# 4 MIDI scenes

The scene memories allow automated control over which channels, groups, AUX signals, stereo output, and matrix groups are turned on or off. Each of these busses has individual control, and in addition, the condition of all of these controls can be memorized as a "scene" for later recall.

In addition to the convenience of scene recall from the PM3500, a MIDI controller (or another PM3500 series console connected via MIDI) can be used to change scenes using Program Change commands, or to turn individual busses on and off using Control Change commands.

Scene memories are stored in the console's internal memory, and can be bulk downloaded or uploaded to another PM3500 series console, or sequencer or MIDI Data Filer.

The CHECK function allows previewing of a scene before recall. The check mode also allows the changing of a scene's parameters without affecting the current setup until the operator confirms the changes. This allows the operator to plan ahead and change the entire setup with the press of a button.

A brief description of the keys and switches which control these functions is given in [30] (page 19) through [37] (page 20).

This section describes their operation in more detail.

There are three scene memory modes: NORMAL, CHECK and UTILITY.

## 4.1 Normal mode

The console is in NORMAL mode when neither the UTILITY key [30] (page 19) nor the CHECK key [34] (page 19) is lit. In this mode, the ON/EDIT switches [17] (page 15) on all busses act as ON/OFF switches. Stored scenes can be recalled using the DIRECT RECALL keys, the number keys or the UP/DOWN keys and RECALL key. Current scenes can be stored to memory.

### 4.1.1 Using the DIRECT RECALL keys to recall scenes

The first eight scenes (1 through 8) can be recalled with the appropriate DIRECT RECALL keys [37] (page 20). The scene will be selected, and the appropriate DIRECT RECALL key will light.

If there is no data stored in the scene corresponding to the DIRECT RECALL key which has been pressed, the display will show "`N_dt`" for a short time.

If the console is in CHECK mode (the CHECK key is lit), the console will exit the CHECK mode, will enter NORMAL mode, and the scene will be recalled.

If the UP or DOWN key is pressed while one of the DIRECT RECALL keys is lit, the scene number which is selected by the UP and DOWN keys will flash on the display and may be recalled using the RECALL key.

If one of the number keys is pressed while one of the DIRECT RECALL keys is lit, the display will start to show the number entered. When the number corresponding to the desired scene has been entered, the scene may be recalled using the RECALL key.

Scenes which contain no data will be displayed with a "`/`" following the memory number. If a scene contains no data, but has been edited since the last time it was stored or initialized with no data, a "`?`" will be shown following the scene number.

### 4.1.2 Using the UP and DOWN keys to recall scenes

Use the UP and DOWN keys to change the number on the display until the desired scene number is shown. The number will be shown on the display. Press RECALL to recall the scene.

If the scene whose number is selected contains no data, the number will flash slowly on the display.

If there is no data stored in the scene and you recall it, the display will show "`N_dt`" for a short time.

### 4.1.3 Using the number keys to recall scenes

When a scene is currently recalled, pressing one of the number keys will clear the display, replacing its contents with the digit whose number key has been pressed. Subsequent presses of the key will add digits to the number (which will flash). Digits will be added to the number as long as it is a valid one (for instance, "`1`", "`1`" and "`0`" will display "`110`", but "`2`", "`0`" and "`0`" will simply display "`20`", as "`200`" is not a valid scene memory number.

If the scene memory is the correct one, press RECALL to recall it, otherwise, use the number keys or UP and DOWN keys to correct the scene number before pressing RECALL.

If there is no data stored in the scene corresponding to the number shown, the display will show "`N_dt`". However, if the console is connected to another MIDI device, and Program Change is enabled, the appropriate MIDI Progam Change message will be transmitted.

### 4.1.4 Using MIDI to recall scenes

Make sure that there is a MIDI sender whose MIDI OUT is connected to the MIDI IN of the console.

The console's receive channel should match the transmit channel of the MIDI sender (see "Storing scenes" on page 56).

Sending a Program Change message from the MIDI sender will change scenes on the console. If there is no data stored in the scene corresponding to the number shown, the display will show "N_dt". However, if the PGM utility is set to "LOCL", the Program Change message will be transmitted, even if no data is stored in the scene.

MIDI Program Changes 0 through 127 correspond to scenes 1 through 128 on the console.

### 4.1.5 Storing scenes

Use the ON/EDIT switches of the appropriate busses to make changes to the current scene. The current scene's number will be followed by an "E", indicating that the scene has been edited. If the current scene contains no data, and it has been edited, the display will show the current scene number followed by a "?".

Use the UP and DOWN keys to select the scene memory number into which you want to store the settings.

Alternatively, use the number keypad to select the scene memory number into which you want to store the settings in the same way that it is used to recall scenes (see "Using the number keys to recall scenes" on page 55).

Press the STORE key. The display will flash the word "SURE" and the scene number alternately.

If the memory is protected, (see "Memory Protect" on page 58), the setting cannot be stored, and the display will show "M_pr".

To confirm the store, press STORE again, otherwise, press any other key in the SCENE MEMORY group.

### 4.1.6 Controlling individual channels through MIDI

Make sure that there is a MIDI sender whose MIDI OUT is connected to the MIDI IN of the console.

The console's receive channel should match the transmit channel of the MIDI sender (see "MIDI channel" on page 58).

Sending a Control Change message from the MIDI sender will change the ON/OFF status of the busses on the console. The exact relationship between Control Change messages and bus switches is given on the table on page 59.

This operation is enabled or disabled with the UTILITY function (see "MIDI Control Change" on page 58).

## 4.2 CHECK mode

The console is in CHECK mode whenever the CHECK key is lit. If the CHECK key is not lit, and you want to enter CHECK mode, press the CHECK key once. The exact way of exiting CHECK mode depends on the function, as described below.

### 4.2.1 Recalling scenes in CHECK mode

Enter CHECK mode (press the CHECK key so that it is lit).

Use the UP and DOWN keys to change the number shown on the display. Alternatively, use the number keypad, as described earlier (see "Using the number keys to recall scenes" on page 55). There is one important difference between CHECK mode and EDIT mode: in CHECK mode, the ENTER key should be used to confirm the number being entered. For example, to check memory number 21, press "2", "1", "ENTER".

As the scene number changes on the display, the green CHECK LEDs on each bus will light to show that bus' ON/OFF status in the scene. This gives advance warning of the scene condition, without actually recalling the scene.

To recall the scene being previewed, press the RECALL key.

To exit CHECK mode without recalling the previewed scene, press the CHECK key so that the key is not lit.

### 4.2.2 Pre-setting scenes in CHECK mode

Enter CHECK mode (press the CHECK key so that it is lit).

When in CHECK mode, the ON/EDIT key of the bus toggles the CHECK indicator ON or OFF, giving an indication of what the scene will be when the scene is stored and subsequently recalled.

Select a scene into which you want to store the settings, using the procedures above (see "Recalling scenes in CHECK mode" on page 56), but do not recall it.

Preset the scene using the bus ON/EDIT switches. Note that the sound does not change, and the bus CHECK LEDs toggle, according to the setting of the ON/EDIT switches, which are functioning as EDIT switches in this mode.

When the settings are ready to be stored, press the STORE key. The display will briefly show "SURE" and then flash the number of the scene memory.

If the memory is protected, (see "Memory Protect" on page 58), the setting cannot be stored, and the display will flash "M_pr".

To confirm the store, press STORE again, otherwise, press any other key in the SCENE MEMORY group.

To recall the scene immediately, press the RECALL key. The setting will be recalled (become the current scene), the console will exit CHECK mode (the CHECK key will not be lit), and the green CHECK LEDs will go out, with the corresponding ON LEDs lighting up.

## 4.3 UTILITY mode

The utility mode is active whenever the UTILITY mode key is lit. Note that you cannnot enter the UTILITY mode from CHECK mode (press the CHECK key to turn it off, and then press the UTILITY key).

Repeated presses of the UTILITY key will cycle through the different utilities as shown below. Pressing the UTILITY key after the last utility (MIDI Bulk Dump Request) will exit UTILITY mode (the UTILITY key will not be lit). The UTILITY mode can also be exited be pressing the UTILITY key for a second and then releasing it.

Use the UP and DOWN keys to select the parameter for the operation.

Press STORE to execute the utility function in either of the following cases: changing from stereo to mono or vice versa with the STCU function, or using the Init Bulk Dump Request (RQST) function.

To exit UTILITY mode, press and hold down the UTILITY key for at least one second, then release it. The console will exit UTILITY mode (the UTILITY key will no longer be lit).

### 4.3.1 Program Change messages

These can be transmitted and received (or transmission and reception can be disabled using the PGM utility). Program Change messages will be transmitted when a scene is recalled and can also be received. Transmission and reception will take place over the MIDI channel as set in the MIDI channel (M-CH) utility, or all channels, if OMNI is set on.

### 4.3.2 Control Change messages

These can be transmitted and received (or transmission and reception can be disabled using the CTRL utility). Control Change messages corresponding to the ON keys (see the table on page 59) will be transmitted and received on the MIDI channel as set in the MIDI channel (M-CH) utility, or all channels, if OMNI is set on.

### 4.3.3 System Exclusive messages

System Exclusive messages can always be transmitted on the MIDI channel as set in the MIDI channel (M-CH) utility or all channels, if OMNI is set on.

Bulk Dump Requests will be transmitted and subsequent Bulk Dump datawill be received on the MIDI channel as set in the MIDI channel (M-CH) utility.

After Bulk Dump data has been received, scene memories will be changed according to the data contained in the Bulk Dump.

The format of System Exclusive messages is shown on page 60.

### 4.3.4 Running status

Running status will be cleared if no datahas been received after 300ms or a MIDI reset has been executed.

### 4.3.5 Echo back

When echo back has been selected using the ECHO utility, data received at the MIDI IN terminal will be echoed out through the MIDI OUT terminal, except in the following circumstances:

Bulk Request transmissions, System Exclusive messages more than 1kbyte in length, Active Sensing messages (FEh), and System Common messges (F4h, F5h) undefined by the MIDI Standard.

Since messages are re-transmitted only after all the data of the message has been received, it is possible that some RealTime system messages may have their order transposed. Running Status will also be affected.

Of course, if there is a loop in the MIDI network between the IN and OUT MIDI terminals of the PM3500, data will circulate endlessly around this loop. Take care to avoid this condition.

## 4.3.6 UTILITY mode functions

| Utility | Display | Description | Parameters |
|---|---|---|---|
| **Battery check** | BATT | Allows the checking of the internal backup battery. Displays "LOW" when the battery voltage falls below 2.5V, otherwise shows *.*V where *.* is the voltage | |
| **Memory Protect** | M-PR | Enables/disables the saving of scene data and the reception of MIDI Bulk Dumps. When ON, the memory is protected, and attempting to write scenes to the memory will result in the display showing "M_pr" | ON, OFF |
| **Solo protect** | S-PR | Protects accidental enagement of the SOLO mode with the SOLO switch [26] (page 18). If the SOLO switch is pressed while solo protection is on, the display will show "S_pr". | ON,OFF |
| **AUX and GROUP stereo cueing** | STCU | For this function, the AUX and output groups are grouped into pairs (A12, A34, A56, A78, G12, G34, G56, G78). The required pair is selected using the UP and DOWN keys. If the pair name is displayed followed by a "x", the pair will be cued monaurally. If the pair name is followed by a "o", the pair will be cued in place (stereo). Change between mono and stereo using the STORE key. When the console is shipped, all pairs will be cued monaurally. | Mono (x) or stereo (o) for the selected pair |
| **Memory Initialize** | INIT | Completely clears one or all scene memories. | all, 1 through 128 |
| **MIDI channel** | M-CH | Sets the MIDI channel on which outgoing MIDI data will be transmitted, and incoming MIDI data will be received. | CH1 through CH16 |
| **MIDI Program Change** | PGM | Selects whether MIDI Program Change messages will be sent when scenes on the console are changed, and whether incoming MIDI Program Change messages will change the current scene | LOCL, ON, OFF |
| **MIDI Control Change** | CTRL | Selects whether MIDI Control Changes will be sent when ON/EDIT switches are used, and whether incoming MIDI Control Change messages will turn busses on and off. | ON, OFF |
| **Program Change OMNI** | OMNI | Selects whether incoming MIDI Program Change messages will be recognized from only one channel (as set above) – OMNI OFF, or from all channels (OMNI ON) | ON, OFF |
| **MIDI Echoback** | ECHO | When ON, MIDI THRU signals are added to the MIDI OUT signals. | ON, OFF |
| **MIDI Bulk Out** | BULK | Sends the contents of all or one scene memories as MIDI Bulk Dump data to another console or a MIDI data filing device. | ALL, 1 through 128 |
| **MIDI Bulk Dump Request** | RQST | Sends a "Bulk Dump Request" signal from the MIDI OUT.If another console is connected via MIDI, this will initiate a bulk dump of the specified scene memories. The two consoles should be connected MIDI OUT->MIDI IN and MIDI IN<-MIDI OUT | ALL, 1 through 128 |

The last character of the display will change briefly as follows under the following circumstances: Bulk Dump data is being received and written to memory: "r", Bulk Dump data is being received, but memory protection is on: "p", a Bulk Dump request has been received and/or Bulk Dump data is being transmitted: "s". The appropriate character will be shown for 0.25 seconds.

## 4.3.7 Control Change messages and bus ON/EDIT switches

| No | ON/Edit key | No | ON/Edit key | No | ON/Edit key | No | ON/Edit key |
|---|---|---|---|---|---|---|---|
| 0 | *1 | 32 | Input 32 | 64 | Group Output 4 | 96 | |
| 1 | Input 1 | 33 | Input 33 | 65 | Group Output 5 | 97 | |
| 2 | Input 2 | 34 | Input 34 | 66 | Group Output 6 | 98 | |
| 3 | Input 3 | 35 | Input 35 | 67 | Group Output 7 | 99 | |
| 4 | Input 4 | 36 | Input 36 | 68 | Group Output 8 | 100 | |
| 5 | Input 5 | 37 | Input 37 | 69 | Stereo output | 101 | |
| 6 | Input 6 | 38 | Input 38 | 70 | | 102 | |
| 7 | Input 7 | 39 | Input 39 | 71 | | 103 | |
| 8 | Input 8 | 40 | Input 40 | 72 | | 104 | |
| 9 | Input 9 | 41 | Input 41 | 73 | Aux output 1 | 105 | |
| 10 | Input 10 | 42 | Input 42 | 74 | Aux output 2 | 106 | |
| 11 | Input 11 | 43 | Input 43 | 75 | Aux output 3 | 107 | |
| 12 | Input 12 | 44 | Input 44 | 76 | Aux output 4 | 108 | |
| 13 | Input 13 | 45 | Input 45 | 77 | Aux output 5 | 109 | |
| 14 | Input 14 | 46 | Input 46 | 78 | Aux output 6 | 110 | |
| 15 | Input 15 | 47 | Input 47 | 79 | Aux output 7 | 111 | |
| 16 | Input 16 | 48 | Input 48 | 80 | Aux output 8 | 112 | |
| 17 | Input 17 | 49 | Input 49 | 81 | Matrix out 1 | 113 | |
| 18 | Input 18 | 50 | Input 50 | 82 | Matrix out 2 | 114 | |
| 19 | Input 19 | 51 | Input 51 | 83 | Matrix out 3 | 115 | |
| 20 | Input 20 | 52 | Input 52 | 84 | Matrix out 4 | 116 | |
| 21 | Input 21 | 53 | | 85 | Matrix out 5 | 117 | |
| 22 | Input 22 | 54 | | 86 | Matrix out 6 | 118 | |
| 23 | Input 23 | 55 | | 87 | Matrix out 7 | 119 | |
| 24 | Input 24 | 56 | | 88 | Matrix out 8 | 120 | |
| 25 | Input 25 | 57 | | 89 | | 121 | |
| 26 | Input 26 | 58 | | 90 | | 122 | |
| 27 | Input 27 | 59 | | 91 | | 123 | |
| 28 | Input 28 | 60 | | 92 | | 124 | |
| 29 | Input 29 | 61 | Group Output 1 | 93 | | 125 | |
| 30 | Input 30 | 62 | Group Output 2 | 94 | | 126 | |
| 31 | Input 31 | 63 | Group Output 3 | 95 | | 127 | |

*1. Blank spaces in this table indicate that the corresponding Control Change is not used

## 4.4 Error messages on the display

The console will show error messages on the display in certain circumstances. Here is a list of the messages and their meaning:

| Message on display | Meaning |
|---|---|
| R_er | MIDI data receive error |
| BUFF | The MIDI receive buffer is full – you may want to turn off the ECHOBACK function to clear this. |
| M_pr | An attempt has been made to store a scene memory while memory protection is on |
| Csum | A MIDI Bulk Dump has been received, but a checksum error has occurred |
| Low | The battery voltage has dropped below 2.5V |
| N_dt | An attempt has been made to recall a scene which contains no data |
| N_no | An attempt has been made to initialize all scene memories when no data has been stored |
| Er_* | A system error has occurred. The message will appear for five seconds. Make a note of the number and contact your nearest Yamaha service center. |

## 4.5 MIDI bulk dump formats

BULK OUT data is transmitted and received in the following format:

| Purpose of data | Binary value | Hex value | Explanation |
|---|---|---|---|
| Status | 11110000 | F0h | System Exclusive message |
| ID number | 01000011 | 43h | Manufacturer ID (Yamaha) |
| Sub status | 0000xxxx | 0nh | n=0 through 15 (MIDI channel 1 through 16) |
| Format number | 01111110 | 7Eh | Universal Bulk Dump |
| Byte count (high byte) | 00000000 | 00h | 34 (24 + 10) bytes – decimal – 22h |
| Byte count (low byte) | 00100010 | 22h | |
| Data name | 01001100 | 4Ch | 'L' |
| | 00101101 | 4Dh | 'M' |
| | 00100000 | 20h | [space] |
| | 00100000 | 20h | [space] |
| | 00111000 | 38h | '8' |
| | 01000001 | 41h | 'A' |
| | 00110100 | 34h | '4' |
| | 00110000 | 30h | '0' |
| | 01001100 | 4Ch | 'M' |
| | 0xxxxxxx | mmh | mm= o through 127 (scene memory number) |
| Data (d01) | 0000xxxx | 0nh | ON/OFF data where 0=OFF, 1=ON for Control Change 4 through 1 |
| Data (dnn) | ... | | |

| Purpose of data | Binary value | Hex value | Explanation |
|---|---|---|---|
| Data (d24) | 0000vxxx | 0nh | Memory validity flag (v) and ON/OFF data where 0=OFF, 1=ON for Control Change 95 through 93 |
| Checksum | 0xxxxxxx | eeh | ee=INVERT('L'+'M'+{d01 +... dnn + ...d24}+1) AND 07fh |
| EOX | 11110111 | F7h | End of Exclusive |

MIDI Bulk Dump requests are transmitted and received in the following format:

| Purpose of data | Binary value | Hex value | Explanation |
|---|---|---|---|
| Status | 11110000 | F0h | System Exclusive message |
| ID number | 01000011 | 43h | Manufacturer ID (Yamaha) |
| Sub status | 0010xxxx | 2nh | n=0 through 15 (MIDI channel 1 through 16) |
| Format number | 01111110 | 7Eh | Universal Bulk Dump |
| Data name | 01001100 | 4Ch | 'L' |
| | 00101101 | 4Dh | 'M' |
| | 00100000 | 20h | [space] |
| | 00100000 | 20h | [space] |
| | 00111000 | 38h | '8' |
| | 01000001 | 41h | 'A' |
| | 00110100 | 34h | '4' |
| | 00110000 | 30h | '0' |
| | 01001100 | 4Ch | 'M' |
| | 0xxxxxxx | mmh | mm= o through 127 (scene memory number) |
| EOX | 11110111 | F7h | End of Exclusive |

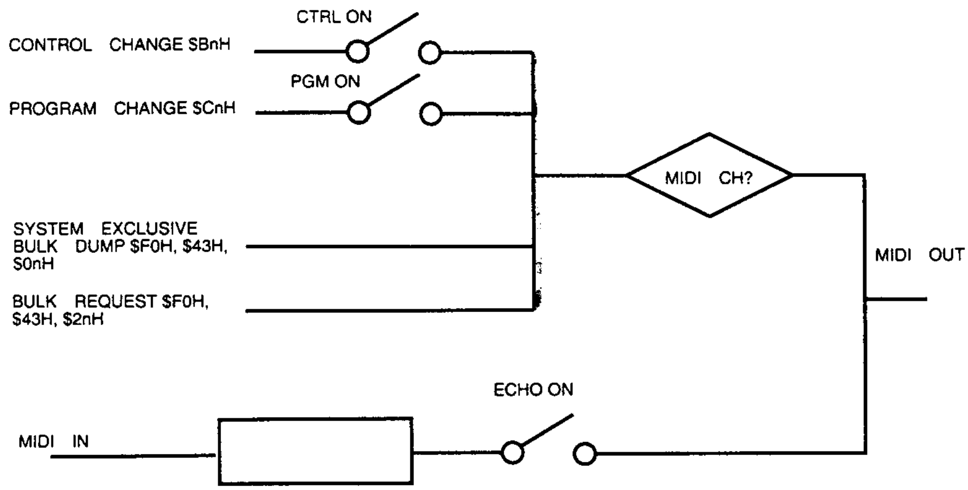
YAMAHA [ Mixing Console ] Date:24-NOV-1994
Model PM3500 MIDI Implementation Chart Version : 1.0

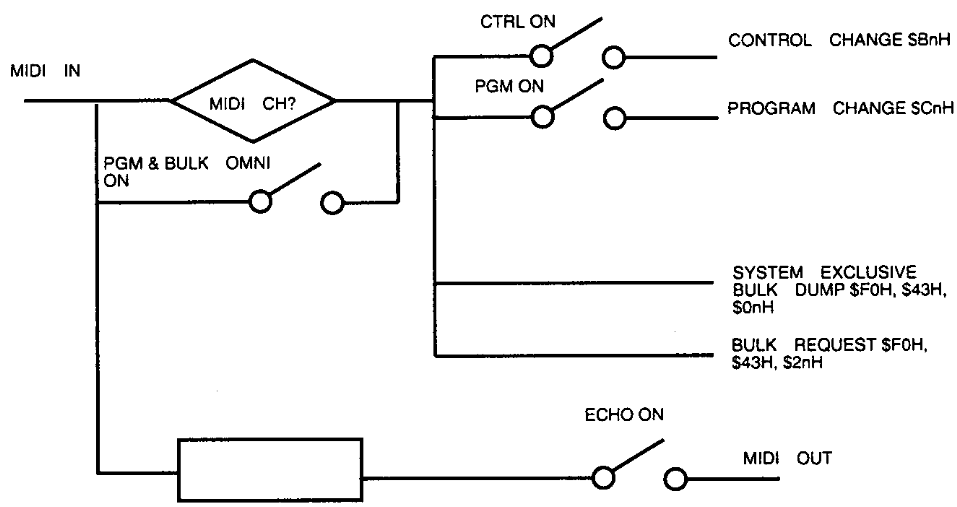
| Function ... | | Transmitted | Recognized | Remarks |
|---|---|---|---|---|
| Basic Channel | Default | 1 - 16 | 1 - 16 | memorized |
| | Changed | 1 - 16 | 1 - 16 | |
| Mode | Default | x | OMNI ON/OFF | memorized |
| | Messages | x | OMNI ON/OFF | |
| | Altered | ************** | x | |
| Note Number | | x | x | |
| | : True voice | ************** | x | |
| Velocity | Note ON | x | x | |
| | Note OFF | x | x | |
| After Touch | Key's | x | x | |
| | Ch's | x | x | |
| Pitch Bender | | x | x | |
| Control Change | 1 - 88 | x | o | *1 |
| | : True # | 0/127 | | |
| Prog Change | | o | o | *2 |
| | : True # | 1 - 128 | | |
| System Exclusive | | o | o | Bulk Dump/Request |
| System Common | : Song Pos. | x | x | |
| | : Song Sel. | x | x | |
| | : Tune | x | x | |
| System Real Time | :Clock | x | x | |
| | :Commands | x | x | |
| Aux Mes-sages | :Local ON/OFF | x | x | |
| | :All Notes OFF | x | x | |
| | :Active Sense | x | x | |
| | :Reset | x | x | |

Notes: *1 See Control Change chart.
*2 For program 1 - 128, memory 1 - 128 is selected.

Mode 1: OMNI ON, POLY Mode 2: OMNI ON, MONO o: Yes
Mode 3: OMNI OFF, POLY Mode 4: OMNI OFF, MONO x: No

## 4.7 Midi flowcharts (transmission and reception)



# 5 Installation notes

## 5.1 Planning an installation

Before installing the console, 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 can be used to support the console. It should be capable of supporting at least the weight of the console (see the specifications in Section 3) 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. For custom installations, the dimensions listed in Section 3 (see "Dimensions" on page 41) of this manual can be given to the carpenter or other personnel responsible for building the console support.

Provide a location within 10 feet (3.5m) of the console for housing the PW4000 power supply. This supply may be rack-mounted, or may be placed on a shelf. For touring or critical fixed applications, it may be advisable to purchase a spare PW4000 supply and to mount it next to the main supply; a changeover is then possible in the rare event of a problem.

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

## 5.2 Power mains

### 5.2.1 Verify the correct mains voltage

PW4000 power supplies sold in the U.S.A. and Canada are designed to operate with 110 to 120 V, 60Hz AC power mains. The General Export model operates on 220 to 230 V, 50Hz AC mains. The British model operates on 240 V 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 dealer for assistance.

### 5.2.2 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 ensure 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 is 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 PW4000 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 cable 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 ade-**

**quate, check it again after turning on the PW4000 with the console 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 PW4000 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.**

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### 5.2.3 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.

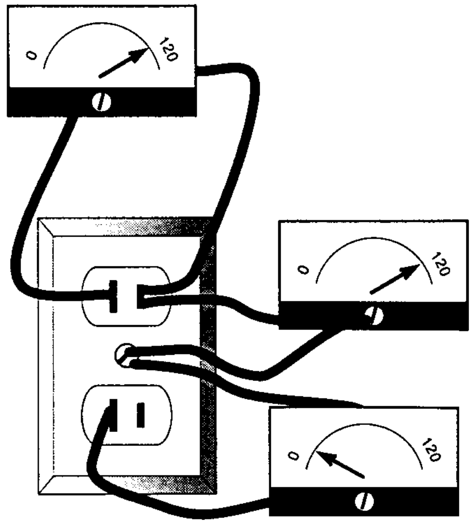
However, you must be sure that the screw is grounded.

Connect the adaptor's green wire to the screw on the two-wire outlet.

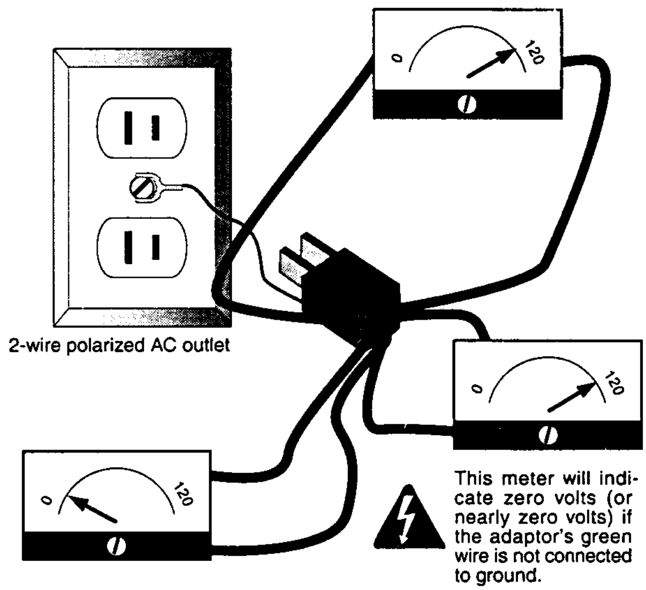
Plug the adaptor into the outlet.

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.




*Testing a 2-wire AC outlet*




*Testing a 2-wire AC outlet and a 3-prong to 2-prong adaptor*

### 5.2.4 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.

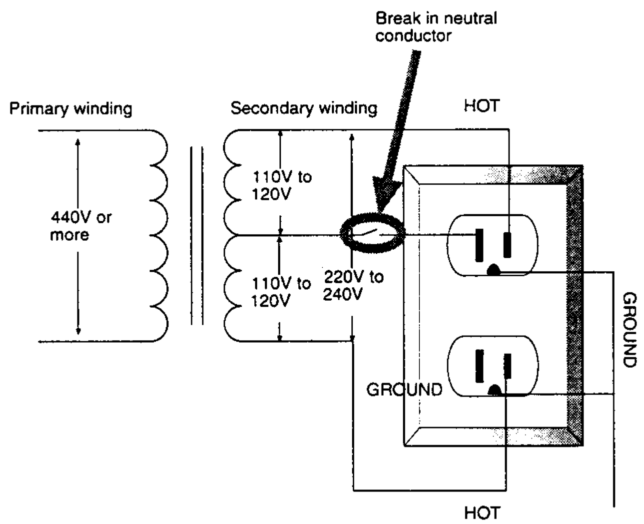
### 5.2.5 Improperly wired AC outlets: lifted neutral

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 240V available from the power service instead of the desired 110 to 120V.

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

If the PW4000 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.

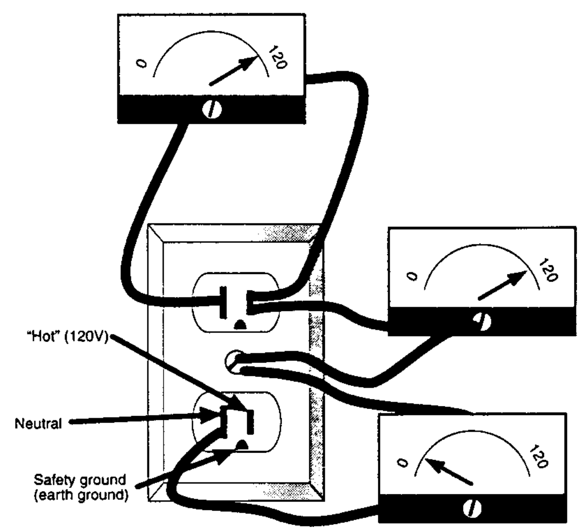
If you detect any voltage between the larger slot (white wire) in an outlet and the ground terminal (round prong, green wire) when there is no load on that line, you should contact a licensed electrician to check it out and correct the situation.




**WARNING**

**In North American 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.**




3-wire AC outlet

The third prong (socket ground) and center screw of the outlet are internally connnected and grounded.

### 5.2.6 AC safety tips

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.

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.

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).

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.

### 5.2.7 Power source integrity

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

### 5.2.8 Turn-on sequencing

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

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

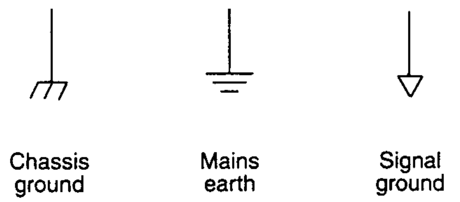
Ground is the electrical reference against which potentials (voltages) are expressed. In a practical audio system, a number of different independent references exist in various local subsystems. These may or may not be at the same electrical potential. If handled properly, they certainly need not be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

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

**Signal Ground** – the reference point against which signal potentials in a specific piece of equipment or group of components are expressed.

**Earth Ground** – the local electrical potential of the earth. In practice, earth is the potential of the central, rounded terminal in a U.S. standard three-prong 120-volt outlet. Earth is sometimes obtained from a metal cold water pipe (though this practice has been criticized recently as unreliable due to increasing use of non-conductive ABS plastic pipe sections), or from a chemical earthing rod sunk into the moistened ground.

**Chassis Ground** – the chassis connection point of a specific component. In equipment fitted with a three prong AC plug, the chassis is normally connected to earth, with provision to connect signal ground to earth as well. Equipment having a two prong AC plug will normally have the chassis connected to signal ground.




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

## 5.3.1 Why is proper grounding important?

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

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

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

## 5.3.2 Ground loops

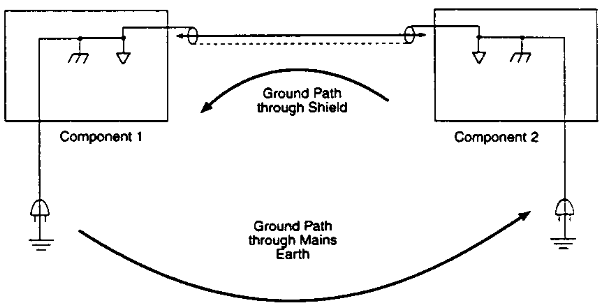
AC line-frequency hum is, without question, the single most common problem in sound systems, and the most common cause of hum is ground loops.

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

Ground loops often are difficult to isolate, even for experienced audio engineers. 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 redesign 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).

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

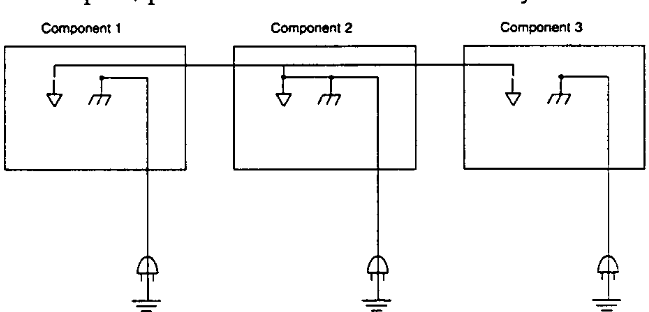
The figure below shows a typical ground loop situation. Two interconnected pieces of equipment are plugged into grounded AC outlets at separate locations, and signal ground is connected to earth in each of them. The earth ground path and duplicate signal ground path form a loop which can pick up interference.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



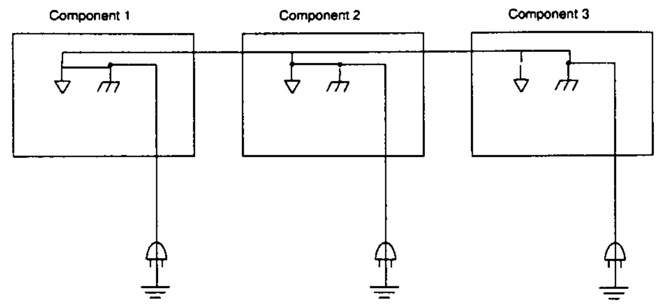

### 5.3.3 Basic grounding techniques

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

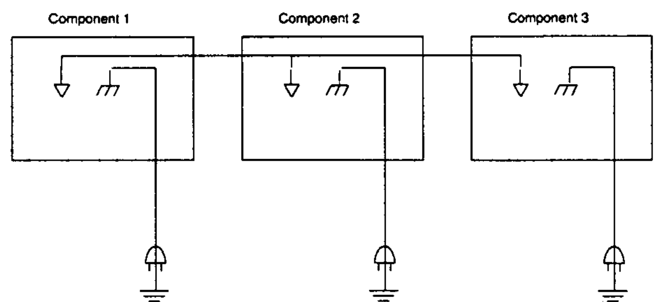
The figure below illustrates the single-point grounding principle. Chassis ground in each individual component is connected to earth; signal ground is carried between components and connected to earth at one central point. This configuration is very effective in eliminating line frequency hum and switching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



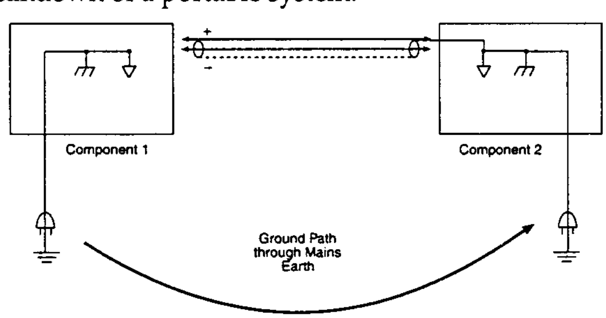

Multiple point grounding is shown in the next illustration. This situation is common in systems that use unbalanced equipment having the chassis connected to signal ground. It has the advantage of being very simple in practice, but it is not very reliable † particularly if the connection configuration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




The floating ground principleis shown below. Note that signal ground is completely isolated from earth. This scheme is useful when the earth ground system carries significant noise, but it relies on the equipment input stages to reject interference induced in cable shields.

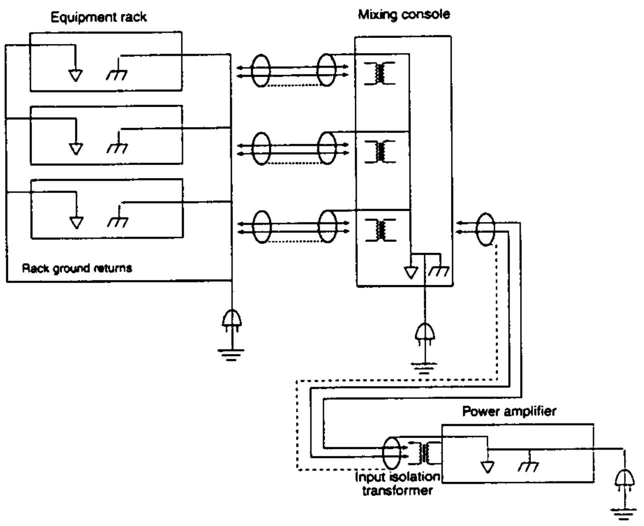



The principle of telescoping shields is shown below. This scheme is very effective in eliminating ground loops. If shields are connected only to earth, unwanted signals that are induced in them can never enter the signal path. Balanced lines and transformers are required to implement this approach, since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



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

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




### 5.3.4 Balanced lines and ground lift switches

By using balanced signal lines between two pieces of sound equipment, 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, including the Yamaha PC series of amplifiers, is equipped with ground lift switches on the balanced inputs.

**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.**

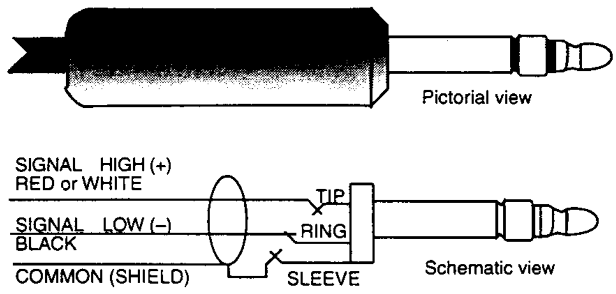
In those audio devices which anticipate ground loops by providing "ground lift" switches next to XLRs or three-wire phone jacks, the ground lift switch makes and breaks the connection between the connector's shield and the chassis of the particular device. Ground lift switches are usually found on "direct 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).

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.

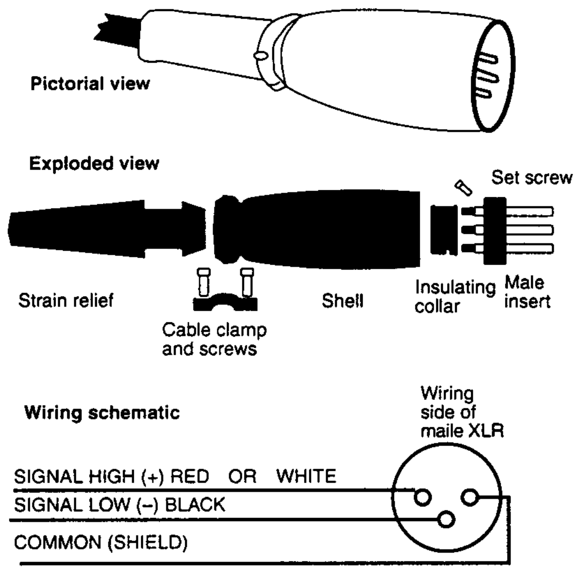
## 5.4 Audio connectors and cables

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.

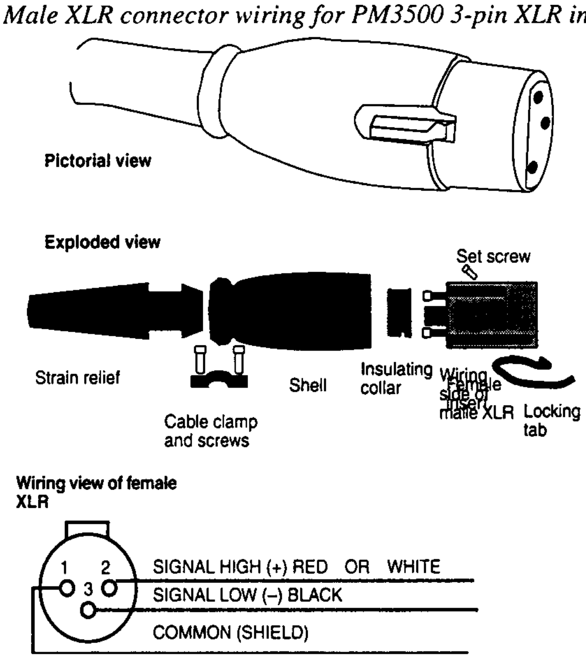
The PM3500 is fitted with only two types of audio connectors: 3-pin XLRs, both male and female, and 3-circuit (tip/ring/sleeve) 1/4" phone jacks, also known as stereo phone jacks, although their function is always (except in the case of the headphone jacks) to carry a balanced mono signal rather than a stereo signal.




*TRS phone plug wiring for PM3500 insert IN/OUT jacks*




*Male XLR connector wiring for PM3500 3-pin XLR inputs*




*Female XLR connector wiring for PM3500 3-pin XLR outputs*

### 5.4.1 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.

### 5.4.2 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 console 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.

## 5.4.3 Balanced versus unbalanced wiring

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

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

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

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

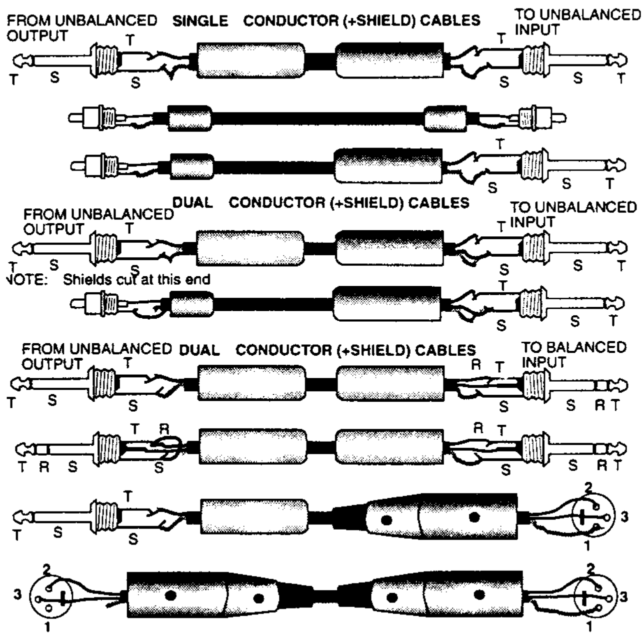
Balanced wiring is more expensive to implement than unbalanced wiring. It is often used, however, because it offers useful advantages, especially in portable sound systems. There is nothing inherently "better" or more "professional" about balanced wiring; the application dictates whether one system or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

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

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

Not all balanced wiring has a shield. In older telephone systems, many miles of cable were run with no shielding in order to save money (now fiber optic cables are replacing costly copper with inexpensive glass or plastics). Out in the open, wires are subjected to radio interference and to hum fields emitted by power lines. Balancing the two signal hot wires with respect to ground gives long lines immunity to external interference. Twisting two wires together theoretically subjects each wire to the same amount of electrostatic or electromagnetic noise. A balanced input will then cancel the unwanted noise signals common to both wires, while passing the desired audio signal, as illustrated in Figures 4-14.

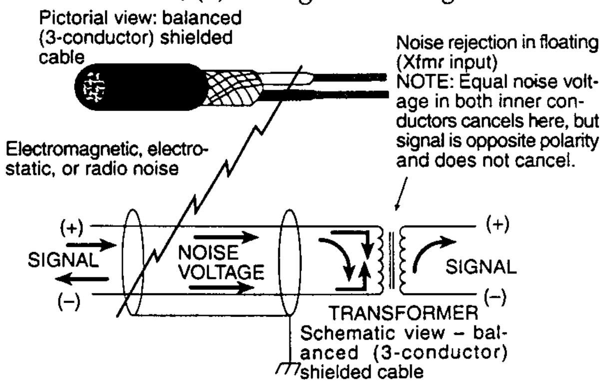
The RFI (radio frequency interference) cuts across both conductors, inducing equal voltages in the same direction. These voltages "meet" in the differential amplifier (or transformer), and cancel out, while the signals generated by the microphone flow in opposite directions in each conductor, and hence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



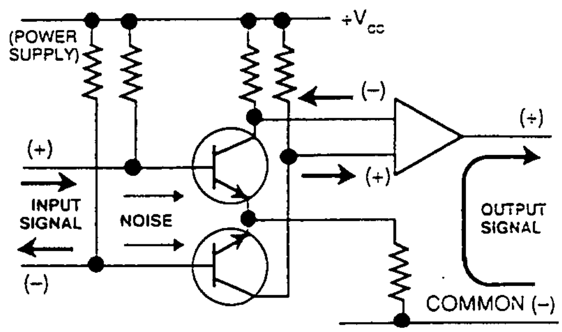

Connect pin 1 at both ends of a mic cable.
For a line cable, cut the shield here, as shown.

## 5.4.4 The pros and cons 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 PM3500, 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.




*Transformer (balanced) floating input*




*Balanced (differential) input*

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 ratio (CMRR) of an electronically balanced input may be insufficient to cancel the noise induced in the cable. In such cases, input transformers can be useful. Also, there is incomplete ground isolation 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. (In reality, the transducer capsule in a microphone is generally isolated and insulated from the mic case, so an electronically balanced input still would not permit a current to flow through the mic… assuming everything is wired correctly in the microphone.) If 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.

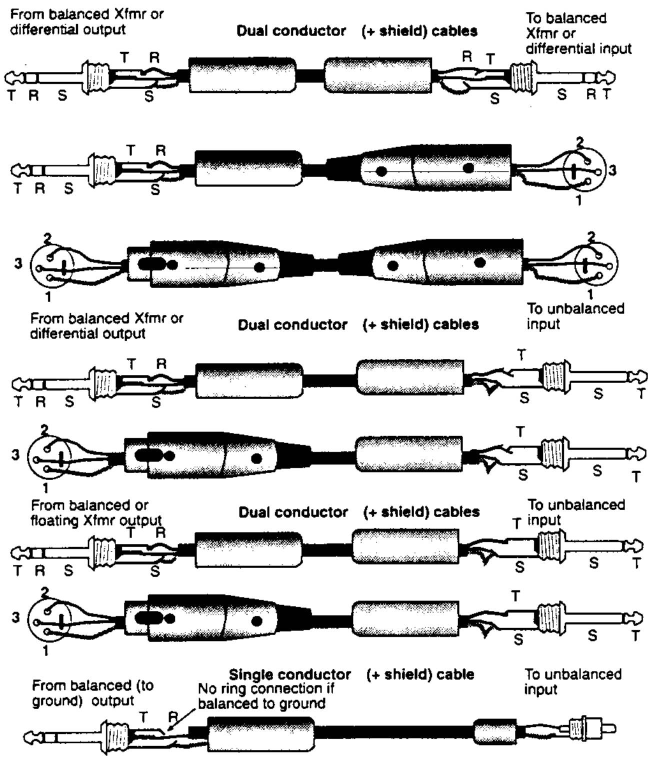
If the transformer is also used to prevent a low impedance input from overloading a high impedance output, it is known as a "bridging" transformer (not to be confused with the "bridged" connections of a stereo power amp output in mono mode).

In general, the PM3500 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 IT3000) can be installed in individual input modules. Similarly, PM3500 outputs can be transformer isolated by purchasing one or more optional output transformer sets. The Model OT3000 output transformer set contains 8 transformers, with XLR connectors, in a compact 19-inch rack mountable box that is external to the PM3500. 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 microphone. One can digitize the audio signal and transmit it by means of modulated light in fiber optics, but this can be 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.**




### 5.4.5 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 (about 8m), 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 (about 30m) or more are acceptable. For very low impedance sources of 50-ohms or less, cable lengths of up to 1000 feet (300m) 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.

## 5.5 Direct boxes (DI boxes)

The so-called "direct box" or "DI box" is a device used 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, 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, originally designed by the late Deane Jensen. Schematics are presented at the end of this chapter (page 78). While these designs are believed to work well with the PM3500, 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.

### 5.5.1 Passive guitar direct box

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

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

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

The transformer was designed specifically for use in a guitar direct box. When connected to a typical electric guitar pickup, and an XLR channel input on a PM3500, 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 PM3500 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 JT-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 approximate 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 PM3500'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.

### 5.5.2 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 PM3500 or any condenser microphone power supply.

The circuit can be constructed on a piece of perforated 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.

## 5.6 Configuring equipment racks

The great majority of audio equipment manufacturers make provision for their electronic products to be mounted in EIA standard 19 inch wide equipment racks. (The equipment may be only 17 to 18 inches in width, or even less. The rack ears that mount to the rack rails extend to 19 inches.) Panel heights for rack mounting equipment are standardized on multiples of a single rack unit space (1 RU) of 1.75 inches.

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

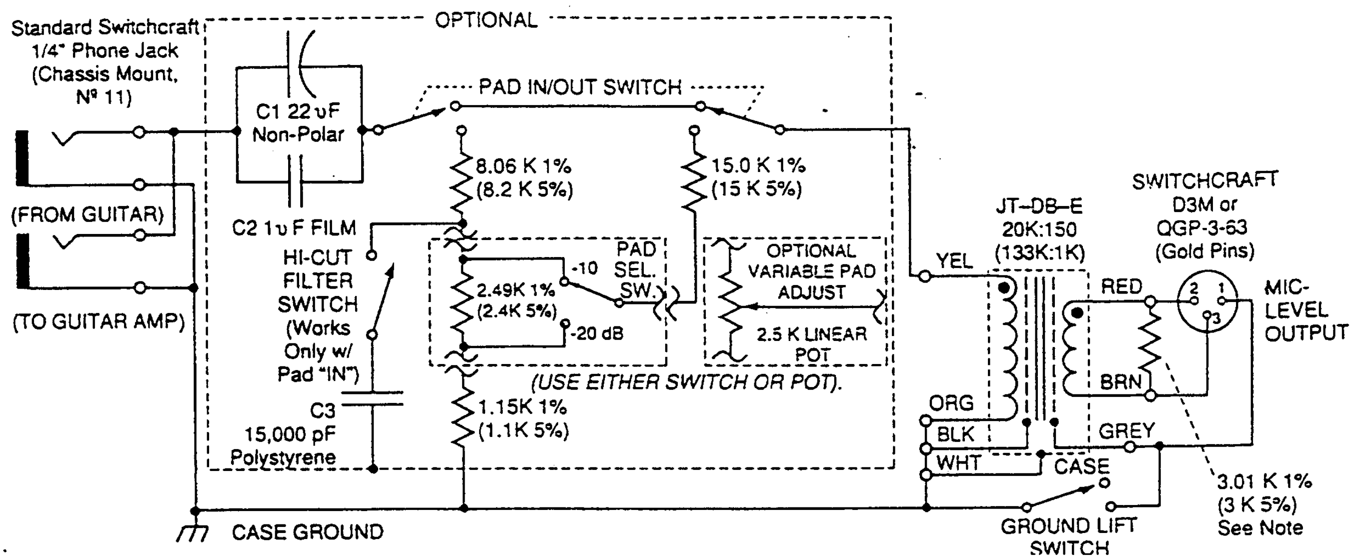
Before actually mounting the selected components, it is wise to carefully plan out each rack with an eye to signal flow, heat flow, and weight distribution. It might be best to mount together components that function as a group: the equalizer, active crossover and power amplifier for a single loudspeaker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

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

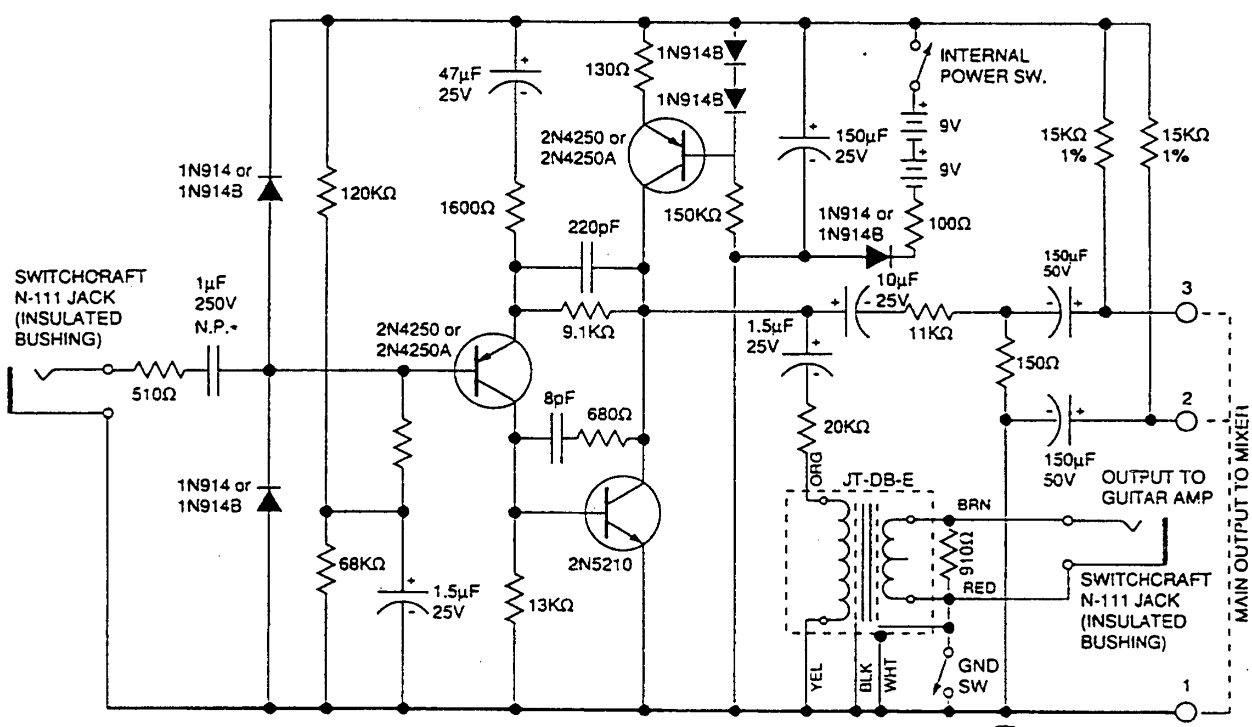
The hallmark of a professional rack is the care that is taken with the internal wiring. Color coding and/or clear and logical cable marking facilitate troubleshooting and reflects an understanding of the electronic signal flow. Related groups of connections are neatly routed and bundled with cable ties.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

## 5.7 Direct box schematics

### 5.7.1 Passive box




### 5.7.2 Active box




*Amperex "Nugget" Series

# 6 Gain structure and levels

## 6.1 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., +4dBu) when one merely wishes to describe a general sensitivity range. For this reason, most audio engineers think of operating levels in three general categories:

### 6.1.1 Mic level or low level

This range extends from no signal up to about –20dBu (77.5mV), or –20dBm (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 20dB above this level (to 0dBu or higher), this remains the nominal, average range.

### 6.1.2 Line level or medium level

This range extends from –20dBu or –20dBm to +30dBu (24.5V) or +30dBm (24.5V 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 –10dBu/dBm (245 millivolts), +4dBu/dBm (1.23V) or +8dBu/dBm (1.95V).

### 6.1.3 Speaker level and high level

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

**NOTE**

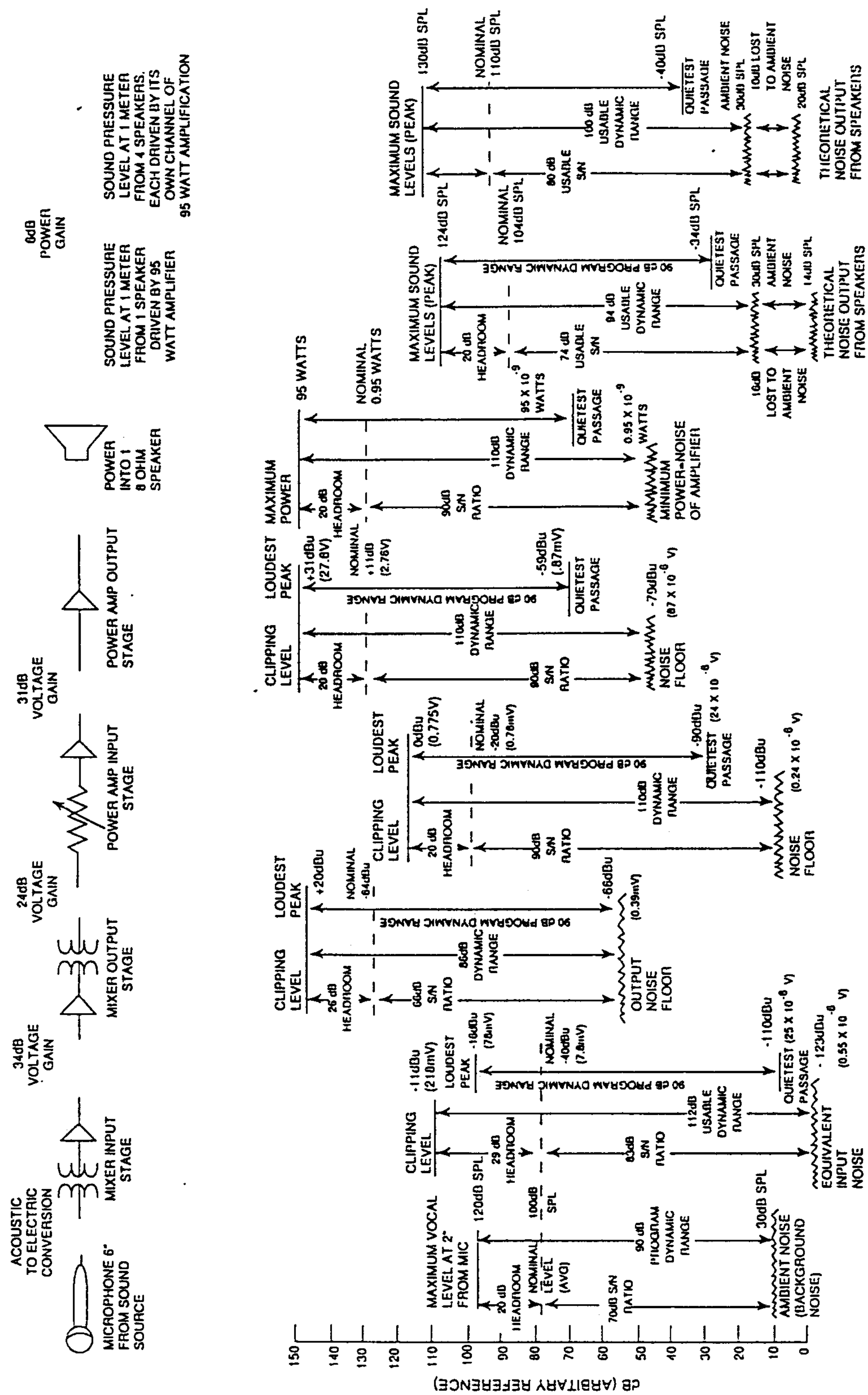
**A piece of consumer sound equipment ("hi-fi") may operate at considerably lower nominal (average) line levels than +4dBu. This is typically around –16dBu (123 mV) to –10dBu (245mV) into 10,000Ω or higher loads. Peak output levels in such equipment may not go above +4dBu (1.23V). The output current available here would be inadequate to drive a 600Ω 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:

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

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

## 6.2 Dynamic range and headroom




*Dynamic range and headroom in sound systems*

### 6.2.1 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.

### 6.2.2 The relationship between sound levels and signal levels

A concert with sound levels ranging from 30dB SPL (near silence) to 120dB SPL (threshold of pain) has a 90dB 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 120dB SPL, the maximum line levels (at the console's output) may reach +24dBu (12.3V), and maximum power output levels from a given amplifier may peak at 250 watts. Similarly, when the sound level falls to 30dB SPL, the minimum line level falls to –66dBu (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 90dB: +24dBu – (–66dBu) = 90dB. This dB SPL to dBu or dBm correspondence is maintained throughout the sound system, from the original source at the microphone, through the electrical portion of the sound system, to the speaker system output. A similar relationship exists for any type of sound reinforcement, recording studio, or broadcast system.

### 6.2.3 A discussion of headroom

The average line level in the typical commercial sound system just described is +4dBu (1.23V), corresponding to an average sound level of 100dB 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 20dB. Why is this so? Subtract the nominal from the maximum and see: 120dB SPL– 100dB SPL = 20dB. The headroom is always expressed in just plain "dB" since it merely describes a ratio, not an absolute level; "20dB" is the headroom, not "20dB SPL". Similarly, the console output's electrical headroom is 20dB, as calculated here: +24dBu– (+4dBu) = 20dB. Again, "20dB" is the headroom, not "20dBu". 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 20dB of headroom (20dB above nominal = 100 times the power).

### 6.2.4 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 –56dBu (1.23 millivolts), and a peak output level of +18dBu (6.16V). The dynamic range of this system would only be 74dB. Assuming the original program still has an acoustic dynamic range of 90dB, it is apparent that 16dB 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 16dB 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 +24dBu output capability and a -66dBu or better noise floor, or +18dBu output capability and -82dBu noise floor, would be able to handle the full 90dB dynamic range. Thus, for high quality sound reinforcement or music reproduction, it is necessary that the sound system be capable of low noise levels and high output capability.

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

### 6.2.5 A general approach to setting levels in a sound system

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

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

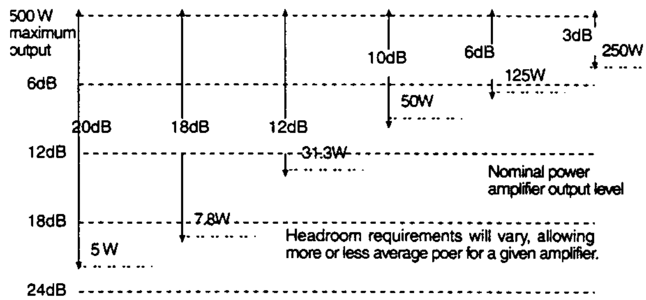
If line amplifiers, electronic crossovers, equalizers or other signal processing devices are inserted in the signal chain, signal levels at the input of these units should be set so the dynamic range of each unit is optimized. In other words, set the input level at each device as high as possible without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

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

### 6.2.6 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, 15dB to 20dB of headroom is desirable. For most sound reinforcement applications, especially with large numbers of amplifiers, economics play an important role, and a 10dB 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 6dB is not unusual. To achieve such a low headroom figure, an extreme amount of compression and limiting will be necessary; while the sound may be somewhat unnatural, the message will "cut through."




Let's go through an actual setup procedure for a high quality, music reproduction system. First choose a headroom figure. For maximum fidelity when reproducing music, it is desirable to allow 20dB of headroom above the average system output. While some extreme musical peaks exceed 20dB, the 20dB figure is adequate for most programs, and allowing for greater headroom can be very costly. A 20dB 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 PM3500 meters (0 VU +4dBu, which allows 20dB headroom before the console reaches its maximum +24dBu output level).

Remember that with a 20dB 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 20dB 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 20dB headroom figure would be made as follows:

1. Initially, set the attenuators on the power amp at maximum attenuation (usually maximum counter-clockwise rotation). Feed a sine wave signal at 1000Hz to the console input at an expected average input level (approximately -50dBu (2.45mV) for a microphone, +4dBu (1.23V) for a line level signal. The exact voltage is not critical, and 1000Hz is a standard reference frequency, but any frequency from 400Hz to about 4kHz may be used.
2. Set the input channel fader on the console at its marked "nominal" setting, and adjust the channel Gain so that the channel's LED meter read zero. Be sure this channel is assigned to an output bus (i.e., one of the group busses or the stereo bus).
3. Set the master fader for the bus to which the channel is assigned so that the output level is 20dB below the rated maximum output level for the console. Suppose, for example, the maximum rated output level is +24dBu (12.3V); in that case, the output level should be adjusted to +4dBu (1.23V), as indicated by a "zero" reading on the console's VU meter (0 VU corresponds to +4dBu with a steady-state sine wave signal output per factory calibration).
4. If the rated maximum input level for the graphic equalizer to which the console output is connected is +24dBu (12.3V), then no adjustment or padding of the input to the EQ is required. If the maximum input level is lower, for example +18dBu, then there would be reduced headroom in the EQ unless its input is attenuated. Subtracting +4dBu from +18dBu leaves only 14dB of headroom, so in order to maintain the desired 20dB of headroom, 6dB of attenuation must be dialed in at the EQ input, or a 6dB resistive pad should be inserted between the console output and the equalizer input. The nominal signal level at the input to the equalizer should now be -2dBu (616mV), which can be checked with a voltmeter.
5. Assume that the maximum rated output level of the equalizer in this example is +18dBu (6.16V). Adjust the master level control on the equalizer so that its output level is 20dB below the rated maximum, or -2dBu (616mV). 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 +4dBu, which may seem to be 6dB less sensitive than required for the necessary headroom. However, any boost applied with the EQ will raise the nominal level of the signal at the EQ output, so this may help preserve adequate headroom in the console. Remember, though, that applying boost with an equalizer can reduce headroom within the EQ itself, so you may want to turn down the EQs output level to preserve the headroom.**

6. Finally, starting with the attenuator(s) on the power amplifier at maximum attenuation (maximum counterclockwise rotation), slowly decrease the attenuation (raise the level), observing the amplifier's output level. When the POWER output is 1/100 of the maximum rated power (1/10 of the maximum output voltage), the amplifier has 20dB 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 20dB for the loud peaks.

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

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

## 6.3 Gain overlap and headroom

As explained previously, the PM3500 can deliver +24dBu 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 +4dBu sensitivity rating will be driven 20dB into clipping if driven with the full output capability of the console. 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 +24dBu 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 PM3500 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 6dB 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 console outputs are connected to a tape machine. Many professional tape machines are subject to tape saturation at input levels above +15dBu. Why would one want +24dBu 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 +18dBu, for example, that clipping would overlay a very harsh distortion on the 3dB of "soft" saturation on the tape. Because the PM3500 does not clip until its output reaches +24dBu, there is less chance of applying harsh distortion to the tape. Today, however, there is another consideration: digital recording technology. Here, the available dynamic range of the digital tape recorders or direct-to-disk recorders is so great that all the headroom a console can provide is advantageous.

# 7 Optional functions

The console 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 this console, a number of optional functions have been built in, and can be selected by moving factory preset switches or jumpers within certain modules.

**WARNING**

**Underwriter's Laboratories (UL) requires that we inform you there are no user-serviceable parts inside the PM3500. 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.**
**We at Yamaha additionally caution you never to open the console and remove or install a module for the purpose of inspection, replacement or changing the preset switches unless the power has first been turned off. If a module is removed or installed with power on, the circuitry may be damaged. Unless you are a qualified service technician, do not plug in the AC cord while the interior of the power supply is exposed; dangerous voltages may exist within the chassis, and lethal shock is possible.**

## 7.1 Removing and installing a module

1. Loosen the screws ① on the front of the meter bridge. The number of screws differs according to the number of input channels. (Fig.1)
2. Loosen the screws ② on the rear of the meter bridge. (Fig.2)
3. Open the meter bridge as shown in Fig.1.

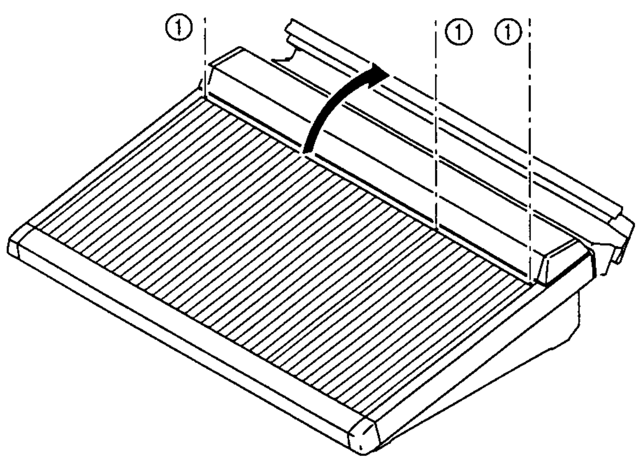



Fig.1

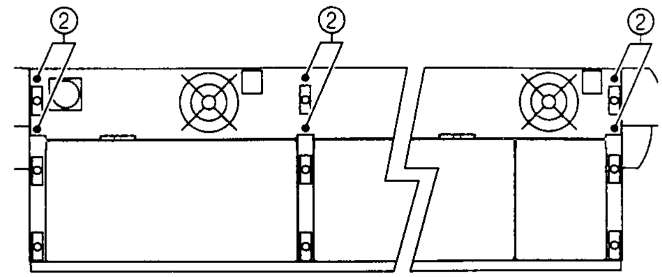



Fig.2

4. Remove the ground connection ③ on the ground bridge. (Fig.3)
5. Remove the ground bridge holding screws so that the ground bridge ④ can be raised and the three connections ⑤ disconnected. (Fig.3)

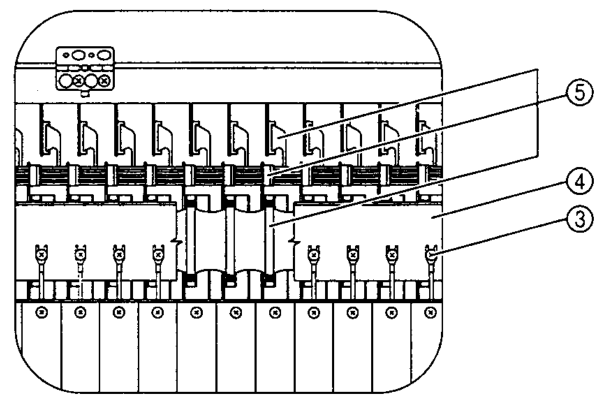



Fig.3

6. Loosen the retaining screws on the top and bottom of the module. These screws ⑥ should remain in the module. (Fig.4)
7. Lift up the bottom of the module, then carefully pull the module out of the console. (Fig.4)

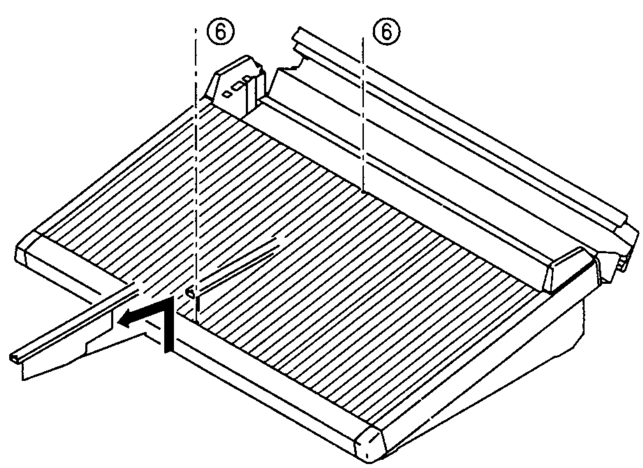


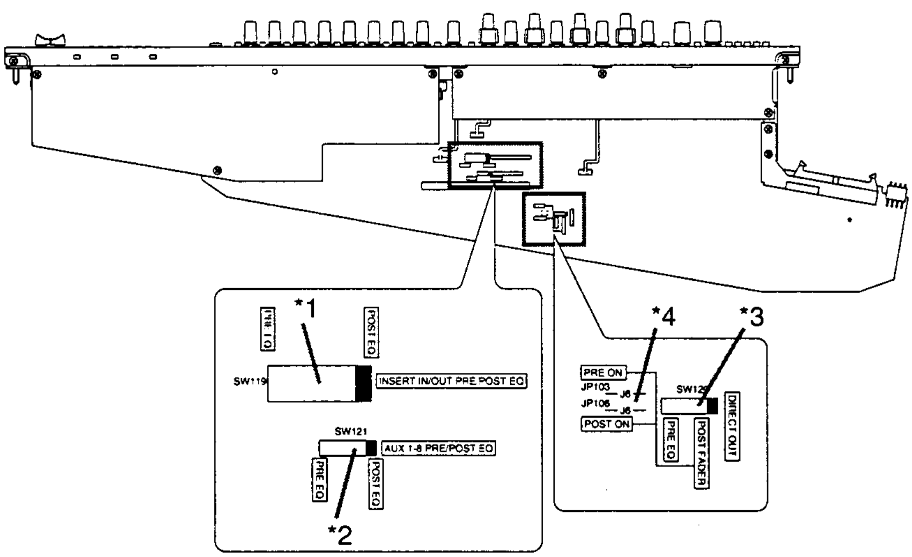

Fig.4

Installation of a module should be carried out by performing this procedure in reverse.

**NOTE**

**If you are moving a module to a different location in the mainframe, one which had housed no module or a different type of module, then you will have to also move the rear connector panel. Monaural and stereo input modules may be placed anywhere in the frame, and you can exchange them freely. However, there should be no more than a total of 52 input channels per mainframe.**
**In addion, you should note that the rear panels for monaural and stereo modules will also have to be changed. Monaural and stereo rear panels are each 4 modules in width. Contact your Yamaha dealer for details of availability of these rear panels. See "Optional equipment and parts for the PM3500 console" on page 45 for full details of partr numbers, etc.**

## 7.2 Monaural input modules




Monaural input modules can have the following settings changed by switches or jumpers mounted on the module boards:

| | Setting | Options | Factory setting |
|---|---|---|---|
| *1 | Insert I/O | pre-EQ / post-EQ | post-EQ |
| *2 | Aux 1 through 8 | pre-EQ / post-EQ | post-EQ |
| *3 | Direct out | pre-EQ/ post-fader | post-fader |
| *4 | Direct out | pre-ON / post-ON | pre-ON |

### 7.2.1 Monaural input module insert I/O pre- or post-EQ (switch)

A slide switch in each monaural input module permits the INSERT I/O point to be altered. As shipped, the console is set so that the INSERT OUT point is derived after the EQ. If you want the Insert Out to be pre-EQ, move the switch to the appropriate position, as illustrated.

### 7.2.2 Monaural input module direct out: pre-EQ or post-fader (switch) pre-ON and post-ON switch (jumper)

A slide switch in each monaural input module permits the direct out point to be altered. As shipped, the console is set so that the direct out point is derived post-fader . If you want the direct out to be pre-EQ, move the switch to the appropriate position, as illustrated.

As shipped, the direct out point comes ahead of the Channel ON switch, and is thus not affected by the scene memory functions. By changing internal jumpers, you can alter the Direct Out point to be Post-ON switch, also illustrated.

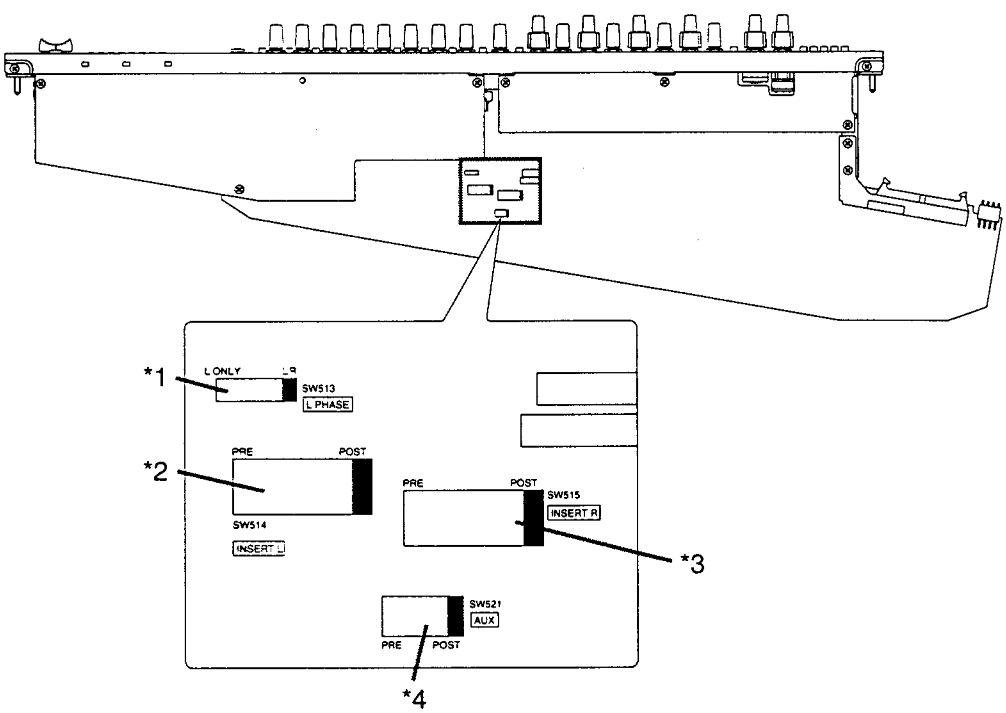
### 7.2.3 Monaural input module: AUX 1 through 8 pre-EQ or post-EQ (switch)

A slide switch in each monaural input module affects the points of all eight AUX busses. As shipped, the console is set so that these points are derived after the EQ. If you want the AUX sends to be derived pre-EQ, move the switch to the appropriate position, as illustrated.

### 7.2.4 Monaural input module: group 1 through 8 pre-EQ or post-EQ (switch)

A slide switch in each monaural input module affects the points of all eight group busses. As shipped, the console is set so that these points are derived after the EQ. If you want the groups to be derived pre-EQ, move the switch to the appropriate position, as illustrated.

## 7.3 Stereo input modules




Stereo input modules can have the following settings changed by switches or jumpers mounted on the module boards:

| | Setting | Options | Factory setting |
|---|---|---|---|
| *1 | Phase | L & R / L only | L & R |
| *2 | L insert I/O | pre-EQ / post-EQ | post-EQ |
| *3 | R insert I/O | pre-EQ / post-EQ | post-EQ |
| *4 | Aux 1 through 8 | pre-EQ / post-EQ | post-EQ |

### 7.3.1 Stereo input module: phase reversal (switch)

A slide switch in each stereo input module permits the changing of the phase switch function. As shipped, the phase switch will affect the polarity of both the left and right channels. If you want the phase switch to affect only the L channel, move the switch to the appropriate position, as illustrated.

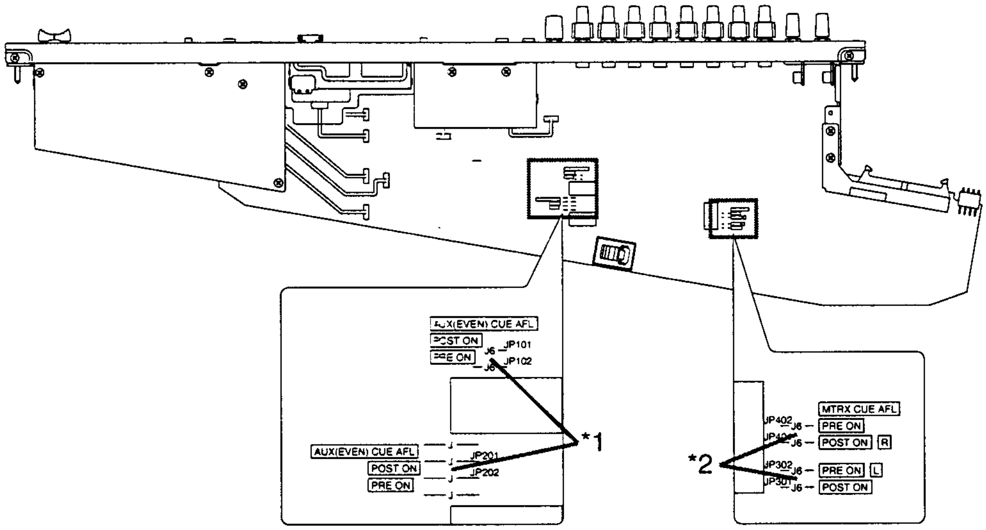
### 7.3.2 Stereo input module: L and R insert I/O insert I/O pre- or post-EQ (switch)

Two slide switches in each stereo input module permits the L and R INSERT I/O point to be altered independently. As shipped, the console is set so that both NSERT OUT points are derived after the EQ. If you want either or both of the Insert Outs to be pre-EQ, move the relevant switch(es) to the appropriate position, as illustrated.

### 7.3.3 Stereo input module: AUX 1 through 8 pre-EQ or post-EQ (switch)

A slide switch in each stereo input module affects the points of all eight AUX busses. As shipped, the console is set so that these points are derived after the EQ. If you want the AUX sends to be derived pre-EQ, move the switch to the appropriate position, as illustrated.

## 7.4 Group master modules 1




The group master modules (1) can have the following settings changed (which affect the operation of the aux and matrix busses housed on the module) by switches or jumpers mounted on the module boards:

| | Setting | Options | Factory setting |
|---|---|---|---|
| *1 | AUX CUE | pre-ON /post-ON | post-ON |
| *2 | Matrix CUE AFL | pre-ON / post-ON | post-ON |

### 7.4.1 Group master module 1: AUX CUE AFL pre- /post-ON jumpers

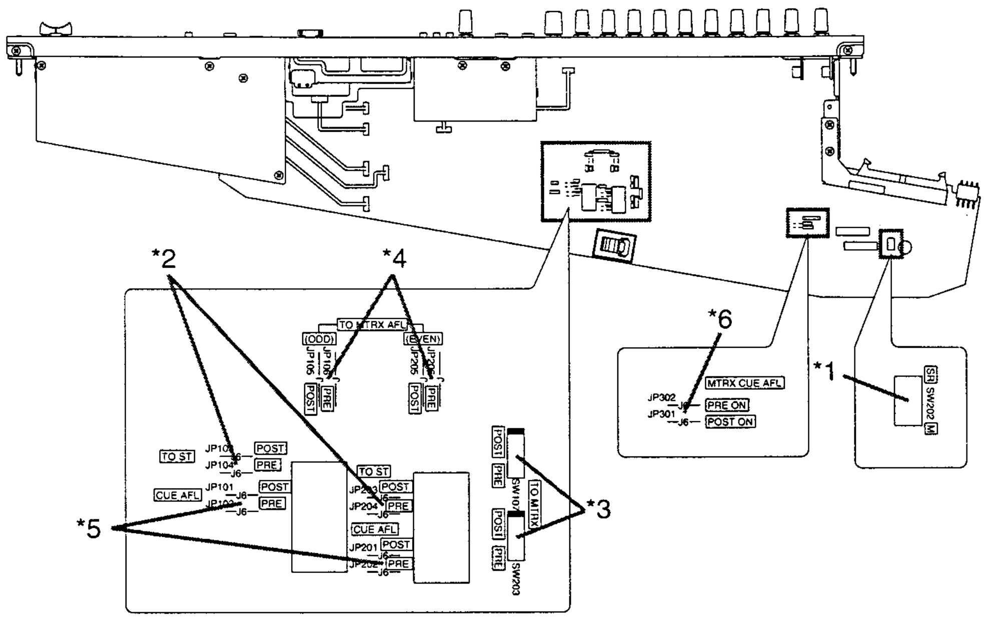
As shipped, aux CUE AFL levels are derived post-ON switch. If these jumpers are changed, the aux AFL levels are independent of the matrix ON switches.

### 7.4.2 Group master module 1: matrix CUE AFL pre- /post-ON jumpers

As shipped, matrix CUE AFL levels are derived post-ON switch. If these jumpers are changed, the matrix CUE AFL levels are independent of the matrix ON switches. Note that there are two jumpers for each side of the stereo matrix bus.

## 7.5 Group master modules 2

The group master modules (2) can have the following settings changed (which affect the operation of the group and matrix busses housed on the module) by switches or jumpers mounted on the module boards:




| | Setting | Options | Factory setting |
|---|---|---|---|
| *1 | Group sum gain | 0dB /+6dB | 0dB |
| *2 | Group to ST | pre-ON /post-ON | post-ON |
| *3 | Group to matrix | pre-fader/ post-fader | post-fader |
| *4 | Group to matrix | pre-ON / post-ON | post-ON |
| *5 | Group CUE AFL | pre-ON / post-ON | post-ON |
| *6 | Matrix CUE AFL | pre-ON / post-ON | post-ON |

### 7.5.1 Group master module 2: group sum gain (0, +6dB) switch

Each group module can have the group sum gain boosted by 6dB when the switch shown in the illustration above is moved to the "M" (monitor) position from the "SR" (sound reinforcement) position. The factory setting is the "SR" (0dB boost) position. The switch is logically prior to the sum peak indicator. Note that one switch affects both groups.

### 7.5.2 Group master module 2: group-to-stereo jumpers)

This pair of jumpers can change the group-to-stereo switch [55] (page 24) function from its factory setting of post-ON switch, to pre-ON. In other words, if these jumpers are moved, the group-to-stereo switch is independent of the ON switch status.

### 7.5.3 Group master module 2: group-to-matrix pre- /post-fader jumpers

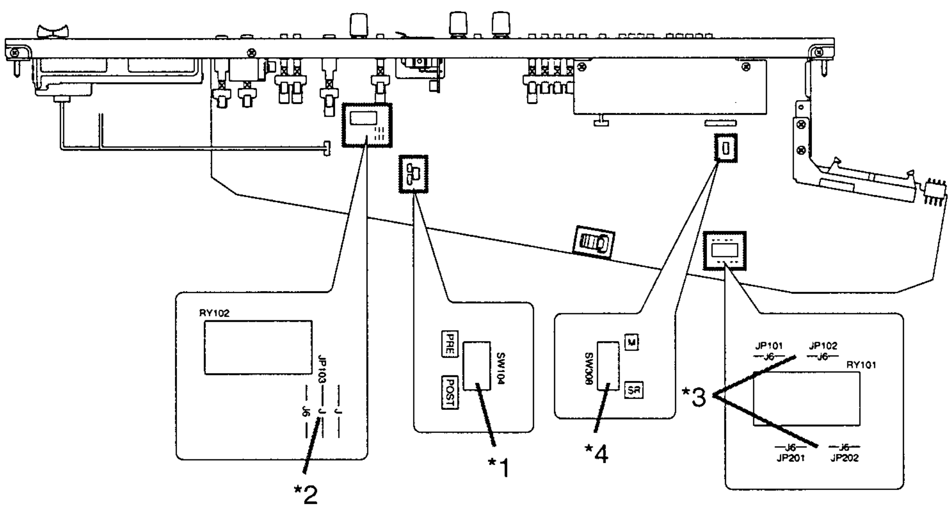
This pair of jumpers can change the group-to-matrix switch [56] (page 24) function from its factory setting of post-group fader [58] (page 25), to pre-fader. In other words, if these jumpers are moved, the group-to-matrix level is independent of the group fader setting.

### 7.5.4 Group master module 2: matrix CUE AFL pre- /post-ON jumpers
As shipped, matrix CUE AFL levels are derived post-ON switch. If this jumper is changed, the matrix CUE AFL levels are independent of the matrix ON switches.

### 7.5.5 Group master module 2: group CUE AFL pre- /post-ON jumpers
As shipped, group CUE AFL levels are derived post-ON switch. If these jumpers are changed, the group CUE AFL levels are independent of the group ON switches.

### 7.5.6 Group master module 2: group-to-stereo pre- / post ON jumpers
This pair of jumpers can change the group-to-matrix switch [56] (page 24) function from its factory setting of post-ON switch [60] (page 25), to pre-ON. In other words, if these jumpers are moved, the group-to-matrix switches are independent of the ON switch status.

## 7.6 Stereo master module




The stereo master module can have the following settings changed by switches or jumpers mounted on the module board:

| | Setting | Options | Factory setting |
|---|---|---|---|
| *1 | Stereo-to-matrix | pre-fader/ post-fader | post-fader |
| *2 | Stereo-to-matrix | pre-ON / post-ON | post-ON |
| *3 | Stereo CUE AFL | pre-ON / post-ON | post-ON |
| *4 | To GRP level | 0dB / -6dB | 0dB |

### 7.6.1 Stereo master module: stereo-to-matrix pre- /post-fader jumpers
This pair of jumpers can change the group-to-matrix switch [73] (page 27) function from its factory setting of post-stereo fader [78] (page 27), to pre-fader. In other words, if these jumpers are moved, the stereo-to-matrix level is independent of the stereo fader setting.

### 7.6.2 Stereo master module: stereo-to-matrix pre- / post ON jumpers
This pair of jumpers can change the group-to-matrix switch [73] (page 27) function from its factory setting of post-ON switch [76] (page 27), to pre-ON. In other words, if these jumpers are moved, the stereo-to-matrix switch is independent of the ON switch status.

### 7.6.3 Stereo master module: stereo CUE AFL pre- /post-ON jumpers

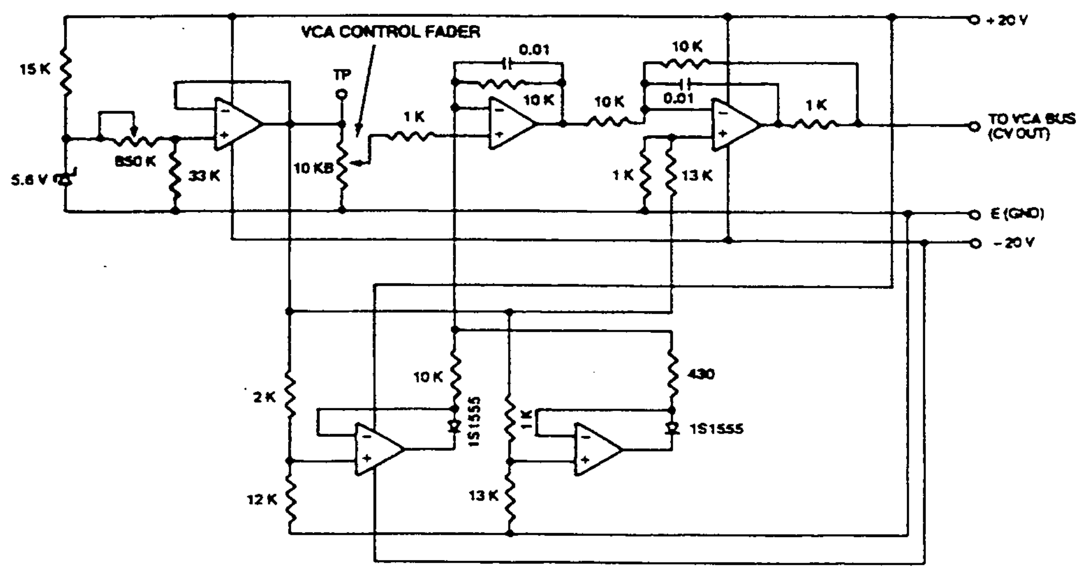
As shipped, stereo CUE AFL levels are derived post-ON switch [76] (page 27). If these jumpers are changed, the stereo CUE AFL levels are independent of the stereo ON switch setting.

### 7.6.4 Stereo master module: talkback-to-group gain (0, –6dB) switch

The stereo module can have the gain of the talkback bus routed to the group and AUX busses attenuated by 6dB when the switch shown in the illustration above is moved to the "–6dB" (monitor) position from the "0dB" (sound reinforcement) position. The factory setting is the 0dB position.

## 7.7 Hints on circuitry for remote control of the VCA masters

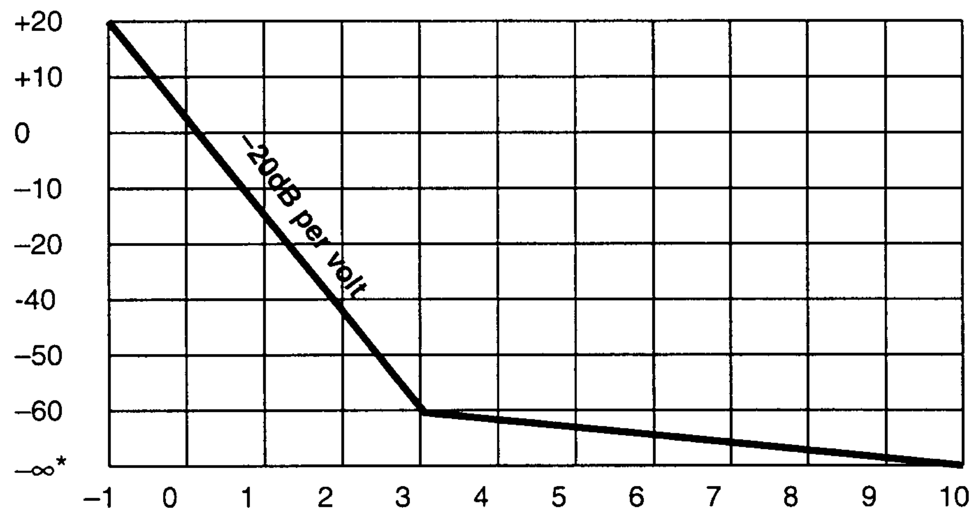
The VCA CONTROL [115] (page 35) connector on the rear panel is provided primarily so that two consoles may be linked, and just one console's VCA MASTER FADERS will affect both consoles input channels. However, it is possible to create an independent controller so that these functions can be remotely controlled from the console. One possible application would be to remotely adjust mix levels in the middle of a venue even though the console is located in a booth. Another possible application would be the creation of a limited automation system. Yamaha does not offer detailed instructions for this type of remote control. However, we do present here a schematic diagram of the VCA control fader circuit which, if constructed externally by a competent technician and interfaced via the VCA CONTROL connector, can do the job.




| Yamaha part no. | Quantity | Suffix letter | Item | Value or type |
|---|---|---|---|---|
| UA21410 | 2 | K | Mylar capacitor | 0.01µF, 50V |
| HU07543 | 1 | F | Metalized film resistor | 430Ω, 1/4W |
| HU07610 | 4 | F | Metalized film resistor | 1kΩ, 1/4W |
| HU07620 | 1 | F | Metalized film resistor | 2kΩ, 1/4W |
| HU07710 | 4 | F | Metalized film resistor | 10kΩ, 1/4W |
| HU07712 | 1 | F | Metalized film resistor | 12kΩ, 1/4W |
| HU07713 | 2 | F | Metalized film resistor | 12kΩ, 1/4W |
| HK05715 | 1 | J | Carbon resistor | 13kΩ, 1/4W |

| Yamaha part no. | Quantity | Suffix letter | Item | Value or type |
|---|---|---|---|---|
| HK05733 | 1 | J | Carbon resistor | 33kΩ, $^1/_4$W |
| IG06920 | 3 | | IC amp | MJM2041DD |
| HT56009 | 1 | B | Semi-fixed VR (trimmer) | 50kΩ |
| IF00004 | 2 | | Diode | 1S1555 |
| IF00214 | 1 | | Zener diode | RD5.6ED2 |
| VA25610 | 1 | B | Slider VR (fader) | 10kΩ |

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




*** CHANNEL ON relay opens when fader is at –∞ position**

The VCA and MUTE connections are illustrated on page 35. In order to mute a group, ground the conductor corresponding to that group. The console's VCA MASTER/SLAVE and/or MUTE MASTER/SLAVE switch(es) must be set to the SLAVE position in order for the corresponding remote control to take effect on the designated busses and mute groups.

**WARNING**

Only qualified service technicians should attempt to construct and connect any circuit to interface with the PM3500 VCA CONTROL connector. A circuit or wiring error could severely damage the console, and such damage is not covered under the terms of the PM3500 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 PM3500 Service Manual should be consulted before actually building any remote control device.

# 8 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 PM3500.

## 8.1 Console gain structure

In the GAIN STRUCTURE AND LEVELS section of this manual on page 79, we discuss some general considerations regarding levels and system setup. What of the proper gain structure within the PM3500? 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.

### 8.1.1 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 on page 41, the nominal input level ranges from -70dBu (0.25mV) to +10dBu (2.4V). These are the levels that will supply the ideal signal level throughout the module with the PAD set to 0dB or -30dB, the input GAIN control as required, fader set to its nominal position, and no VCA groups assigned. Actually, a wider range of levels can be accommodated if the fader is adjusted to other-than-nominal position; from -90dBu (0.025mV) minimum to +24dBu (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 extreme cases, 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 trade-offs 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 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?

### 8.1.2 What affects gain structure?

Firstly, it is important to 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.

### 8.1.3 Establishing the correct input channel settings

In the case of the PM3500, the input channel LEDs [25] (page 17) make it relatively simple to obtain the correct gain structure at the input stage. Begin with the PAD set at maximum attenuation (-30dB), the GAIN control centered, and apply the typical input signal to the channel input. If none of the LEDs are illuminated, or perhaps just the SIGNAL LED, disengage the attenuation PAD switch to remove the 30dB of attenuation. Adjust the GAIN control as required so that the red PEAK LED flashes on only occasionally, during the loudest program peaks, and the NOMINAL LED flashes frequently or remains on. This establishes the correct channel sensitivity for the initial setup (you may wish to alter these values during an actual program mix, as explained in subsequent paragraphs).

**NOTE**

**It is a good idea to set the group master and stereo master faders at a very low level during the initial stages of setup. This will prevent uncomfortable or even dangerously loud signals from reaching the outputs while preliminary mix setup is established.**

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

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

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

This same procedure should be followed for each input channel. Once this is done, the bus levels can be examined. Set the VU meter assign switches [88] (page 29) to examine the stereo bus levels and, where applicable, the GROUP OUT levels. One bus at a time, monitor the group mix (use the headphones and the corresponding group CUE switch), and create a rough mix of all input channels which feed this group. Bring down the input faders (or individual input channel-to-bus assign controls) for those sources which are too prominent in the mix; avoid raising input faders to make other sources more prominent. Once this rough mix is established, raise the corresponding group master fader to the nominal position (0dB on the scale). If the signal level on any of these busses becomes too hot (red meter LED flashing on more than occasionally or VU meter pegged at the top of the scale), do not back off the group master fader. Instead, pull down all the input channel faders (or turn down all input channel-to-bus assign controls) which feed this group by an equal amount. (If the channels also happen to be assigned to a given VCA master, you can pull down that VCA master, which, in turn, will reduce the signals applied to the group bus). This will leave the group master fader at the desired nominal position, will preserve the desired balance between input channels, and will keep the bus level from being too hot. Finally, release the group CUE switch.

This same procedure applies to setting the stereo master levels where input channels are assigned directly to the stereo bus. However, you will have to adjust the group master levels on any groups which are assigned to the stereo bus.

### 8.1.4 Establishing the correct group master settings

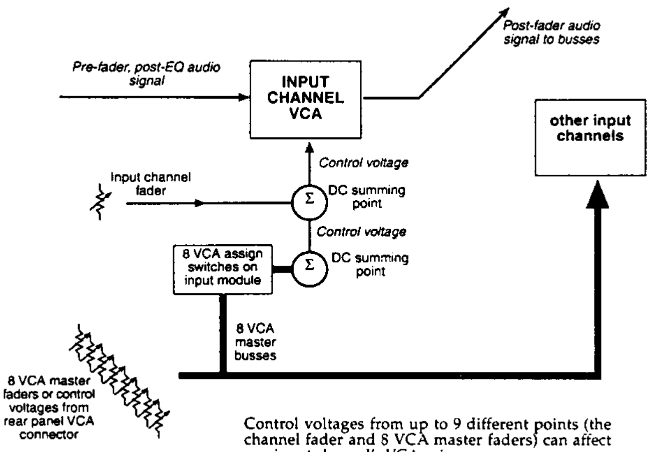
Follow the same procedure for each of the other group masters. Once all group masters are calibrated in this manner, the stereo mix can be similarly calibrated. Any group outputs which are to be applied to the stereo mix should be so assigned. Any input channels which are to be applied directly to the stereo mix should be so assigned. Monitor each stereo mix by engaging the stereo CUE switch [77] (page 27), and adjust the various stereo PAN pots as desired.

If you're not sure about the stereo position of a given group mix in the stereo perspective, you can temporarily cue that group by pressing its CUE switch. If you want to cue several group busses together, make sure LAST CUE mode is not selected (see "Cue groups and cue priority" on page 16).

With the various signals applied to the stereo mix, bring up the stereo master faders 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 master faders (if the group-to-stereo switches are engaged, or lower the input channel faders or channel-to-stereo assign controls (if the input channels' direct-to-stereo assign capability is in use). Lower all the affected source faders by a similar amount so as to preserve the mix balance.

### 8.1.5 How VCA control affects gain structure

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




Control voltages from up to 9 different points (the channel fader and 8 VCA master faders) can affect any input channel's VCA gain.

**CAUTION**

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

### 8.1.6 Channel muting and gain structure

As pointed out earlier, adding inputs to a mix will increase mix levels. If optimum mix levels are established with some input channels muted, and those channels are later turned on (either with the channel ON switch or through the scene memories), then the bus levels may increase unacceptably, and the levels of all input channels applied to the offending bus or busses may have to be reduced. Similarly, if some groups are added to the stereo master mix after those gains have been calibrated, then stereo bus levels may increase unacceptably, requiring either a reduction in all group master levels or minor adjustments of the stereo master faders.

## 8.2 Further hints & conceptual notes

### 8.2.1 What is a VCA, and why is it used?

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

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

An alternative approach involves the use of a VCA. In the PM3500, there is one VCA in each input module. That VCA takes the place of the post-fader booster amplifier in a conventional console configuration. The PM3500 channel fader is a variable resistor, but it does not have audio flowing through it. Instead, it adjusts a DC voltage output (from 0 volts at nominal position, to -0.5 volts at maximum gain, to +10 volts at "infinite" attenuation position). The DC output voltage from the channel fader is applied to the channel's VCA control input.

The VCA is a special amplifier that is designed to operate at unity gain when the fader is at nominal position, can provide some gain with the channel and/or VCA master faders set above nominal, but primarily is designed to attenuate the signal as the fader is lowered (you can think of "VCA" as "Voltage Controlled Attenuator", although technically that is a distinctly different device). So far, there is no big advantage to this

VCA approach over the conventional console, where the audio flows through the channel fader.

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

For reasons described in Section 7.2.2, conventional group master faders are also provided in the PM3500.

### 8.2.2 The distinction between the group busses and the VCA master "groups"

The PM3500 provides the operator with two different means to control multiple input channels from a single fader. One approach is to assign multiple inputs to a given group with the input-to-group assign controls [1] (page 11), and to then use the group master fader [58] (page 25) to control those signals. With this approach, the actual audio output signal from each of the assigned input channels is applied to a bus wire via 18K ohm summing/isolation resistors. The signal on the group bus is then fed into a combining (summing) amplifier in the Master module, is routed through the GROUP INSERT IN/OUT jacks [105] (page 33), is then controlled by the group master fader, and is fed to GROUP OUT [110] (page 34) and any other post-group master fader circuits.

An alternativee approach to control multiple input channels from a single fader is to use the VCA system. The audio signal in each input channel does not actually pass through the channel fader [19] (page 15). Instead, that fader applies a DC control voltage to a VCA (Voltage Controlled Amplifier) in the input module. The audio signal flowing through that VCA is, in turn, increased or decreased in level according to the control voltage applied to the VCA. One advantage of the VCA is that the control voltage applied to it can come from more than one point. In fact, when one or more of the input channel's VCA ASSIGN switches [18] (page 15) is engaged, control voltage from the correspondingly numbered VCA master faders [50] (page 23) is also applied to the channel VCA. The circuitry is such that the VCA master will cause the assigned input channel(s) post-fader output levels to ride up and down, scaled to the channel fader setting. Of course, the channel(s) output signal must still be assigned somewhere.

**NOTE**

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

If the signal on several channels is assigned directly to the stereo bus using the channels' stereo assign controls [3] (page 11), then the VCA master to which those channels are assigned will act like a group-to-stereo fader. If the channels' output is assigned to a group bus using a group assign control [1] (page 11), then the VCA master fader [50] (page 23) to which those channels are assigned will control the level applied to the group master fader [58] (page 25), which is somewhat redundant but does serve some useful purposes.

What cannot be done with a group master fader that can be done with a VCA master fader is controlling the post-fader group assign levels from groups of input channels. While it is true that group master faders affect the overall bus output level on the group busses, each of these busses can be considered a discrete output. Of the many input channel controls that may be feeding a given group master fader, some can be controlled by one VCA master, and others by another VCA master. Thus, when "subgrouping" is done using the VCA master faders, the output of affected input channels is controlled more completely. That is, the channels' group and stereo outputs are all affected by the assigned VCA master(s).

What cannot be done with a VCA master fader that can be done with a group master fader is the processing of a single, mixed signal. Consider, for example, that a given group of signals must be compressed—say the backup vocal mics or the drummer's monitor feed. If the several input channels which accommodate backup vocals are all assigned to a single group master fader, then one compressor/limiter can be inserted in the group INSERT IN/OUT patch point [105] (page 33), affecting the mixed signal on that group mixing bus. On the other hand, if those same input channels were instead controlled as a "group" by a VCA master fader, and the channel outputs were assigned to various group mixing busses, then it would be impossible to compress the backup vocal mix. Instead, multiple compressor/limiters would have to be inserted in the individual channel INSERT IN/OUT patch points [94] (page 31). The latter approach is more costly, and also applies the effect to all the channel's outputs, rather than just to a specific group.

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

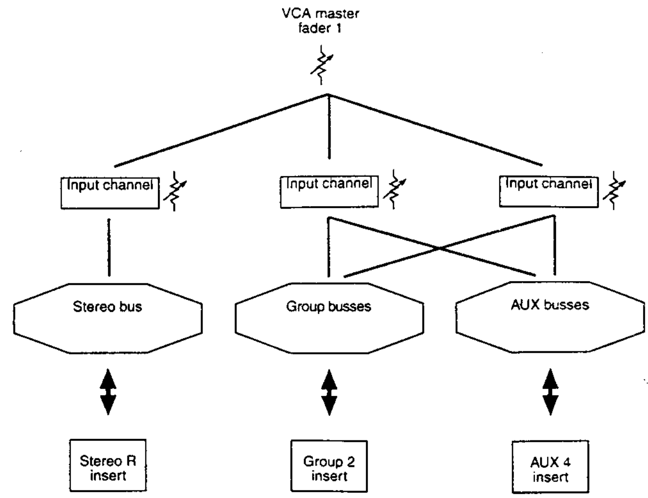
In some cases, multiple channels that are assigned direct to the stereo bus can be controlled in groups by the VCA masters, while other channels can be assigned to different group master faders, and the group masters, in turn, can be assigned to stereo.

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

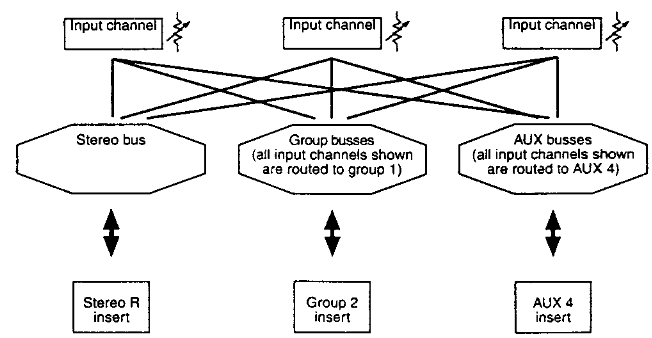
If instead of using conventional group master faders, VCA master faders were used to control the scenes, and one input was assigned to two (or more) VCA masters, the above level "build up" would not occur, and the correct balance would be retained. That's because when VCA master faders are set to nominal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

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

Ultimately, the selection of VCA or conventional group master fader assignments should be dictated by the specific requirements of the application.




In this example, the input channels and output busses have been selected arbitrarily. The VCA master fader controls the three input channels, and controls their outputs to all the busses (assuming post-fader AUX sends). However, no single INSERT point can process the whole of this VCA-controlled group of inputs.




In this example, the input channels and output busses have been selected arbitrarily. The group 1 master fader controls the post-input fader signals from all of these input channels. In the same way, the AUX 4 master send level fader controls the AUX 4 output from all of these input channels. Using this arrangement, a single effects unit can process the grouped signals if it is placed at the group or AUX insert point.

## 8.2.3 Using the channel INSERT IN jack as a line input

The input channel INSERT IN jacks [94] (page 31) are electronically balanced, line level inputs that come after the channel PAD switch and GAIN control. These jacks may be used to accommodate any balanced or unbalanced +4dBu nominal line input source. Why would one want to use the 1/4" phone jack INSERT IN rather than the XLR channel input? There are several possibilities. Certainly, the most obvious is that if the input source is equipped with a +4dBu phone jack output, then the INSERT IN jack enables a standard phone plug-to-phone plug cable to be used without any adaptor. However, the INSERT IN jack also can save time.

If the PM3500 is being used for theatre production or TV production, then CD or tape machine returns (playback from the CD machine or tape recorder) can be plugged into the INSERT IN jacks, while microphones or other line level sources can be plugged into the channel XLRs. When recording the tracks or using live mics for other purposes, the channels' PAD switches and GAIN controls can be set, as needed, for the various input sources. When playing back a CD or tape, the PAD switches and GAIN controls need not be readjusted; instead, simply engage the channel INSERT ON switches [13] (page 13) to select the recorded material. The same concept applies where the console is used for multiple stage setups (as in subsequent scenes in a theatrical presentation, or different sets for a live musical show). Provided one of the sources is a +4dBu line level source, it can be connected to the INSERT IN, and the other mic or line level source can be connected to the channel XLR; the INSERT switch then permits instantaneous selection of one or the other input source without need to disconnect and connect cables.

# 9 Applications

## 9.1 General applications

The PM3500 is designed primarily for audio mixing in live sound reinforcement applications. Its exceptional flexibility, however, will appeal to those who need a high quality mixing console for other applications, including TV show and music video production, AV audio production and general recording. We explain a few reasons below why the PM3500 is well suited to these applications, but, rather than focus on specific en-user applications, we feel it is more important to point out how some of the PM3500 sub-systems can be used to accomplish specific mixing tasks. It is up to you, as the sound engineer or mixing console operator, to use these capabilities to their best advantage in your specific situation. This manual cannot, by the nature of things, be fully comprehensive, and we expect that many users will devise unique means to connect and use the PM3500. In fact, we at Yamaha encourage you to share these special applications with us, so that we may, in turn, share the general concepts with other PM3500 users.

### 9.1.1 Theater

The PM3500 includes features that make it ideal for theatrical sound reinforcement. The 128 scene memories enable all the sound sources for a particular scene to be preset to that they can be recalled easily and conveniently (with a single switch, in the case of the first eight memories). Since the console has up to 94dB of gain, distant microphones and quiet speaking voices will present no problems. When less amplification is required, the eight VCA groups of the PM3500 make it possible to alter the balance of different groups of faders in a way impossible for conventional group faders; the VCAs can affect all the outputs from an input module, and they can control overlapping groups of inputs for "additive" or "subtractive" mixes.

The console's Mix Matrix can be used as an assignable output mixer. In the same way as some lighting mixers, the Mix Matrix allows up to 14 sources (the stereo master busses, the eight group busses, two dedicated matrix sub inputs and the stereo sub input) to be remixed into four stereo and four monaural mixes. These matrix outputs can drive various primary speaker systems, effects speaker systems, as well as lobby, dressing-room and other remote speaker systems, as well as OB feeds and the like. The inputs to the matrix can be mixed independently, as required, for each area. If simultaneous recording is required, the matrix can be set to mix signals from ahead of the group and stereo master faders, so that the group and stereo outputs can be used for independent multitrack and two-track recording mixes. Control room outputs make it possible to monitor the console outputs while working in an isolated booth—they even carry the CUE signal, so the operator doesn't even have to wear headphones.

The 40 and 48 input module versions have center masters, so that two operators can work together to handle the show conveniently. The low profile means better sightlines from a high balcony, and the rugged construction means that the console can travel reliably along with the show.

### 9.1.2 Production

It can be a challenge to get the basics of a soundtrack on tape while trying to mix sound for a live show. The PM3500 makes the task simpler by providing independent mix capabilities for the live show and the tape recording. In total, the following groups are available: eight output groups, four stereo matrix groups, four monaural matrix groups, and eight AUX busses, as well as the stereo mix. All inputs and bus outputs are balanced, low-impedance, so that long lines can be used without noise problems.

Where the extra margin of grounding isolation and common mode rejection is critical, the PM3500 can be fitted with optional transformers on a per-channel basis.

Eight group masters, the eight separate VCA groups, and the 128 scene memories with MIDI control enable the operator to keep track of the inputs, switching them on and off, and adjusting the levels with the touch of a finger...all precisely on cue. Speaking of which, the extensive cue system with its input priority allows any input or output to be examined "in place" without affecting the output signals. The solo mode, muting all but the selected input, simplifies pre-production setup and troubleshooting.

The PM3500's Mix Matrix provides a "mixer within a console". In video work, for example, the 8-track tape recorder can be fed with discrete output mixes 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 these groups, with the levels adjusted for more listenable reference monitoring or foldback. Alternatively, 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 more than eight tracks must be recorded. The matrix can also be used for monitor or foldback mixes, and if this is being done, the sub inputs can be used for effect return, so that monitoring is "wet" while the recording is "dry". The Direct Out jacks on each module make it possible to feed up to 48 tracks of a multitrack recorder with the cleanest, direct signals.

With the built-in talkback and dual monitor system (with dimming and muting during talkback), console outputs can be monitored (using an external amp and speaker system) without using headphones.

### 9.1.3 Post-production

Once a show has been photographed on video, film or multi-image media, it is time for the crucial post-production job of mixing sound effect, music and/or dialog. Sometimes there is no "original" production soundtrack, and all recording is done in the post-production phase, while at other times, the post-production task is primarily one of enhancement. The PM3500 is well suited to the task. Its many inputs can be switched to handle almost any input level, from the lowest level microphones to very "hot" electric guitars, 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 removed using the four-band parametric equalization on each input channel, as well as the sweepable high-pass filter on each input. The insert loops on each channel and bus make it possible to patch in whatever processing is required. The signal processor can be switched in or out of the loop from the front panel using the INSERT switches. The 128 scene memories allow selective channel and bus muting, MIDI-controlled if necessary, for precise synchronization, and the eight VCA groups which additively alter the signal level on any channels assigned to the VCA groups. The INSERT IN connectors can also be used to accommodate the returns from a multitrack recorder, allowing either the multitrack output or the channel input to be selected without repatching.

The mix matrix provides four stereo and four monaural output busses which can be fed from up to 14 sources (stereo, stereo sub in, matrix sub in and eight groups). This "mixer within a console" allows groups of similar instruments (or vocals) to be grouped and controlled with the group mater faders and then remix these groups. For instance, in film work, there might be mixes required for the left, center, right, surround, etc. or for stereo music, stereo dialog, stereo effects, etc. The matrix can provide these mixes easily without external patching. When the VCA groups are combined with this flexibility, a "double-group" capacity is added, extending the capabilities of the PM3500 even further.

The control-room outputs and cue facilities, together with the headphone outputs, make monitoring even more convenient. The talkback capabilities can be used for control room-to-studio communication.

### 9.1.4 Video work

Video production today uses more live music, more pre-recorded sources, and more special effects than ever before. Music videos, stereo Hi-Fi VCRs and stereo TV broadcast have elevated the importance of video sound quality. With its high quality sound, and extraordinary flexibility, the PM3500 is a logical choice for many video sound production requirements. With 24, 32, 40 or 48 input modules (the number of actual inputs increases as stereo modules are added), the PM3500 can handle the substantial numbers of mics, instruments and pre-recorded sources needed in today's video world. The comprehensive SUB facilities allow even more inputs by linking two consoles.

With the eight group busses, different groups of sources can be assigned to a group and controlled with a single fader. The stereo bus can be used for an independent, directly-assigned mix of the inputs, or it can act as a master, fed from the group faders. With the eight auxiliary busses, this gives eighteen output mixes, in addition to the matrix. The matrix allows the eight groups, the stereo mix, and two sub stereo mixes to be sub-mixed into any of four stereo and four monaural matrix outputs. These matrix outputs can be used to create live foldback mixes of the various groups so that performers can hear what's happening during the production, while the other outputs provide the recording mixes. MIDI-controllable scene and control changes allow dynamic automated muting and unmuting of signals as required, synchronized to SMPTE cues on tape if necessary. The VCA busses allow additive mixing of any inputs assigned to them, giving even more control over the mix.

The control room outputs can be used to feed a local monitoring system, and the input cue priority allows the operator to check any input channel or auxiliary return at the touch of a single switch. The talkback system allows engineers and producers to communicate with the rest of the staff involved in the production.

With the eight AUX sends, four of which can be used as stereo sends, the most sophisticated of today's signal processors can be incorporated in the system along with the PM35000. Input channels are used for the returns, meaning that there's no compromise in the treatment of these signals.

### 9.1.5 Sound reinforcement

The PM3500's electronically balanced inputs are of extremely high quality, and optional input transformers can be fitted if required. With the wide-ranging pad and gain controls, together with each input's nominal indicator, gain settings can be made quickly and accurately to confirm the nominal position on each input. Gain is there when it's needed, and when not required, the noise is minimized. The 4-band fully parametric equalization on all input modules, together with the sweepable high-pass filter, allow precise tailoring of the sources.

There are eight groups which can be used for sub-mixing vocal and/or instrumental sections, and these can be mixed to mono or stereo house feeds using the stereo bus, the four stereo matrix busses, and the four monaural matrix busses. If the Mix Matrix is used to feed the house, the stereo bus can be used as two additional group busses. With eight AUX sends, all switchable between pre- and post-fader, there is no shortage of effect or foldback monitor sends. Full control over effect returns can be achieved by routing these through the stereo input modules. With the VCA masters, even more flexibility is provided for group mixing: a VCA group can affect all outputs from a given set of input channels, for instance.

The 128 scene memories and MIDI capability of the PM3500 allow dynamic muting and enabling of more than just the input channels. The check capability and "off-line editing" also help to prevent embarrassing mistakes in scene recall.

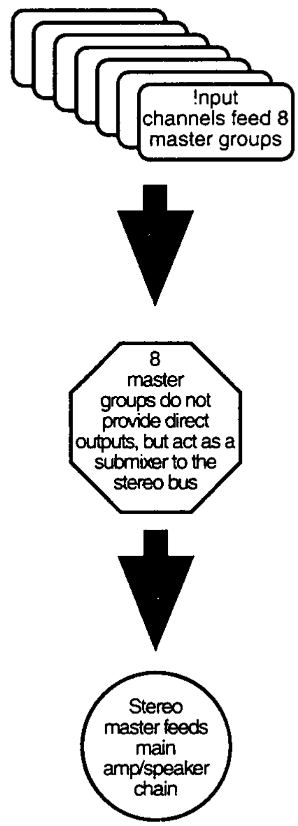
Above all, the PM3500 is practical—all controls are logically located, and their status shown by reliable LEDs where appropriate. Without providing a bewildering array of bargraph LEDs, the signal level of each input can be seen with the three indicators where it needs to be seen—by the channel fader. The master VU meters with integral peak level indicators are large, clear, and provide excellent monitoring of the console status.

Extensive cue capabilities with a pre-show solo mode, talkback and a lineup oscillator make the PM3500 a supremely practical choice for the road, as do the aluminum chassis with aircraft-style ribs and braces, a low profile which blocks fewer seats in the house, and the reliable, rugged power supply.

## 9.2 Setup concepts

### 9.2.1 Deriving a stereo mix from groups 1 through 8

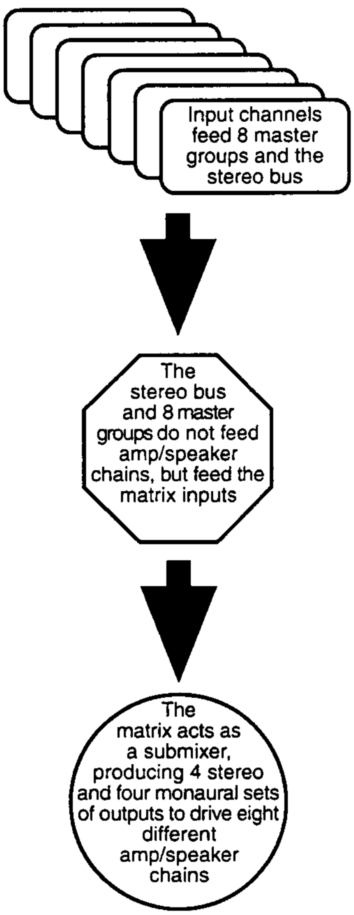
There are a number of ways to obtain a stereo mix from this console. One technique is to use groups 1 through 8 for subgrouping input channels. The post-group master fader signals [57] (page 20) can be assigned to the stereo bus using the group-to-stereo switches [54] (page 19) and pan controls. The stereo master faders then become the overall stereo output for the mixed groups. Here, the channel-to-stereo switches are not used, except for those channels being used as effect returns being routed directly to the stereo bus. This is a very straightforward way of producing a stereo mix (or dual mono mixes) without involving the VCA groups or the mix matrix.



## 9.2.2 Using the mix matrix with the subgroups and stereo bus to function as 10 subgroups

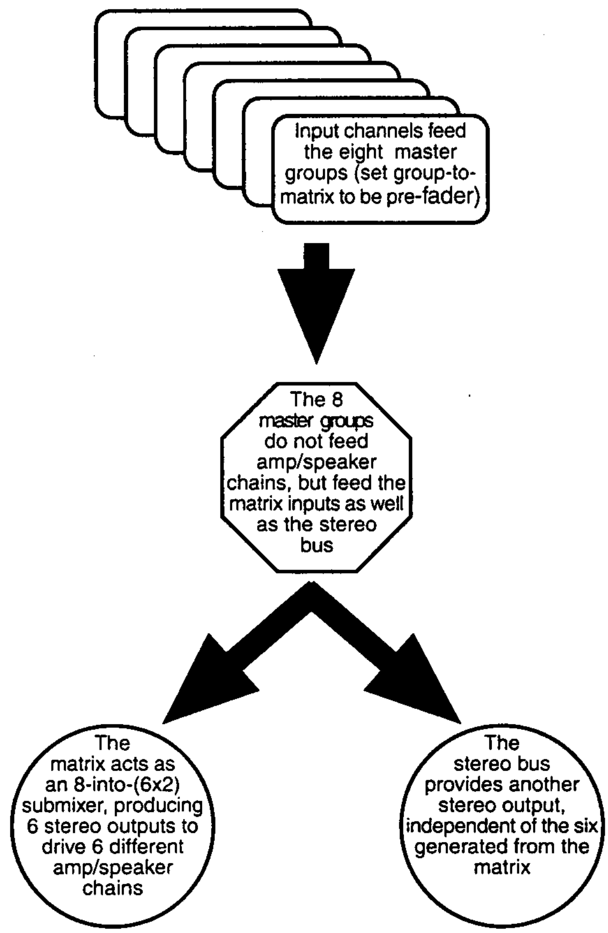
The mix matrix can be used to create up to four stereo mixes and four monaural mixes from the group and stereo busses (10 in all). In this case, input channels may be directly assigned to the stereo bus [3] (page 10) and use the panpot to push the signal hard left or hard right to the appropriate stereo bus. The two stereo master faders can then be treated as two more group faders. Make sure that the groups are not assigned to the stereo bus if you are using the console in this way.

The mix matrix can then be used to provide a 10 into (4x2) + (4x1) mix.




## 9.2.3 Creating 7 independent stereo mixes (or 14 independent mono mixes) with the mix matrix

In this application the internal jumpers must be reset so that group-to-matrix feeds are derived pre-group fader rather than post-group fader—see "Optional functions" on page 59. The eight group busses can then be assigned to the stereo bus [54] (page 19) and also assigned to the matrix [55] (page 19). This allows the balance of the group signals in the stereo mix to be adjusted using the group master faders, and the balance of the group signals in the matrix groups to be adjusted using the matrix level controls [52] (page 19). The first four matrix groups are stereo, and the remaining four can be used as two stereo pairs. Alternatively, the groups can be panned hard right and hard left to the stereo bus and inside the matrix, allowing the stereo bus to function as two mono busses and the matrix to be used as 12 mono busses—14 mono busses in total.

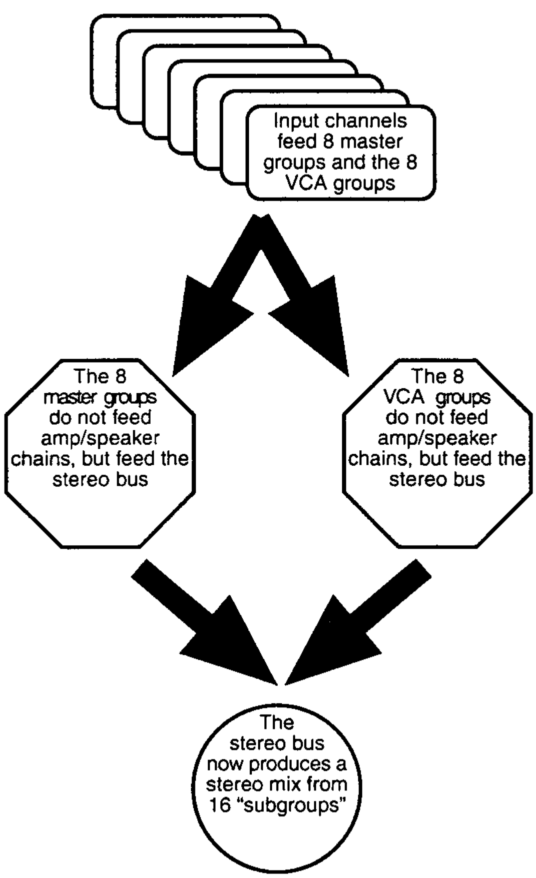


### 9.2.4 Using the VCA masters and the group masters to obtain the functional equivalent of 16 subgroups

For this exercise, we can assume that the object is to obtain a stereo output (or a pair of mono outputs). Some of the input channels can be assigned to the group busses using the group assign switches [1] (page 10). These groups are then controlled by the group master faders, and are routed to the stereo bus using the group-to-stereo switches [54] (page 19). At the same time, other groups are not assigned to the master groups, but are routed directly to the stereo bus. To control these inputs in a group fashion, they are assigned to the VCA groups (only one VCA group per input). The group levels for these inputs can then be controlled by the VCA faders. In this way, the console provides sixteen subgroups, all routed to the stereo (or dual mono) outputs.

**NOTE**

**If there are any groups requiring "blanket" processing of the whole group, such as compression of a drum group or reverb treatment of a vocal group, these should be assigned to the groups, rather than the VCA groups. This is because the groups have INSERT points [104] (page 26) which can be used for this treatment, where the VCA groups lack this facility.**




### 9.2.5 Using the VCA masters to control the same input channels in order to handle different scenes

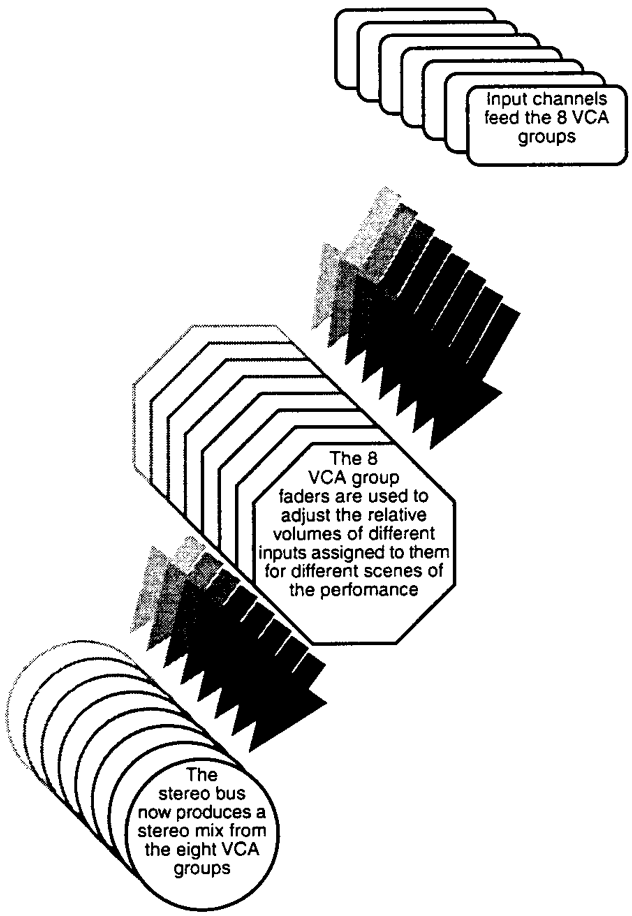
In a multi-scene theatrical presentation, or a multi-set concert, for example, there may well be a need for the same input channels to be mixed at different levels to meet the differing stage requirements. It is, of course, possible to make copious notes, and painstakingly adjust (with considerable effort!) 40 or 48 faders. However, the PM3500's VCA groups make such superhuman effort unnecessary.

The eight VCA faders may be thought of here as eight "scene" controllers. The actual output mix and speaker assignments can be handled as usual by the group and matrix busses. However, the VCA faders will determine the channels that actually contribute to the console outputs at any given time.

If an input channel is needed for only one scene, than that channel should be assigned to the VCA group used to control the levels for that scene. If the group is needed for more than one scene, then more than one VCA group may need to be assigned. Of course, the number of scenes can be more than eight, as some scenes may require more than one VCA fader to be brought up, others may just require one, or may require different settings. The end result is that only eight VCA master faders, rather than 40 or 48 channel faders, need be monitored.

Of course, the scene memories can also be used to mute unwanted groups of channels (brass section mics, for example, on numbers with no brass).

An interesting conceptual example of VCA control involves a group of input channels that are assigned to the left and right sides of a stereo mix. VCA master 1 can be used to control those channels assigned primarily to the left side of the mix, VCA master 2 controls those assigned to the right, and all these channels are also assigned to VCA 3. In this way, overall stereo fades can be made using VCA master 3, and the overall balance between left and right can be made using VCA master faders 1 and 2. This may or may not be a solution which meets your specific requirements, but we offer it as an example of how the VCAs can be used to help you in your work in ways other than the obvious ones.



# 10 Maintenance

## 10.1 Cleaning the console

### 10.1.1The 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.

### 10.1.2Power supply air filters

The reticulated foam air filters on the front of the power supply screen 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 (hex) wrench, remove the four cap screws that secure each front grille. The foam elements may now be removed and rinsed in cool water. For greasy or stubborn dirt, dip the elements in a mild solution of detergent and water, then rinse with clear water. Blot and/or air dry the elements thoroughly before returning them to the amplifier. DO NOT USE SOLVENTS TO CLEAN THE FOAM ELEMENTS.

### 10.1.3Pots 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.

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 P3500 dealer or service center.

### 10.1.4The 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.

**NOTE**

**For module removal and replacement see page 85.**

### 10.1.5Meter lamp replacement

The VU meters and meter-assign indicators are illuminated by LEDs which should not require replacement.

Contact your Yamaha dealer or service facility should a meter illumination LED fail.

## 10.2 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 PM3500 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 channels 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 switch engaged. | Release master SOLO mode to activate all channels which should be on. |
| | The affected input channel does not have its ON switch on | Turn on the channel(s) |
| Certain input channels or groups of channels cannor be heard at group, stereo, post-fader aux sends or matrix outputs | The affected input channel(s) have VCA assign switches engaged, and the VCA fader(s) to which the channel(s) are assigned are set to minimum, | De-assign the channel(s) from the VCA group(s) or raise the appropriate VCA master fader(s). |
| | The affected input channel(s) have VCA assign switches engaged, and the remote VCA connection is causing the VCA master level to go to minimum | Disconnect the VCA connector. If the output is restored, check the remote device. |
| Certain input channels or groups of channels cannot be heard at group outputs, group-to-stereo outputs or group-to-matrix outputs. | The affected input channels are assigned to a group fader which is set to minimum level, and the group-to-stereo outputs or group-to-matrix feeds are set to post-fader | Raise the appropriate group master fader |
| An individual input channel cannot be heard group, aux or matrix outputs | Channel ON/off switch is off, or its PAD and GAIN controls are set so that the input sensitivity are too low. | Turn the channel on. Set the pad for a lower value and/or the gain at a higher value. |
| | The channel insert switch is engaged, and a plug is connected at the channel's INSERT IN jack, but no signal is applied to that plug. | Disengage the channel's insert switch or chack the signal at the INSERT IN jack. |
| | A phantom-powered condenser microphone or direct box is connected to the channel and is not receiving phantom power. | Check that the channel's and master 48V switches are on. |
| There is no output, and no console functions work at all. | Power is not reaching the PM3500. | Make sure that the PW4000 power supply is on, and that the umbilical cables are properly connected. Check the fuses and the AC voltage. |
| Fuses are OK, and the power supply turns on, but the console does not turn on. | The umbilical cord is not connected correctly. | Reconnnect the umbilical cors. |

## 10.3 What to do in case of trouble

The PM3500 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 PM3500 dealer.

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