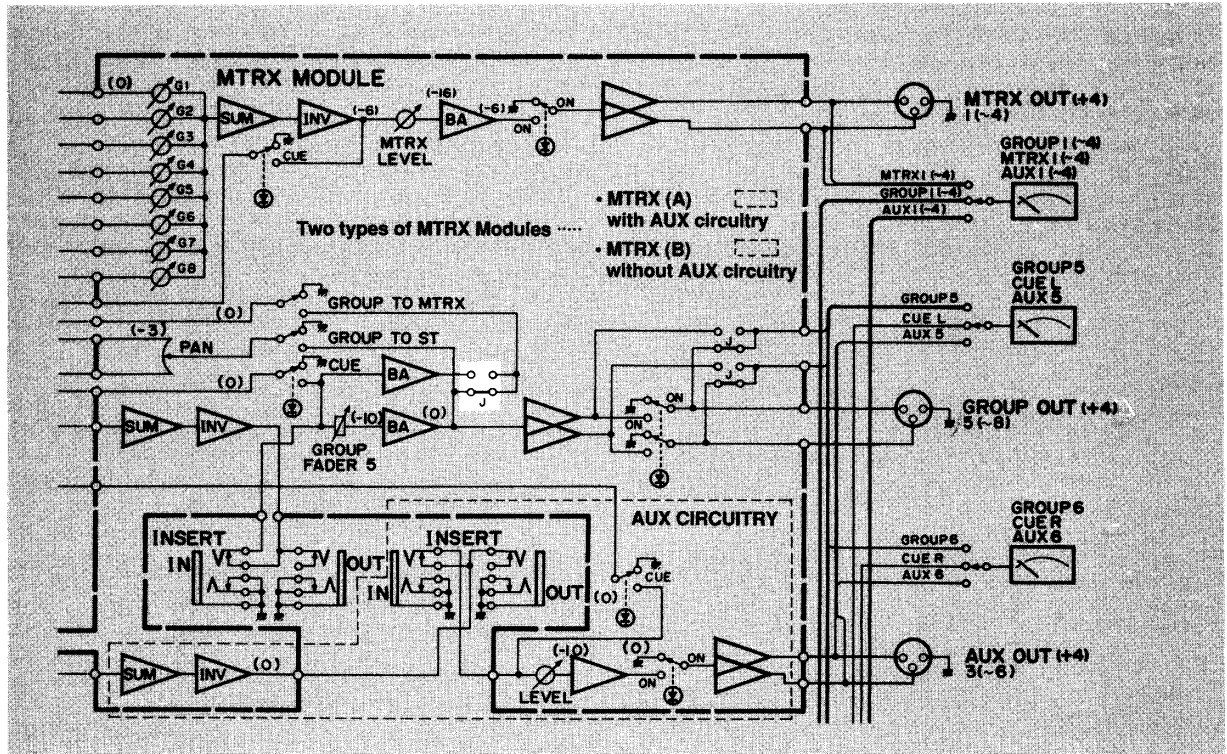
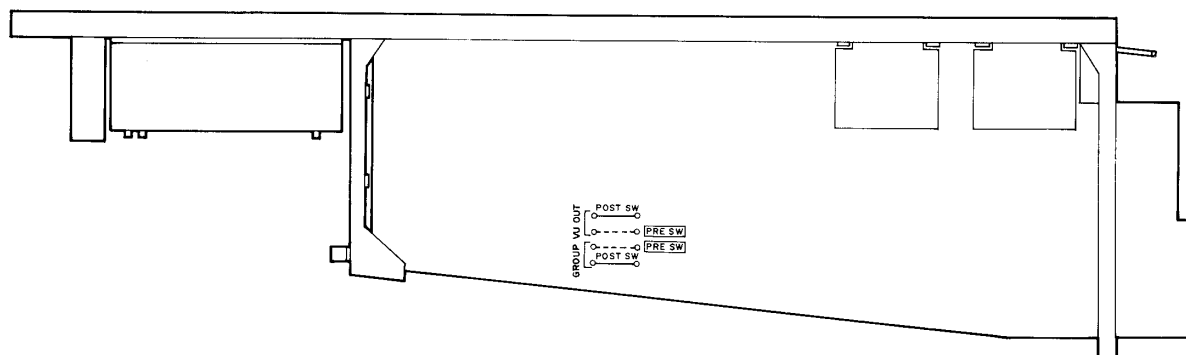
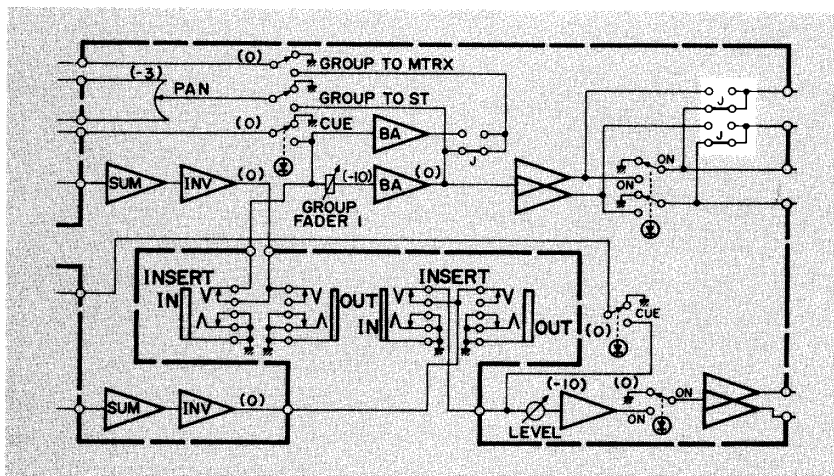


FIGURE 6-4-B.

6.5 METER FUNCTION IN "GROUP" POSITION: PRE OR POST GROUP ON/OFF SWITCH

There are eight VU meters which can be switched to monitor the GROUP output levels. Specifically, as factory wired, these meters normally monitor the Group output after the Group Master Faders and On/Off

switches. Thus, if a Group Output is switched off, there will be no meter deflection. In some cases, you may wish to preview group levels before turning on the output. An internal jumper on each group module permits you to do this.



(RTRN)

FIGURE 6-5-A. INTERNAL JUMPER WIRING FOR GROUP METERING BEFORE OR AFTER THE GROUP ON/OFF SWITCH.

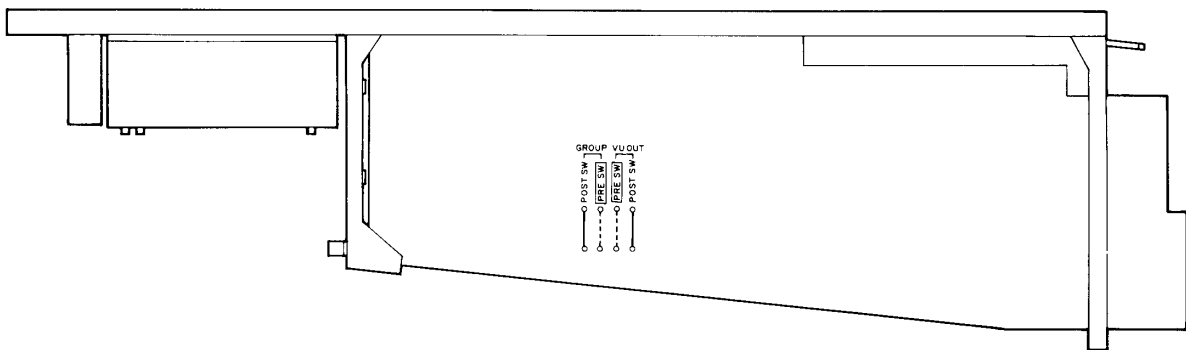
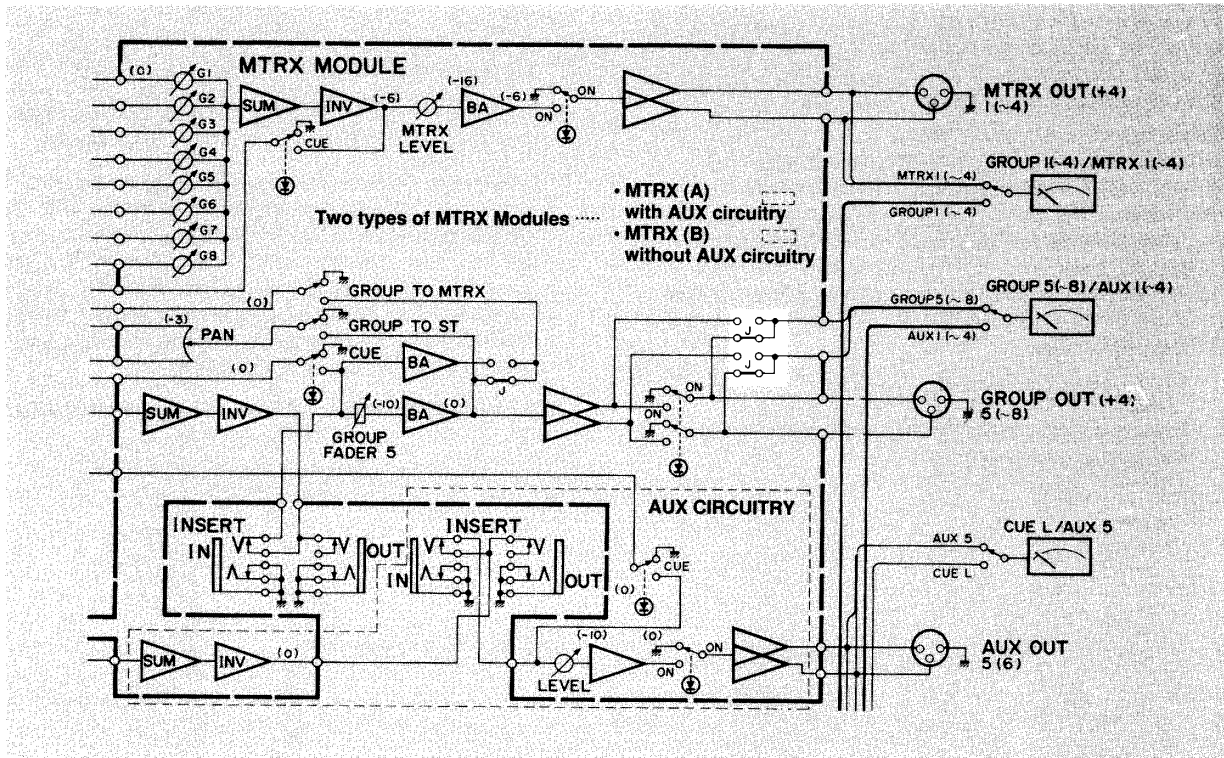


FIGURE 6-5-B.

6.6 INSTALLATION OF OPTIONAL INPUT TRANSFORMERS

The PM1800A standard input module is equipped with a balanced, differential input preamplifier for the XLR connector. That preamp, along with some circuitry for the resistive attenuation pads, is located on a small printed circuit board that “piggy back” mounts to the module’s main circuit board. Refer to Figure 6-6.

An optional transformer balancing option may be installed by a Yamaha PM1800A dealer or a qualified electronic service technician. The modification kit contains a replacement circuit board for the original differential preamplifier, and a separate input transformer. In order to install the kit, the following steps must be performed.

1. Shut off the power to the console.
2. Remove any input module(s) to be converted from the console mainframe.
3. Hold the module with the fader to the left, and lay the module on its side, controls facing away from you.
4. Remove the “IN1 4/4” board.
5. Install the new board (that comes wired to the transformer) in place of the “IN1 4/4” board.
6. Install the transformer by securing its bracket to the lower left edge of the module frame with the screw provided. Dress the cable that joins the transformer and its circuit board neatly. You may wish to tie it to the board so that after the module is reinstalled, the cable does not become pinched between modules or between the module and mainframe. Refer to Figure 6-6.
7. Reinstall the input module into the mainframe.

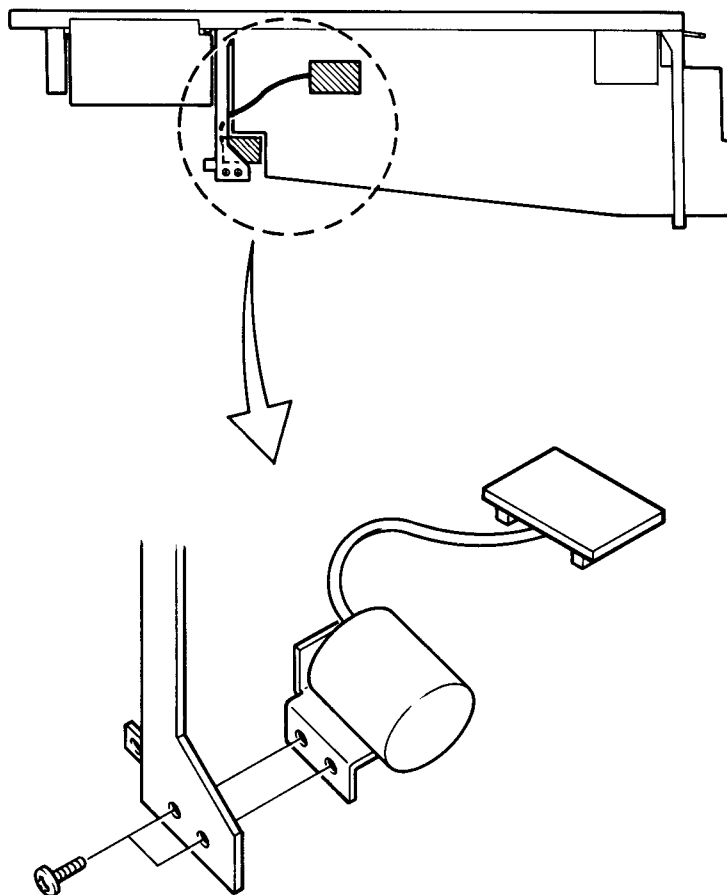


FIGURE 6-6. OPTIONAL INPUT TRANSFORMER INSTALLATION. (IT1800A)

6.7 HINTS ON CIRCUITRY FOR REMOTE CONTROL OF THE MUTE GROUPS

The MUTE CONTROL connector on the PM1800A rear panel is provided primarily so that two consoles may be linked, and just one console's MUTE MASTER switches will affect both console's input channels. However, it is possible to create an independent controller so that this function can be remoted from the console. One possible application would be the creation of a limited automation system. Yamaha does not offer detailed instructions for this type of remote control. However, we do present here information which will allow a competent technician to do the job.

The MUTE connections are illustrated in Figure 2-9. In order to mute a group, simply ground the conductor corresponding to that group. Naturally, the console's MUTE MASTER/SLAVE switch must be set to the SLAVE position in order for the corresponding remote control to take effect. The MUTE connections not only permit remote muting by an external switching system, they also permit two PM1800A consoles, or a PM1800A and PM3000 to be interconnected so that one console's master mute switches can control both console's muting functions.

WARNING:

Only qualified service technicians should attempt to construct and connect any circuit to interface with the PM1800A MUTE CONTROL connector. A circuit or wiring error could damage the console, and such damage is not covered under the terms of the PM1800A Warranty. Improper grounding could also create noise and/or safety hazards. This information is provided only to illustrate the extent of such a modification; the PM1800A Service Manual should be consulted before actually building any remote control device.

Operating Notes and Hints

This section is not meant to be comprehensive. Instead, it focuses on a few areas which we feel require special attention, or where a better understanding of the function can lead to far more utility or better sound quality from the PM1800A.

7.1 CONSOLE GAIN STRUCTURE

In the GAIN STRUCTURE AND LEVELS section of this manual, we discuss some general considerations regarding levels and system setup. What of the proper gain structure within the PM1800A? How can the many faders and other level controls that affect a given signal all be adjusted for the optimum results? These are important questions to ponder, and we hope you will take some time to study the possibilities.

7.1.1 What is the Proper Gain Structure?

Let's begin with the XLR channel input to the console. According to the INPUT CHARACTERISTICS chart in the SPECIFICATIONS section, the nominal input level ranges from -60 dBu (0.75 mV) to $+4$ dBu (1.23 V). These are the levels that will supply the ideal signal level throughout the module with the PAD set to 0 dB or -40 dB, the input GAIN control as required, AND fader set to its nominal position. Actually, a wider range of levels can be accommodated if the GAIN control and fader are further adjusted; from -80 dBu (0.075 mV) minimum to $+24$ dBu (12.3V) maximum.

What is the correct gain structure? Simply stated, it is the level at which there remains adequate headroom so that peaks can be accommodated without clipping, while at the same time there is sufficient "distance" above the noise floor that noise does not become objectionable. If a signal is too high in level (too "hot") at a given point in the console, then peaks or, in the extreme, the entire signal, will be subject to distortion. If the signal is too low in level, there may be considerably more headroom and less risk of distortion, but the noise will be that much more noticeable, and quiet passages may be masked entirely by residual noise. The "ideal" level, then, where headroom and noise tradeoffs are optimum, is also known as the nominal level. There is no single value for the correct nominal level; it varies throughout the console. This is what the middle graph line in the GAIN STRUCTURE chart in Figure 3-22 depicts. The top graph line indicates the clipping point. The distance between these two lines, at any point along the horizontal signal flow scale, depicts the available headroom. It is important that wide headroom be available throughout a console, not just at the input and output; otherwise multiple signals applied to the busses may add together such that the mixed level approaches clipping, even though the individual feeds to the mix are within their acceptable nominal range. Sometimes a group or master fader can be adjusted to correct this condition, other times it cannot because the distortion is occurring in an amplifier ahead of the fader, and the only cure is to lower the signal levels applied to the bus. How can one know the best course of action when distortion, or excess noise, is encountered?

7.1.2 What Affects Gain Structure?

First, understand that signal levels can be increased by either increasing amplifier gain (including EQ boost),

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

7.1.3 Establishing the Correct Input Channel Settings

In the case of the PM1800A, the pair CLIP LED adjacent to the input channel PAD and GAIN controls helps you to set the correct gain structure at the input stage. Begin with the PAD set at maximum attenuation (-40 dB), the GAIN control centered, and apply the typical input signal to the channel input. Adjust the GAIN control until the red CLIP LED turns on occasionally. If you cannot get the CLIP LED to turn on at all, or hardly at all, then leave the GAIN all the way up, and move the PAD switch one notch to a lower attenuation value and try again. If the CLIP LED is then on most or all of the time, turn down the GAIN. If it still doesn't turn on at all, or much, switch to the minimum attenuation (0 dB) PAD setting, and adjust the GAIN so the CLIP LED flashes on occasionally during the louder program peaks. This establishes the correct channel sensitivity for the initial setup (you may wish to alter these values during an actual program mix, as explained in subsequent paragraphs).

NOTE:

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

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

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

The signal now may be assigned to any of the eight group mixing busses and to any of the six auxiliary mixing busses. If an aux send is operated in a PRE-fader mode (switch selectable for AUX 1 and 2, and factory prewired this way in AUX 3 - 6, though this can be altered), then the signal level applied to that bus will remain constant regardless of adjustments to the channel Fader, depending instead only on the AUX control setting. In POST-fader mode, the send level will be de-

terminated by both the channel AUX control and the channel Fader.

This same procedure should now be followed for each input channel. Once this is done, the bus levels can be examined. Set the VU meter assign switches to look at the GROUP levels and the AUX OUT levels (you can see STEREO OUT levels all the time, with no switching). One bus at a time, monitor the group mix (use the headphones and the group CUE switch), and create a rough mix of all input channels which feed this group. Bring down the input Faders for those sources which are too prominent in the mix; avoid raising input Faders to make other sources more prominent. Once this rough mix is established, raise the corresponding Group Master Fader toward the nominal position (0 dB on the scale). If the signal level on any of these busses becomes too hot (red LED flashing on more than occasionally or VU meter pegged at the top of the scale), do not back off the Group Master Fader. Instead, pull down all the input channel Faders which feed this Group by an equal amount. This will leave the Group Master Fader at the desired nominal position, will preserve the desired balance between input channels, and will keep the bus level from being too hot. Finally, release the Group CUE switch.

7.1.4 Establishing the Correct Group Master Settings

Follow the same procedure for each of the other Group Masters. Once all Group Masters are calibrated in this manner, the Stereo mix and Master Fader can be similarly calibrated. Any Group outputs which are to be applied to the stereo mix should be so assigned. Monitor the stereo mix by engaging the Stereo CUE switch, and adjust the various stereo PAN pots as desired. Set the Stereo Master BALANCE control [57] to center. If you're not sure about the stereo position of a given input source, you can temporarily place the console in the SOLO mode, then press its CUE/SOLO switch, and you will hear only that source so you can more accurately adjust its position in the stereo field. With the various signals applied to the stereo mix, bring up the Stereo Master Fader to nominal position and check the bus levels on the L and R VU meters; if they are too high, you can lower all Group Faders; bring down all these faders by the same amount so as to preserve the mix balance.

7.1.5 Establishing the Correct Aux Send Master Settings

It is now appropriate to adjust the AUX Send Master controls. You will not alter the input channel Fader settings, in this case, but instead will adjust all AUX controls on all the inputs that feed a given aux bus to obtain the optimum mix. Monitor that bus mix with the corresponding Aux Send CUE switch [34], and then bring up the associated AUX Send Master to nominal level (the pointer mark on the control scale). If the AUX VU meter and/or PEAK LED indicate the bus level is too high, back off on all the correspondingly numbered input channel AUX controls, not the AUX Send Master. Release this Aux CUE switch, and go on to repeat the same procedure for each of the AUX Sends. Remember to switch the AUX meters so they are monitoring the busses which are being calibrated.

7.1.6 Establishing the Correct Mix Matrix Settings

Since the matrix is fed from the group busses, its gains should be adjusted only after the Group Master levels have been calibrated. (It makes little difference whether the GROUP-TO-MATRIX send is pre or post Group Fader, which is changeable via internal jumpers; the Group bus calibration must still be done first to establish the proper levels on the group busses ahead of the Group Masters.

Here, a similar approach can be used, monitoring the matrix outputs one at a time with the Matrix CUE switch, adjusting any individual matrix controls you wish to include in that matrix mix first to the nominal setting (pointer mark), then reducing the setting of some of these controls to obtain the desired mix, and finally bringing up the MATRIX MASTER control to nominal position (pointer mark) and, if necessary, reducing the contributing matrix mix controls by an equal amount to avoid too-high bus levels.

7.1.7 Establishing the Correct Aux Return Settings

With the aux sends calibrated, any external signal processors (effects units such as reverbs, delay lines, phasers, etc.) which are fed from the aux system can be adjusted for optimum input and output levels. Assuming the auxiliary processors are brought into one or more of the PM1800A AUX RTN inputs, those returns are ready to be calibrated. The CUE switch for the Aux Return is of little value here because it derives signal ahead of the Aux Return LEVEL control. Instead, monitor any bus(es) to which the AUX RTN is assigned, and set the return LEVEL control for the desired amount of return signal. If the LEVEL control is at the maximum (#10), then the signal applied to the AUX RTN input is too low in level, and the output of the auxiliary processor should be increased. If the LEVEL control is below 1/3 rotation (about #3), then the output of the auxiliary processor should be attenuated somewhat so that the PM1800A Aux Return LEVEL can be raised closer to nominal (pointer mark). Once all AUX RTN inputs have been so calibrated, it is possible that their additional signal contribution to any assigned busses may have raised the overall bus level too high. Again check the VU meters on affected GROUP or AUX busses. In this case, the bus Masters may be used for minor "touch up" level adjustments. If the level is much too high on a given bus, do not pull down its Master more than a few dB; instead, lower the Faders or Level controls for all signals which contribute to that bus.

7.1.8 Channel Muting and Gain Structure

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

7.2 FURTHER HINTS & CONCEPTUAL NOTES

7.2.1 Understanding and Using the Mix Matrix

The PM1800A Mix Matrix consists of 8 smaller mix level controls [42] and one larger MATRIX MASTER control [43] on each of four Master Modules. These 36 controls can be thought of as a small mixer within the larger console. In general, the matrix is used to create different output mixes from the same set of mixing

busses. The matrix is considerably more convenient and less costly than actually using an external line mixer, and in the case of the PM1800A, it is more flexible as well.

Let's "walk through" the PM1800A mix matrix. Each matrix "channel" (a vertical row of controls) is identical. All eight Group busses are mixed to a mono signal using the individual matrix mix level controls. The overall level of the mix of these 8 sources can be adjusted with the MATRIX MASTER control.

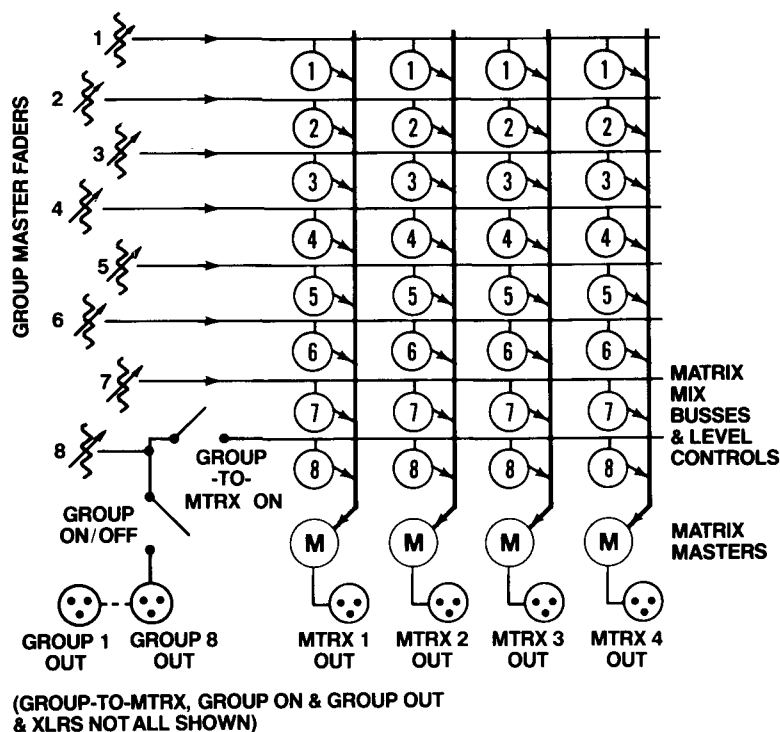


FIGURE 7-1. BLOCK DIAGRAM OF PM1800A MIX MATRIX.

If you examine the block diagram of the matrix provided in Figure 7-1, you will see that the level adjustments made in one channel of the matrix affect only that matrix output. They do not affect levels in any other matrix channel, nor do they affect any other console outputs. On the other hand, assuming the signals are fed to the matrix after the Group Master Faders [40] (which is how the PM1800A is supplied from the factory), then adjustments of the Group bus output levels will affect the levels applied to the matrix.

NOTE:

The signal fed from each Group bus to the matrix is factory wired so that it is derived after the Group Fader. A jumper wire in each Master Module may be reset so that the feed to the matrix is derived ahead of the Group Fader (see section 6.4). In that case, the Group Fader setting would not affect the matrix levels.

The four matrix channels can be used to create four different 8:1 mono mixes, or they may be used to create two different 8:2 stereo mixes, or a combination of mono and stereo mixes. These multiple mixes can be used for a variety of purposes, depending on the application.

In live sound reinforcement, instead of feeding the house sound system directly from the Group outputs [91], or the Stereo output [96], the sound system can be fed from the Matrix outputs [92]. In this case, the Group busses are used for mixing sub-groups of different sources; i.e., brass, drum/percussion, lead vocals, backup vocals, rhythm guitars & bass, lead guitar, keyboards (in stereo), and so forth. The Group Faders then control the overall level of each sub-group of input channels. The matrix channels can be used to create two stereo or four mono mixes from those groups. The mix matrix outputs then feed the power amps and speakers for various zones in the main house, as well as other areas (dressing rooms, lobby, remote feeds, etc.)

The advantages to this approach are numerous. For example, if the brass level is too high in all outputs, only one Group Fader need be adjusted (for the brass sub-group). On the other hand, if there is too much vocal near the front of the audience (due to spill from the vocal stage monitors), you can adjust the one matrix mix level control, corresponding to the vocal Group, in the matrix channel that feeds the near stage house speakers. Similarly, if your system is designed with larger speakers near the front of the house, having better low frequency output than the rear fill speakers, then those speakers should be fed the bass-heavy instruments. By adjusting the matrix mix level controls for the drum/percussion and bass guitar Groups so that more of these subgroups goes to the matrix outputs that feed the near-stage speakers, and less to the rear fill speakers, the overall sound quality in the house will be improved.

For program fades, you have a choice: you can use the Group Master Faders, in which case the previously established balance for each zone of the sound system reappears as soon as these Faders are returned to their correct settings. Or you can use the MATRIX MASTER controls, in which case the previously established program (group) balance remains, but you'll have to recreate the zone-to-zone balance when you bring up

the MATRIX MASTER controls. Of course, you can always use the Group ON/off switches [39] or Matrix ON/off switches [45] to mute the output to the speaker system, thereby eliminating any uncertainty in re-establishing program levels.

If the PM1800A internal jumpers are arranged so that the Group-to-Matrix feeds are derived pre-fader (as described in Section 6.4), then the Group Master Faders *will not* affect the matrix mix levels. In this case, the matrix can be used in much the same way, to create the necessary mono or stereo house feeds, while the group outputs can be mixed independently to feed a multitrack tape recorder. (The stereo output is also useable here, though it may or may not have a useful program balance since the group levels that feed it are optimized for the matrix, not for a specific mix.) Whereas the signals applied to tape are generally recorded at a uniformly "hot" level (high enough to optimize signal-to-noise ratio, and just low enough to avoid saturation), the same group signals can be mixed to achieve the desired program balance for the live sound presentation.

7.2.2 Understanding and Use of the Master Mute Function.

Each input channel is provided with eight MUTE Assign switches [22]. When one of these switches is engaged on a given input channel, that channel becomes subject to control by the correspondingly numbered MUTE MASTER switch [71]. Specifically, when the MUTE MASTER switch is engaged, then the assigned input channel(s) turn Off (assuming they had been turned On in the first place). What this means is that any assortment of input channels can be pre-set to turn off when one or more of the MUTE MASTER switches is engaged (or to turn on when the MUTE MASTER switch is released). This is useful in just about every conceivable application.

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

For a theatrical presentation, different scenes can be unmuted as required, keeping the number of open mics at a minimum, which reduces the tendency for feedback with distant mics in a live sound reinforcement system. For recording, a group of inputs which are primarily used for solo performances can be kept muted until the moment they are needed, thus minimizing noise. For a church, the choir mics can be kept muted until the moment the choir is called upon, thus reducing noise, the "hollow" sound from those open mics, and removing the extra stress on the choir members of having to keep absolutely still during the entire service. These are but a few of the ways that the PM1800A's ability to mute overlapping groups of input channels can be used to advantage.

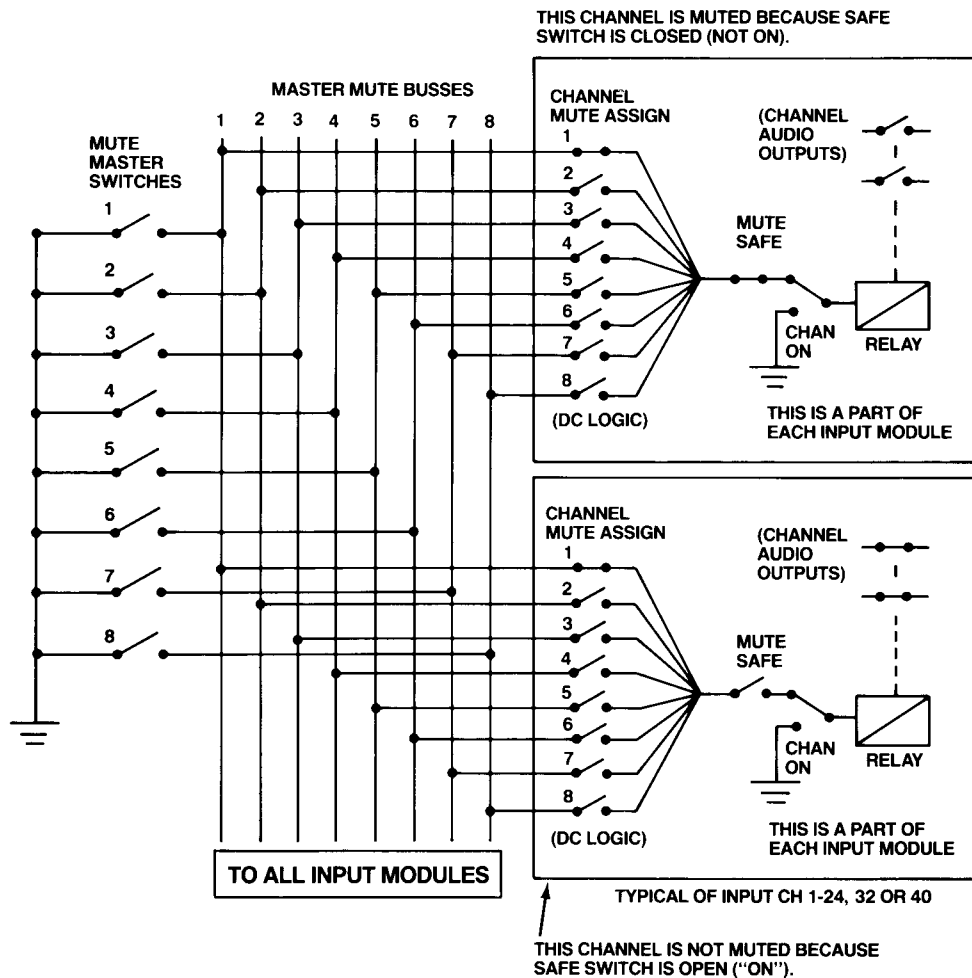


FIGURE 7-2. BLOCK DIAGRAM OF PM1800A MASTER MUTE SYSTEM.

NOTE:

While a similar function could be achieved by using the Group ON/off switches, the functions are really different. Consider that the MUTE MASTER switch kills all the output of the channels, including the aux sends, whereas each Group ON/off switch kills only one group output. Also, consider that some input channels feeding a given group can be killed with one MUTE MASTER, while other input channels may continue feeding that group output. Thus, the mute function is distinctly different than the Group output ON/off switches.

Things can become more complex when an input channel is assigned to more than one MUTE MASTER switch. In this case, the mere act of releasing one MUTE MASTER may not turn on the channel... if the channel is still being muted by so much as one other assigned MUTE MASTER. Should the need arise to turn on a particular input channel without unmuting other channels, and you don't want to disturb the previously assigned MUTE switches, you can override the entire muting system by engaging that channel's MUTE SAFE switch [20]. MUTE SAFE, in effect, blocks any of the channel's MUTE ASSIGN switches [22] so that the

channel will be on so long as its ON/off switch [19] is engaged.

7.2.3 Stereo Panning to the Eight Group Mixing Buses

The input channel bus assignment is very flexible. If the PAN switch [2] is ON, then the PAN pot will place the channel output between any odd-numbered and even-numbered group mixing busses (based on those assign switches [1] which are actually engaged). (When the PAN switch is OFF, the assign switches go direct to each bus and the PAN pot has no effect.) Why would one want to utilize stereo panning between odd and even numbered group busses?

There are instances when more than one stereo mix will be required. In such applications, pairs of group mixing busses can be used as though they were additional stereo mixing busses. For example, suppose a house mix is being done in stereo. Let's say the drums are being mixed in stereo, and must be compressed as a group. One does not want the drum compression to affect the other program, so compressing the stereo output is not the answer. Therefore, the drum input channels can be assigned to a pair of odd and even numbered group busses, and that stereo mix created

with the input PAN pots of the appropriate channels. The INSERT IN/OUT jacks of those two group busses are then patched to a stereo compressor/limiter, such as the Yamaha GC2020B, which affects only the stereo drum mix. The two groups are then mixed together into the main house mix, which is done on the remaining six groups, by engaging their GROUP-TO-STereo switches [38], and panning one fully left and the other fully right with the Group PAN pots [37]. Using this approach, up to 4 group-generated stereo mixes can be processed independently of each other, then mixed into a single stereo program via the GROUP-TO-ST switches. Alternately, the separate stereo programs can be used for completely different purposes and never mixed together (one for a recording feed, one for the house, etc.)

The input PAN switches have a “7-8” position which permits only busses 7 and 8 to be used for a stereo mix (as described above) while the other six groups bypass the PAN pot and can therefore be assigned directly to the Group Faders. This permits a happy combination of both local (input generated) and independent group-generated stereo mixes, or a local stereo mix and a group-to-matrix feed for a completely different output arrangement.

7.3 INTERFACE WITH POPULAR INTERCOM SYSTEMS

In the Introduction to this manual, and in the “Brief Operating Instructions” in Section 2, we mentioned that the PM1800A can be tied to popular intercom systems so that the console operator need wear just one headset for cue and intercom. Intercom systems manufactured by RTS Systems and by Clear-Com utilize different types of connections, and each of these manufacturers is sufficiently popular to be considered a “standard.” The following instructions are provided to facilitate interface with these systems, and with other systems using the same type of connections.

7.3.1 RTS Intercom Systems

RTS utilizes a 3-wire interconnect between their standard intercom stations. They also use standard XLR-3 connectors in most cases so that ordinary microphone cables may be employed. (In some cases, XLR-4 connectors are used). However, DC power is superimposed on the audio signal within the RTS intercom system, and there are actually two audio channels carried in the 3-wire cable, so one cannot necessarily simply plug a standard mic cable between the PM1800A and the RTS intercom (except when plugging the PM1800A Talkback Output into an RTS Program Input). Instead, an adaptor may be necessary.

CAUTION:

Do not connect a standard XLR-3 cable between the PM1800A and the RTS (or similar) intercom system, except as noted above. It is essential that the proper isolation be provided, as explained below. Failure to observe this precaution may result in damage to the console and/or the intercom system. Besides, it simply won't work right.

To apply signal to the PM1800A COMM IN, signal can be derived from any RTS intercom XLR-3 line. Pin 2 of the RTS system carries the audio for channel 1, along with 32 volts DC to power the intercom units. Pin 3 carries the audio for channel 2—it may or may not carry DC power. Pin 1 serves as the DC return and audio common. In order to block the DC from reaching the PM1800A, a 22 microfarad, 50 volt capacitor should be installed in series with the line. As a precaution, this capacitor (or a second one) should be left in the line for channel 2. In fact, as a further precaution, capacitors should be installed on both primary and both secondary leads to the transformer, as this avoids a problem in the event that the adaptor is misconnected. The nominal line impedance for the RTS intercom system is 200 ohms, and nominal level is -7 dBu. In order to optimize the interface, RTS recommends the use of a 1:1 turns ratio (600 ohm) transformer (Bourns Model LM9003). This transformer has center taps which may be used to better match the -7 dBu RTS line level to the $+4$ dBu PM1800A line level. Of course, the PM1800A COMM IN is equipped with a gain control, so it is probably not necessary to use a center tapped transformer. It may not be necessary to use the transformer at all, but be sure to use the isolation capacitors.

Assuming you do follow RTS' recommendation, we suggest using a mini-box to house the transformer and capacitor, equipping the box with clearly labeled XLR-3 connectors. If you use only one protection capacitor, then be sure you DO NOT SWAP INPUT FOR OUTPUT, as the capacitor protection will be lost, and the transformer will take the DC across its winding. Only one intercom channel can be fed to the console. The accompanying illustration depicts a jumper or switch to select the active channel, although if a “fixed” choice is made, then the appropriate conductor for channel 1 or 2 can be hard wired.

The PM1800A COMM IN Pad [83] should be set to “+4” position, and the LEVEL control [62] may be adjusted as required. You can obtain additional gain by wiring the transformer primary so that the audio from the intercom goes to the center tap (terminal “B” in Figure 7-3) instead of the high side of the winding (terminal “A”).

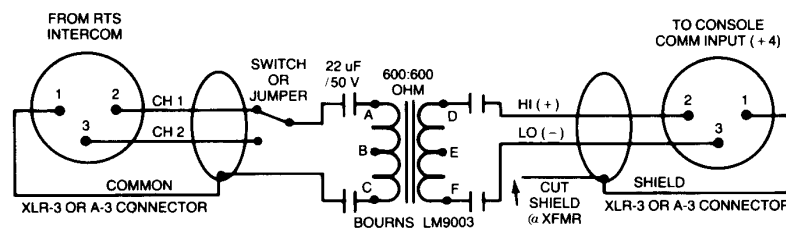


FIGURE 7-3. INTERFACE OF RTS INTERCOM TO PM1800A “COMM IN.”

There are a couple of ways to communicate from the PM1800A to the intercom system. One may connect a standard 3-pin XLR cable from the console's TB OUT connector to the intercom's IFB (Interruptible Foldback) program input. This is the most simple, direct method, and no adaptor is needed. However, it may be that other program is being fed to the IFB system, so it is possible to inject the talkback signal directly into the intercom audio line on channel 1 or channel 2 of the RTS system (also on channel 3 in the larger RTS systems). This is done using the audio coupling input on the intercom power supply. No blocking capacitors are necessary in this case because the coupling input is DC isolated within the power supply. However, RTS still recommends use of a 1:1 600 ohm transformer (again, the Bourns LM9003 or equivalent). In addition, RTS recommends placing a 620 ohm, 1/4 watt resistor in series with the circuit from transformer secondary to the intercom... ostensibly to drop the nominal +4 dBu console output level so it is closer to the -7 dBu nominal intercom line level. Actually, RTS also suggests using a 1K audio taper pot for further signal level adjustments,

but since the PM1800A has a TB LEVEL control, we have omitted the adjustment in the circuit shown in Figure 7-4.

The larger RTS power supplies (Model PS-30 or PS-31) are equipped with a 4-pin XLR connector for the audio coupling input. This connector is wired so that pin 1 carries the audio signal for channel 1, pin 2 for channel 2, pin 3 for channel 3, and pin 4 is the audio common. The interconnect is shown in Figure 7-4A. The smaller RTS power supply (Model PS-8) is equipped with a 1/4" Tip/Ring/Sleeve phone jack for the audio coupling input. This jack is wired with the tip for channel 1, the ring for channel 2, and the sleeve for audio common. The interconnect is shown in Figure 7-4B. In both instances, you have the choice of using a jumper, a switch, or hard wiring the connection to one or the other intercom audio channel. If the signal level is too "hot" for the intercom system, you can turn down the TB LEVEL in the console, or you can wire the secondary of the transformer so that signal is derived from the center tap (terminal "E" in Figure 7-4) rather than the high side of the transformer (terminal "D").

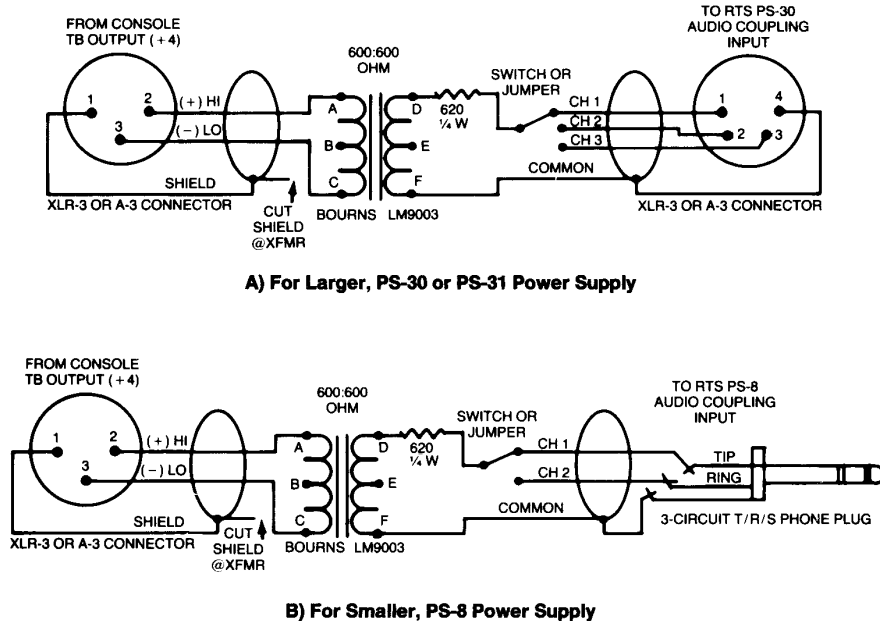


FIGURE 7-4. INTERFACE OF PM1800A "TB OUT" TO RTS INTERCOM "AUDIO COUPLING" INPUT.

For additional information, contact:
 RTS Systems, Inc.
 1100 West Chestnut Street
 Burbank, CA 91506, U.S.A.
 Phone (818) 843-7022
 TWX 910-498-4987
 TELEX 194855

7.3.2 Clear-Com Intercom Systems

Like RTS, Clear-Com also utilizes a 3-wire interconnect between their standard intercom stations, with standard XLR-3 connectors so that ordinary microphone cables may be employed. Again, DC power is superimposed on the audio signal within the Clear-Com

intercom system. While there is but one audio channel on the 3-wire cable, the specific wiring is different from RTS systems, and one still cannot plug a standard mic cable between the PM1800A and the Clear-Com intercom. Clear-Com recommends use of their IF4-4 adaptor. This unit separates the combined audio transmit/receive functions of the 3-wire Clear-Com system, and makes them available on a 4-pin XLR connector. The IF4-4 is a 19" rack-mountable unit that occupies a single 1-3/4" high rack space. It provides 600-ohm, transformer-isolated +4 dBu inputs and outputs that are well suited to the PM1800A COMM IN and TB OUT connections. However, an adaptor cable must be fabricated to split the XLR-4 audio connector

on the IF4-4 so it can be patched to the XLR-3 connectors on the PM1800A.

CAUTION:

Do not connect a standard XLR-3 cable between the PM1800A and the Clear-Com (or similar) intercom system. It is essential that the proper isolation be provided. Failure to observe this precaution may result in damage to the console and/or the intercom system. Besides, it simply won't work right.

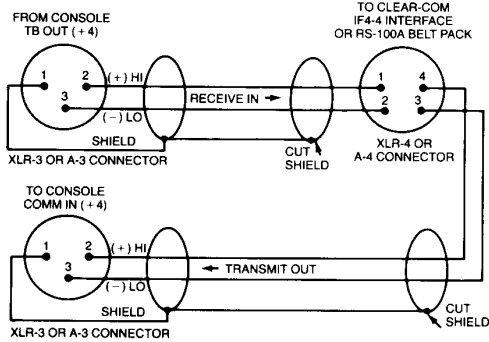


FIGURE 7-5. ADAPTOR CABLE FOR SPLITTING XLR-4 INPUT/OUTPUT CONNECTOR ON A CLEAR-COM IF4-4 INTERCOM INTERFACE, OR ON AN RS-100A BELT PACK. Adaptor required so the IF4-4 or RS-100A can transmit intercom audio to the PM1800A "COMM IN" and receive signal into the intercom audio line from the PM1800A "TB OUT".

A less costly alternative is to use a Clear-Com RS-100A belt pack (or the newer RS-501) as the interface between the console and the intercom system. A custom cable must be wired to link the 4-pin XLR Headset connector on the RS-100A to the PM1800A Comm In and TB Out connectors. This cable is the same as that illustrated in Figure 7-5.

For additional information, contact:
 Clear-Com, Inc.
 1111 17th Street; San Francisco, CA 94107 U.S.A.
 Phone: (415) 861-6666

8.1 GENERAL

The PM1800A is designed primarily for audio mixing in live sound reinforcement applications. Its exceptional flexibility, however, will undoubtedly appeal to those who need a high quality audio mixing console for other applications, including TV show and music video production, AV audio production, and general recording. We explain a few reasons why the PM1800A is well suited to these applications below, but rather than focus on specific end-user applications, we feel it is more important to point out how some of the PM1800A sub-systems can be used to accomplish specific mixing tasks. It is up to you, as the sound engineer or mixing console operator, to best utilize these capabilities in your specific application. This manual is by no means comprehensive, and we expect that many of you will devise unique means to connect and utilize the PM1800A. In fact, Yamaha encourages you to share your special applications with us so that we may, in turn, share the general concepts with other PM1800A users.

8.1.1 Sound Reinforcement

The PM1800A's electronically balanced inputs are of the highest quality, and input transformers can be installed internally where the extra isolation is required. Input channel sensitivity is now broadly adjustable from -80 dBu to -6 dBu by means of a 3-position attenuation pad plus a Gain trim control with 34 dB range, so fader mix settings can be uniformly aligned for faster visual confirmation of the nominal position; there's plenty of gain when it's needed, and noise is minimized when the extra gain is not needed. Four band sweep-type equalization, plus a sweep-frequency high pass filter, facilitate precise shaping of the sound.

Eight group busses can be used to sub-mix various vocal and/or instrumental sections, and these can be remixed to mono or stereo for the house feed by means of either the stereo bus, or the 8x4 Mix Matrix. With another six auxiliary busses, each alterable for pre or post input fader pick-off, there is no shortage of effects sends or foldback (monitor) sends. The four auxiliary returns are each switchable to handle a mono or stereo signal, so the input channels are not "used up" unnecessarily. Stage changes can be aided by the eight Master Mute switches, which, with the touch of a finger, turn on or off assigned groups of input channels.

The PM1800A has other useful features for sound reinforcement, such as: numerous LEDs to display switch status and signal levels with far more reliability than conventional lamps; an all aluminum shell that reduces weight substantially without sacrificing strength; a low profile that blocks fewer seats in the house while providing a good sight line to the stage; an extensive input-priority cue system, plus a solo "in place" mode that mutes other channels for faster setup and faster troubleshooting during sound checks. The console can even be linked to standard intercom systems so the operator doesn't have to wear two sets of headphones.

8.1.2 Theatre

The PM1800A has features that make it ideal for theatrical sound reinforcement. Its eight Master Mute groups, together with the eight Mute assign switches on each input module, enable all the sound sources for

a given scene to be preset so they can be turned on or off at the press of a single switch. Since the console has up to 94 dB of gain, distant microphones and quiet speaking voices will cause no problems.

The console's Mix Matrix can be used as an assignable output mixer. Not unlike a lighting console in concept, the Mix Matrix permits up to 8 sources (the eight group busses) to be remixed into four different output mixes. The matrix outputs can drive various primary speaker systems, effects speaker systems, as well as lobby, dressing room and other remote speakers. The inputs to the matrix can be mixed independently, as required, for each of the areas. If a simultaneous recording is needed, the matrix can be set to mix signals from ahead of the group master faders, so the group and stereo outputs can be used for independent multitrack and two-track tape recording mixes. A Communication input and talkback output facilitate interface to intercom systems.

The 40 input version has a center master, so two operators can work conveniently to handle the show. Its low profile means better sight lines from a high balcony. Its rugged construction means it can travel, reliably, along with the show.

8.1.3 Production

Getting the basics of a soundtrack on tape while you're trying to mix sound for a live show can be a real challenge. The PM1800A simplifies the task by providing independent mix capability for the live sound requirements and the tape recording. You can create 20 different output mixes (eight groups, six aux mixes, a stereo L & R mix, and four matrix mixes). With four aux returns, each of which accepts mono or stereo sources, the input channels are not "used up" just to handle extra effects returns or pre-recorded cues. All channel inputs, aux returns, and bus outputs are balanced, low impedance circuits so long lines can be used without noise; optional transformers are available where the extra margin of grounding isolation and common mode rejection are critical.

Eight group masters and eight Master Mute groups together enable the console operator to more easily "keep track" of the many inputs, switching them on or off, and adjusting their levels at the touch of a finger... precisely on cue. Speaking of which, an extensive cue system, with input priority, enables any output or input to be scrutinized "in place" without affecting the output signals. A solo mode, which mutes all but the selected input, speeds pre-production setup and troubleshooting.

An important feature of the PM1800A for a production environment is the 8x4 mix matrix, a built-in "mixer within a console." In video work, for example, discrete output mixes can be fed to the 8-track tape machine from the group outputs at a suitable level to maintain an ideal S/N ratio while avoiding tape saturation. At the same time, the mix matrix can create working mixes of those groups, with levels adjusted for more "listenable" reference monitoring or foldback. Alternately, some of the aux mix busses can be used for performer cue mixes or foldback, while others can be used for effects sends or to supplement the group mixes when even more tracks must be recorded. After all, eight group outs plus six aux outs equals 14 tracks, and that leaves one spare plus a time code track.

Built-in talkback and communication (intercom link) capability make it easy for the production personnel to coordinate efforts, and the console operator doesn't have to wear two sets of headphones. In fact, the cue output jacks make it possible to monitor the console outputs without any headphones.

8.1.4 Post Production

Once a show has been photographed on video, film or multi-image media, it's time for the crucial post production job of mixing sound effects, music, and/or dialog. Sometimes there is no "original" production soundtrack, and all recording is done in the post production phase, while other times the post production task is primarily one of enhancement. In any case, the PM1800A is well suited to the task. Its many inputs can be switched to handle virtually any input level, from the lowest level mics to very "hot" electric guitars, electric keyboards, and virtually any tape recorder or film chain. Cue switches on just about every input and bus make it possible to check signals "in place" without disrupting the output mixes. Sounds can be precisely tailored, and defects "surgically removed" using the four-band sweep-type equalizers on each input channel, as well as the sweep frequency high pass filters that go as high as 400 Hz. Insert in/out jacks on every input channel and the aux busses make it possible to patch in whatever signal processing is desired. Added convenience is provided by the eight Master Mute groups, which switch assigned input channels on and off instantly.

A mix matrix permits the eight groups to be mixed into four different outputs. This 8x4 matrix, a "mixer within a console," makes it possible to control groups of similar instruments (or vocals) with the group fader, and to then remix those groups. In film work, for example, the mixes might be: left, center, right, surround. If the matrix is fed pre-group fader (internal jumpers permit this) and some input channels PAN switches are in 7-8 position, then Groups 7 & 8 constitute one stereo mix, Groups 1 through 6 can be mixed into two completely independent stereo mixes in the matrix, and the STEREO output can be used as a sort of combined stereo mix. This could be useful, say, for stereo music, stereo dialogue and stereo effects, with a combined stereo reference mix. Cue output jacks, in addition to the headphone output, make monitoring more convenient. There is also a communication input and talkback output that facilitate interface to intercom systems.

8.1.5 Video

In today's rapidly changing video production scene, more live music, more pre-recorded sources, and more special effects are being applied to create soundtracks to which people are paying more attention than ever. Stereo VCRs and stereo TV broadcast will only accelerate the pace of video sound advancement. With its high quality sound and powerful capabilities, the PM1800A is a logical choice for many video sound production requirements. The console's 16, 24, 32 or 40 input channels can handle the numerous mics, instruments and pre-recorded sources for almost any production... and sub inputs allow two consoles to be linked together for that once in a great while when even more inputs are needed.

The PM1800A has eight group busses, so different groups of instruments or mics can be assigned to their own group and controlled with a single fader. As a result of this console's unique PAN function switching, the #7 and #8 Group busses can be used for an independent, direct-assigned mix of the inputs. Alternately, all eight groups can be used for submixes, and assigned to the Stereo fader, which then acts as a "grand master" for the console. The PM1800A also has six auxiliary mixing busses that can be used for effects sends, for headphone cue mixes, or as additional group busses. There's also a Mix Matrix, which can create monitor mixes of the various groups so performers can hear what's happening during the production, while other console outputs simultaneously provide different mixes for recording. Cue output jacks can be used to feed local monitor speakers, and an input priority cue system lets the operator instantly check any input channel or auxiliary return at the touch of a single switch.

With six auxiliary sends, and four aux returns, it's easy to utilize the most sophisticated effects. The aux returns can each be used for a mono or stereo source. If even more returns are needed, input channels may be used (they have the added advantage of four-band sweep-type equalization). Built-in talkback and communication (intercom link) capability make it easy for the producer, director and crew to coordinate efforts, and the console operator doesn't have to wear two sets of headphones.

8.2 SETUP CONCEPTS

8.2.1 Deriving a Stereo Mix from Groups 1 - 8.

There are a number of ways to obtain a stereo mix with this console. One technique is to utilize Groups 1-8 for subgrouping input channels. The post Group Master Fader [40] signals then can be assigned to the stereo mixing bus using the GROUP-TO-ST switches [38] and the Group PAN controls [37]. The Stereo Master Fader [59] then becomes the overall stereo output control for the mixed groups. This is a very straightforward means of achieving a stereo mix (or dual mono output mixes) with subgroup control, and without using the mix matrix.

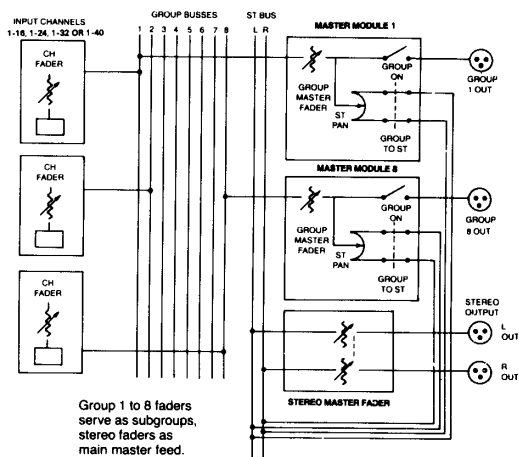


FIGURE 8-1. SYSTEM DIAGRAM WITH GROUPS 1 - 8 AS SUBMASTERS, AND MAIN FEED FROM STEREO MASTERS.

8.2.2 How to Get 4 Independent Stereo Mixes of 8 Mono Mixes (with Submasters) by Using the Stereo Bus, the Mix Matrix and the Input “Pan 7-8” Switches.

This application requires that the console’s internal jumper wiring be altered so that the Group-to-MATRIX feeds are derived pre-Group Fader (see Section 6.4). The first six Group Master Faders [40] may then be assigned to the Stereo Master Fader [59] by engaging their Group-to-ST switches [38]. In this case, the Group Master Faders function as subgroup controls for the overall mixed output controlled by the Stereo Master Faders. These outputs can be used for a stereo program, or for two mono program feeds, depending on the way the Group PAN controls [37] are set. By setting the input channel PAN switches [2] to “7-8” position on certain inputs, those inputs can be panned between busses 7 and 8, and the Group 7 and Group 8 Master faders are then operated together as a stereo master fader for these outputs. (These input channels also may be assigned to other groups, without panning, by simply

engaging their Group Assign #1 through #6 switches [1]). At the same time, other input channels are assigned to the first six Group busses normally, with the PAN switches [2] in the “OFF” or “PAN” mode. These first six groups are then assigned to the mix matrix via the Group-to-Matrix switches [36]. These six groups can then be mixed as required into pairs or individual matrix channels using the #1 to #6 Matrix Mix Level Controls [42] for “subgrouping”, and using the corresponding MATRIX MASTER controls [43] as mono or stereo masters for those mixes (keep the #7 and #8 Matrix Mix controls at zero) Given a total of four matrix channels, this means that two stereo mixes (or four mono mixes) can be created with the matrix. Since these mixes are not affected by the Group or Stereo Master Faders, the four MATRIX OUT connectors [92] plus the two STEREO OUT connectors [96], plus the GROUP 7 and GROUP 8 output connectors [91] can provide a total of four discrete stereo mixes or 8 mono mixes derived from the same eight Group busses.

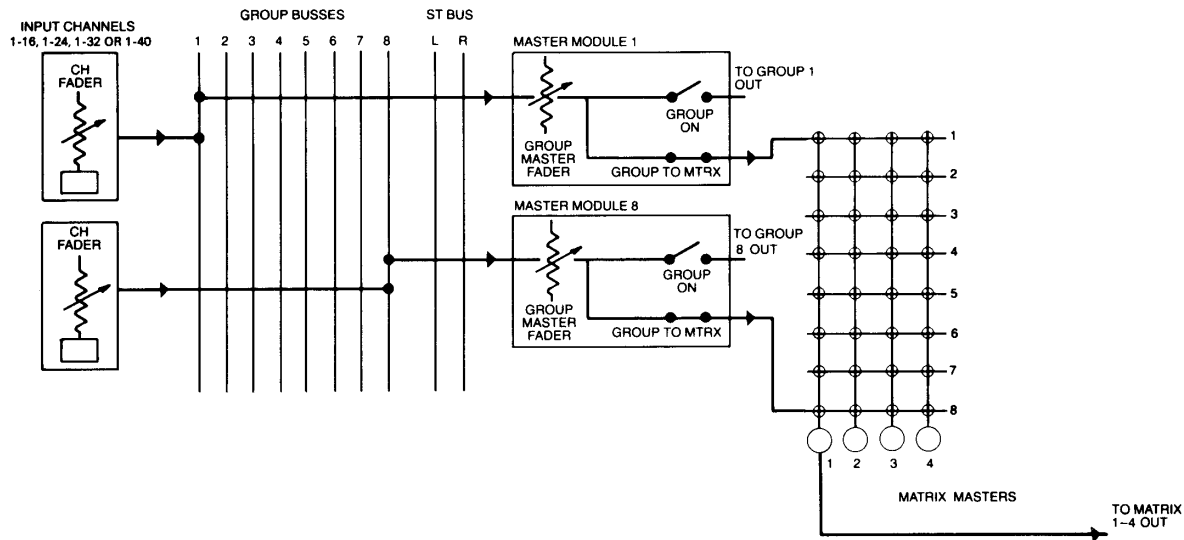


FIGURE 8-2. SYSTEM DIAGRAM FOR 4 INDEPENDENT STEREO OUTPUT MIXES VIA THE STEREO BUS, MIX MATRIX AND GROUP #7 AND #8 OUTPUTS.

9.1 CLEANING THE CONSOLE**9.1.1 The Console and Power Supply Exterior**

The console and power supply are painted with a durable finish. To avoid damage to the paint, control knobs, switch caps and other parts, **DO NOT USE SOLVENTS**. Instead, keep the console as free of dust as practical. Cover it when not in use, and brush or vacuum it periodically. The surface may be cleaned with a soft rag moistened with a dilute solution of non-abrasive detergent and water. If sticky gum is left on the panel (from masking tape or other tape used for channel labeling), it may be necessary to use a specialized solvent. In general, rubber cement solvent will remove tape residue without harming the console; however, it is your responsibility to test any such solvent in an inconspicuous location to ensure it does not attack the console finish or mar any plastic part.

Avoid getting the inside of the console wet from excessively wet rags. **DO NOT USE AEROSOL OR SPRAY CLEANERS**.

9.1.2 Power Supply Air Filter

The reticulated foam air filter on the front of the PW1800 power supply screens cooling air as it is drawn through the unit. When the foam becomes clogged or dirty, it should be cleaned; check it periodically. Using a 3 mm allen wrench, remove the four cap screws that secure the front grille. The foam element may now be removed and rinsed in cool water. For greasy or stubborn dirt, dip the element in a mild solution of detergent and water, then rinse with clear water. Blot and/or air dry the element thoroughly before returning it to the power supply. **DO NOT USE SOLVENTS TO CLEAN THE FOAM ELEMENT**.

9.1.3 Pots and Faders

Yamaha **DOES NOT** recommend the routine use of any contact cleaners or solvents for cleaning pots or faders. Such "preventive maintenance" can actually do more harm than good by removing the lubricating film on certain pots or faders. While treatment with such solvents or cleaners may temporarily "clean up" a noisy control, it can also quickly result in a worn element (due to lack of lubrication) and even greater, incurable noise.

When a component is to be cleaned, use a very small amount of an appropriate cleaner, solvent, or pure isopropyl alcohol. Try to get it on the element, and immediately work the pot or fader several times all the way between stops.

One safe practice that may reduce dirt and dust build up is to rapidly move each control on the console through its full range of travel. Don't "slam" the faders or rotary knobs against their stops, just move them all the way back and forth. In general, cleaning pots and faders is not a trivial task. Some have carbon elements, some have conductive plastic elements, and others have cermet elements. What cleans one part reliably may not work on another. When in doubt, consult your authorized Yamaha PM1800A dealer or service center.

9.1.4 The Console Interior

Dust and dirt are the enemy of electronic and mechanical systems. Switches and controls may wear prematurely due to the abrasive nature of dirt. A coating of dust may, in some cases, be conductive and change the electrical properties of the circuit. Similarly, dirt accumulations can reduce the thermal dissipation from heat sinks and transistors, leading to premature failure. It is advisable to use a soft brush or a vacuum cleaner with a soft brush attachment to clean the console periodically. Depending on the environment, this may be as often as once a month, or as infrequently as once a year. Use care not to bend or dislodge any components. Always do this work with the console power **OFF**.

If a beverage is spilled into the console, try to blot up as much excess moisture as possible immediately. If practical, immediately turn off the power and remove any affected modules. If not, wait until it is practical, and then turn off the power and proceed. Rinse contaminated parts on the module with distilled water, shake off the excess water, blot dry with a soft cloth, and air dry or use a warm (not hot) stream of air from a hair dryer to facilitate drying. If the console interior is contaminated, wipe it clean with a water-moistened cloth.

It is best to clean a spill as soon as possible. Unsweetened black coffee is probably the least harmful. The sugar in sweetened coffee can leave a sticky film on parts, and cream or milk will leave a residue that can be very troublesome. Similarly, sweetened soft drinks and fruit juices can leave sticky residues that degrade the performance of switches, faders and pots.

MODULE REMOVAL AND REPLACEMENT
(see OPTIONAL FUNCTIONS, Section 6.1)

9.2 METER LAMP REPLACEMENT

Two lamps illuminate the face of each VU meter.
To change a meter lamp, first open the meter bridge.
This is done by removing 2 screws from the side and
four screws from the rear panel behind the meter ①

bridge. The bridge is hinged on the rear panel, and can
be swung open for access to the meters ②.

Each replacement lamp (Yamaha part number 75570)
comes with a connector affixed to pigtail leads from the
lamp ③. Withdraw the old lamp from the rear, pulling
it out of its retaining grommet in the meter face, and
unplug the connector from the rear of the meter as-
sembly ④. Insert the new lamp in its place, and secure
the connector.

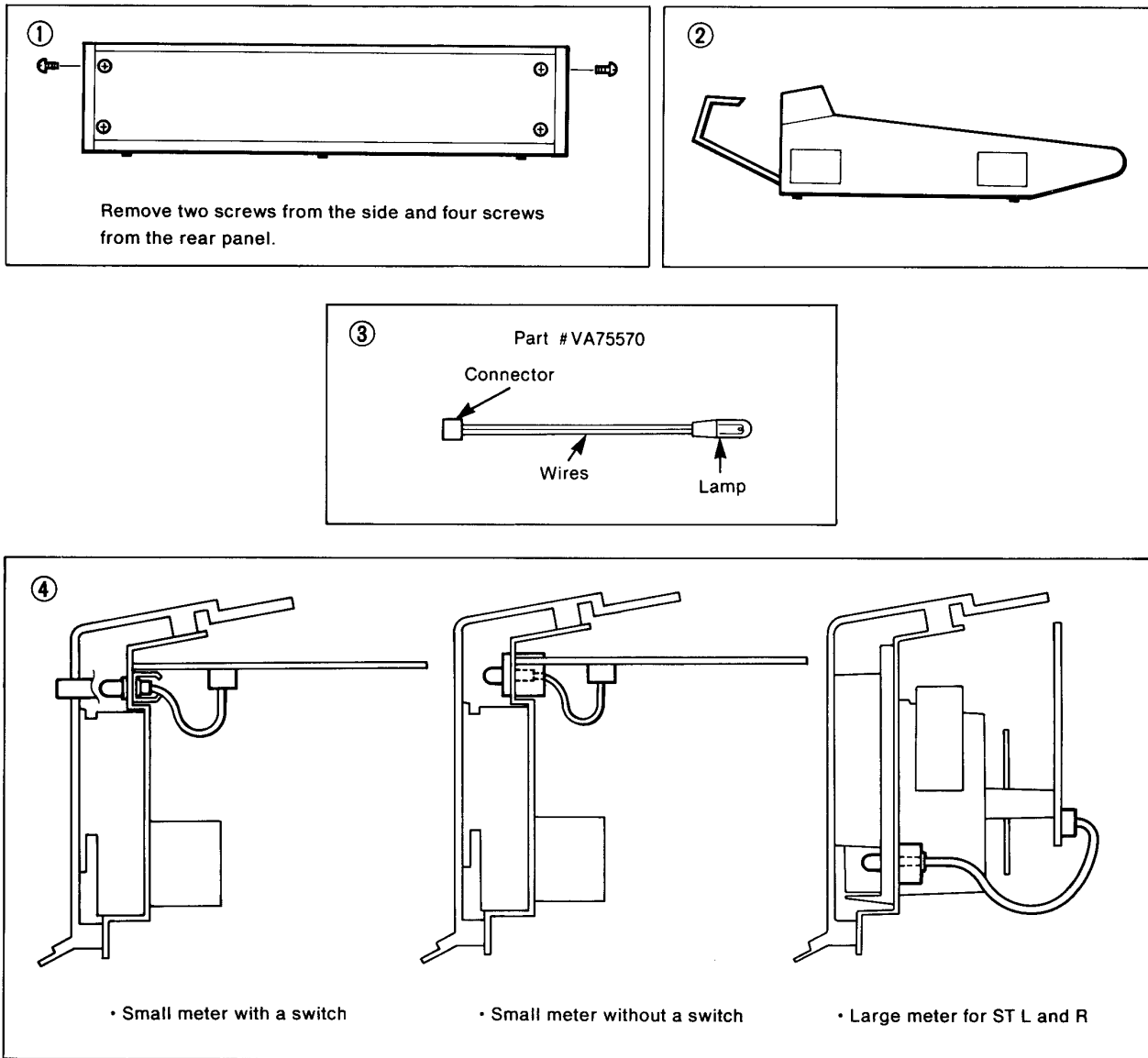


FIGURE 9-1. REPLACEMENT OF VU METER LAMPS.

9.3 WHERE TO CHECK IF THERE IS NO OUTPUT

In general, when something appears not to be working properly in a sound system, it is necessary to have a clear understanding of the system block diagram. One should look for a "good" signal by patching around suspect equipment, modules or circuits. Suspected "bad" cables can be replaced or swapped to see if the problem follows the cable. These techniques should

be known to most experienced sound system operators. In the case of the PM1800A console, however, there are a number of apparent fault conditions, which the operator may inadvertently create simply by setting controls in a particular configuration, whereby no signal reaches the output. The following chart depicts the most likely errors you may encounter, and points out how to correct the problem.

"FAULT" CONDITION	POSSIBLE CAUSE	CORRECTION
Input channel signals do not appear at the Group, Stereo, Aux or Matrix outputs	Console is in SOLO mode, and an input channel to which no signal is applied has its CUE/SOLO switch engaged.	Release master SOLO MODE switch to activate all channels which should be on.
	The affected input channel(s) have MUTE assign switches engaged, and the MASTER MUTE group to which the channel(s) is assigned is set to mute mode.	Disengage the MASTER MUTE switch, or the affected input channel MUTE switch(es).
	The affected input channel(s) have MUTE assign switches engaged, and the remote MUTE connection is causing the MASTER MUTE group to be engaged.	Disconnect the MUTE CONTROL connector to check theory; if output is restored, check remote circuitry.
Certain input channels or groups of channels cannot be heard at Group outputs, Group-to-Stereo feeds, or Group-to-MATRIX feeds.	The affected input channels are assigned to a Group Fader which is set to minimum level (down), and the G > ST and G > MATRIX feeds are post Group Fader.	Raise the Group Fader setting to a higher level.
Individual input channel cannot be heard at the Group, Stereo, Aux or Matrix outputs.	Channel ON/off switch is off, or its PAD and GAIN controls are set so input sensitivity is too low.	Turn On the channel. Set the PAD for a lower value and/or GAIN at a higher value.
	A phantom powered condenser microphone or direct box is connected to the channel and is not receiving phantom power.	Check to be sure channel and master 48V switches are on.
There is no output, and no console functions work at all.	Power is not reaching the PM1800A.	Verify that PW1800 is On and that its umbilical cable is properly connected. Check fuse and AC mains voltage.

9.4 WHAT TO DO IN CASE OF TROUBLE

The PM1800A is supported by Yamaha's worldwide network of factory trained and qualified dealer service personnel. In the event of a problem, contact your nearest Yamaha PM1800A dealer. For the name of the nearest dealer, contact one of the Yamaha offices listed below.

YAMAHA CORPORATION

10-1 Nakazawa-Cho
Box 1
Hamamatsu, 430 Japan

Yamaha Europa G.m.b.H.

2084 Rellingen b.
Hamburg, Siemensstr. 22/34
West Germany

Yamaha Musique France S.A.

Rue ambroise croizat,
Parc d'activités de Paris Est,
77200 Torcy-Marne la Vallée,
France

Yamaha-Kemble Music (U.K.) Ltd.

Mount Ave., Bletchley,
Milton Keynes MK1 1JE,
England

Yamaha Scandinavia AB

Fröfästegatan 1,
(Box 300 53)
400 43 Göteborg, Sweden

Yamaha Music Australia

17-33 Market Street,
South Melbourne, 3205,
Australia

Yamaha Canada Music Ltd.

135 Milner Avenue,
Scarborough, Ontario,
M1S 3R1, Canada

Yamaha International Corporation

6600 Orangethorpe Avenue,
Buena Park, Calif.,
90620, U.S.A.

YAMAHA