3 Specifications

3.1 PM3500M mixing console general specifications

Total Harmonic Distortion (Master Output)	< 0.1% (THD+N)	20Hz – 20kHz @ +14dBU, 600Ω
	<0.01% (2nd to 10th harmonics)	20Hz – 20kHz @ +14dBU, 600Ω
Frequency response (Master Output)	$0\pm{}^{1}/_{3}$dB	20Hz – 20kHz @ +4dBU, 600Ω
Hum and Noise (52CH) (20Hz – 20kHz) Rs=150Ω Input Gain = Max. Input Pad = OFF Input sensitivity = –70dB * Hum and noise are measured with a 6dB/ octave filter @ 12.7kHz; equvalent to a 20kHz filter with infinite dB/ octave attenuation	–128dB	Equivalent input noise
	–98dB	Residual output noise
	–54dB (58dB S/N)	GROUP OUT Master fader and one channel fader at nominal
	–84dB (88dB S/N)	STEREO OUT Master fader at nominal and all channel assign switches off and all group to stereo switches off
	–81dB (85dB S/N)	AUX OUT Master fader at nominal and all channel AUX mix switches off
	–92dB (96dB S/N)	MATRIX OUT Master level control at nominal and all Matrix mix controls at minimum
	–81dB (85dB S/N)	GROUP OUT Master fader at nominal and all group mix switches off
Crosstalk	–80dB @ 1kHz, –70dB @ 10kHz	adjacent inputs
	–80dB @ 1kHz, –70dB @ 10kHz	input to output
Maximum voltage gain	74dB	CH IN to CH INSERT OUT
	90dB	CH IN to AUX OUT (pre-fader)
	100dB	CH IN to AUX OUT (post-fader)
	84dB	CH IN to MONITOR OUT (INPUT CUE)
	64dB	TALKBACK IN to TALKBACK OUT
	10dB	SUB IN to OUT
	10dB	2TR IN to MONITOR OUT
	74dB	CH IN to DIRECT OUT
	90dB	CH IN to GROUP OUT (pre-fader)
	100dB	CH IN to GROUP OUT (post-fader)
	100dB	CH IN to STEREO OUT (CH to ST)
	110dB	CH IN to STEREO OUT (GROUP to ST)
	110dB	CH IN to MATRIX OUT (GROUP to MATRIX)
Mono input PAD switch	30dB	
Mono input gain control	50dB variable	
VCA cue gain trim	20dB (–14dB to +6dB)	
PFL (Input cue) gain trim	20dB (–14dB to +6dB)	

3–Specifications

Input high-pass filter		12dB/octave roll-off	below 20 – 400Hz at –3dB point
Channel equalization ±15dB max		1kHz – 20kHz	HIGH (shelving/peaking, Q= 0.5 – 3)
		400Hz – 8kHz	HIGH MID (peaking, Q= 0.5 – 3)
		80Hz – 1.6kHz	LOW MID (peaking, Q= 0.5 – 3)
		30Hz – 600Hz	LOW (shelving/peaking, Q= 0.5 – 3
Talkback high-pass filter		12dB/octave roll-off	80Hz at –3dB point
Oscillator/noise		Sine wave @ 100Hz, 1kHz, 10kHz or pink noise	Sweepable from 0.2 to 2 times nominal frequency; less than 1% THD @ + 4dB output
Phantom power		+48V DC applied to balanced inputs	via 6.8kΩ current-limiting isolation resistors. Rear-panel PHANTOM MASTER switch turns all on or off. When Master is ON, individual channels' phantom power may be switched with channels' PHANTOM switches
Channel indicators		Built into each monaural and stereo input module (two each per stereo module)	
	PEAK	Red	Turns on when pre- or post-EQ level reaches 3dB below clip level
	NOMINAL	Orange	Turns on when post-EQ level reaches nominal level
	SIGNAL	Green	Turns on when post-EQ level reaches 20dB below nominal level
Σ **Peak indicators**		Red	In each GROUP, AUX and STEREO bus turns on when pre-fader level reaches 3dB below clip level
Scene memories		Direct Recall	Memories 1 through 8
		Switchable recall	Memories 1 through 128
VU meters		2 large, 12 smaller	All switchable, all illuminated, with true VU ballistics
VU meter peak indicators		In each meter (red LED)	Turns on when level reaches 3dB below clip level
Dimensions			
	Height	335mm (13.2in)	all models
	Depth	900mm (35.4 in)	all models
	Width	1822mm (71.7 in)	44-channel, center master model
		2062mm (81.2 in)	52-channel, center master model
	Weight	129kg (283.8 lb)	44-channel, center master model
		145kg (319 lb)	52-channel, center master model
Supplied accessories		PW4000 power supply	
		Umbilical cable for power supply	
		Label (ST,CH)	

3.2 Input characteristics

Connection	PAD	Gain Trim	Actual load impedance	For use with nominal	Input level[a]			Connector in Mixer[b]
					Sensitivity[c]	Nominal	Max. before clip	
CH IN (1 through ch#[d])	0	–70	3kΩ	50Ω – 600Ω mics and 600Ω lines	–96dB (12μV)	–70dB (245μV)	–48dB (3.09mV)	XLR-3-31 type
	30				–66dB (388μV)	–40dB (7.75mv)	–18dB (97.6mV)	
	0	–20			–46dB (3.88mV)	–20dB (77.5mV)	+2dB (976mV)	
	30				–16dB (123mV)	+10dB (2.45V)	+32dB (30.9V)	
TALKBACK IN			3kΩ	50Ω – 600Ω mics	–70dB (245μV)	–50dB (2.45mV)	–28dB (30.9mV)	XLR-3-31 type
2TR IN (1, 2) [L,R]			10kΩ	600Ω lines	–6dB (388mV)	+4dB (1.23V)	+26dB (15.5V)	XLR-3-31 type
GROUP (1 through 8) SUB IN			10kΩ	600Ω lines	–6dB (388mV)	+4dB (1.23V)	+26dB (15.5V)	XLR-3-31 type
STEREO [L, R] SUB IN								
AUX (1 through 8) SUB IN								
MATRIX [L, R] SUB IN								
CUE [L, R] SUB IN								
CH (1 through ch#[d]) INSERT IN			10kΩ	600Ω lines	–22dB (61.6mV)	+4dB (1.23V)	+26dB (15.5V)	Phone jacks (TRS)[e]
GROUP (1 through 8) INSERT IN					–6dB (388mV)			
STEREO [L, R] INSERT IN								
AUX (1 through 8) INSERT IN								

a. In these specifications, when dB represents a specific voltage, 0dB is referenced to 0.775Vrms
b. All XLR connectors are balanced
c. Sensitivity is the lowest level that will produce an output of +4dB (1.23V) or the nominal output level when the unit is set to maximum level
d. 44 channels or 52 channels
e. All phone jacks are balanced (T=hot, R=cold, S=Gnd)

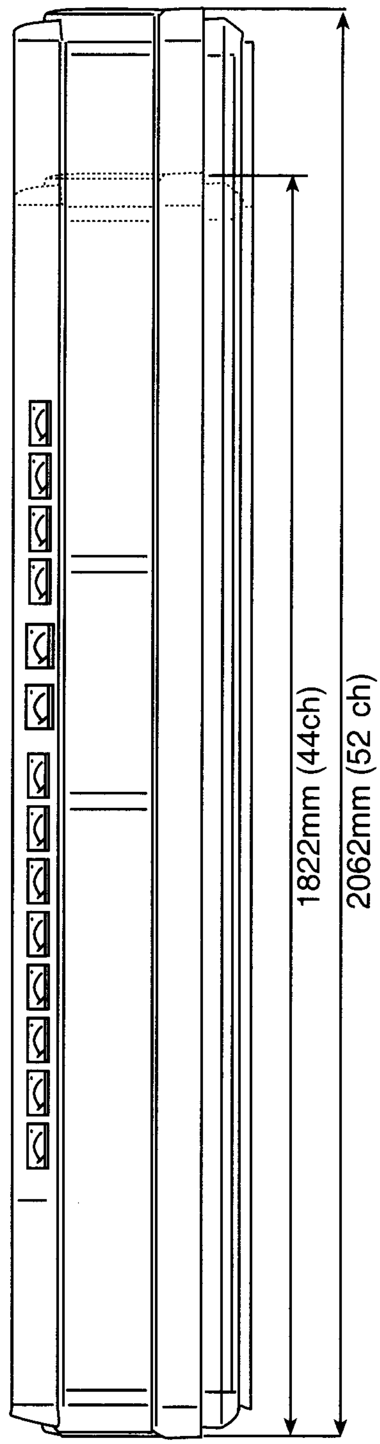
3.3 Output characteristics

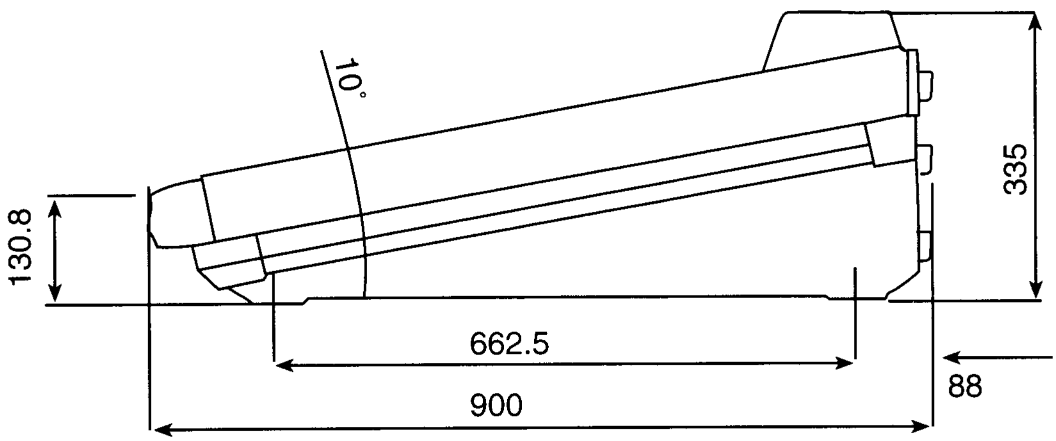
Connection	Actual Source Impedance	For use with nominal	Output level[a]		Connector in mixer
			Nominal	Max before clip	
GROUP (1 through 8) OUT STEREO [L,R] OUT AUX (1 through 8) OUT ST-MATRIX (1 through 4) [L, R] OUT MATRIX (1 through 4) OUT MONITOR (A, B) [L, R] OUT TALKBACK OUT OSC OUT	150Ω	600Ω lines	+4dB (1.23V)	+24dB (12.3V)	XLR-3-32 type[b]
CH DIRECT OUT (1 through ch#[c])	150Ω	600Ω lines	+4dB (1.23V)	+24dB (12.3V)	Phone jack (TRS)[d]
CH (1 through ch#[c]) INSERT OUT GROUP (1 through 8) INSERT OUT STEREO [L, R] INSERT OUT AUX (1 through 8) INSERT OUT	150Ω	10kΩ lines	+4dB (1.23V)	+24dB (12.3V)	Phone jack (TRS)[d]
PHONES (1, 2) [L, R] OUT	15Ω	8Ω phones	75mW	150mW	Stereo phone jack[e]
		40Ω phones	65mW	150mW	

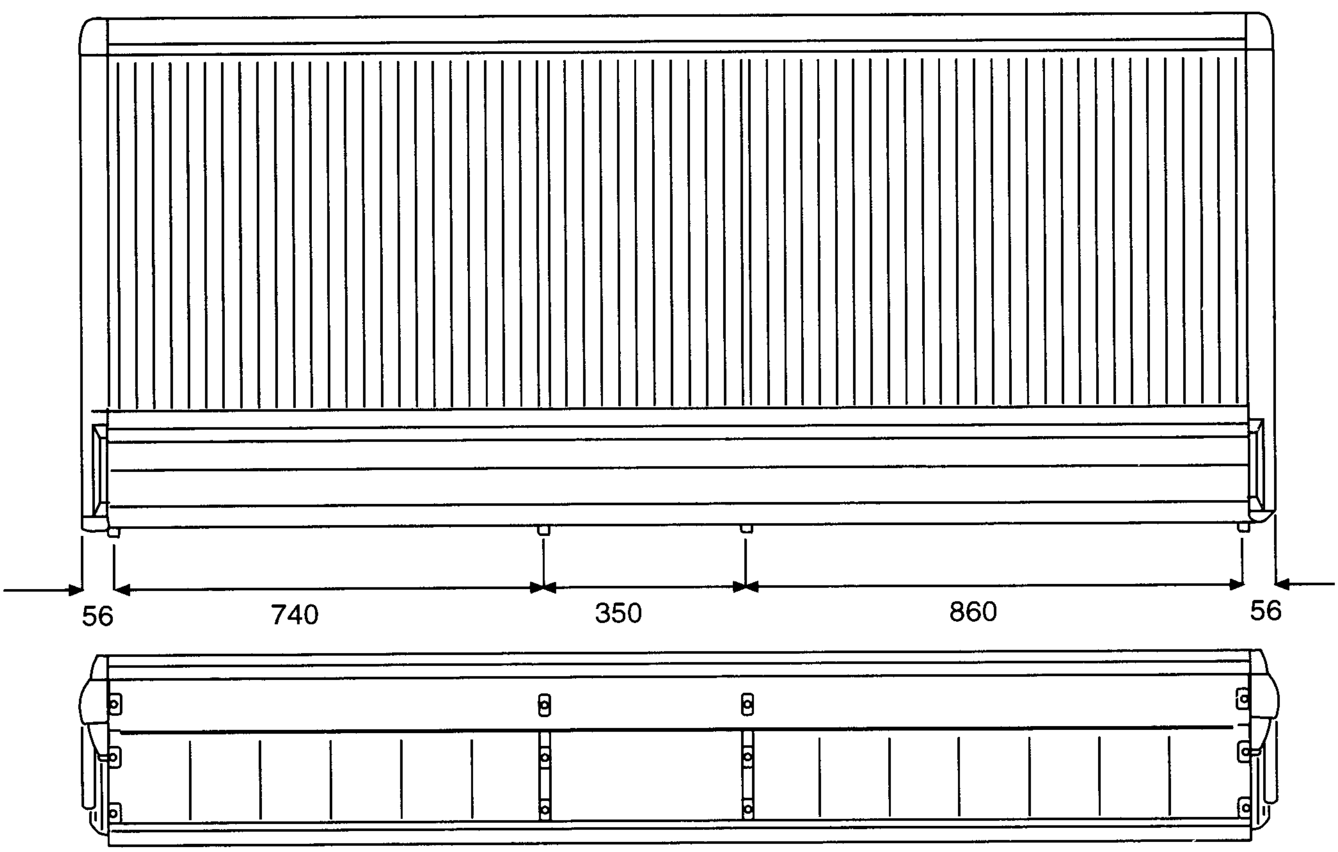
a. In these specifications, when dB represents a specific voltage, 0dB is referenced to 0.775Vrms
b. All XLR connectors are balanced
c. 44 channels or 52 channels
d. Phone jacks are balanced (T=hot, R=cold, S=Gnd)
e. Stereo phone jacks are unbalanced

For European Model
Purchaser/User Information specified in EN55103-1 and EN55103-2.
Inrush Current: not applicable
Conformed Environment: E1, E2, E3 and E4

3.4 Dimensional drawings







All dimensions in millimeters (mm)
Dimensions shown are for 52-channel center console model

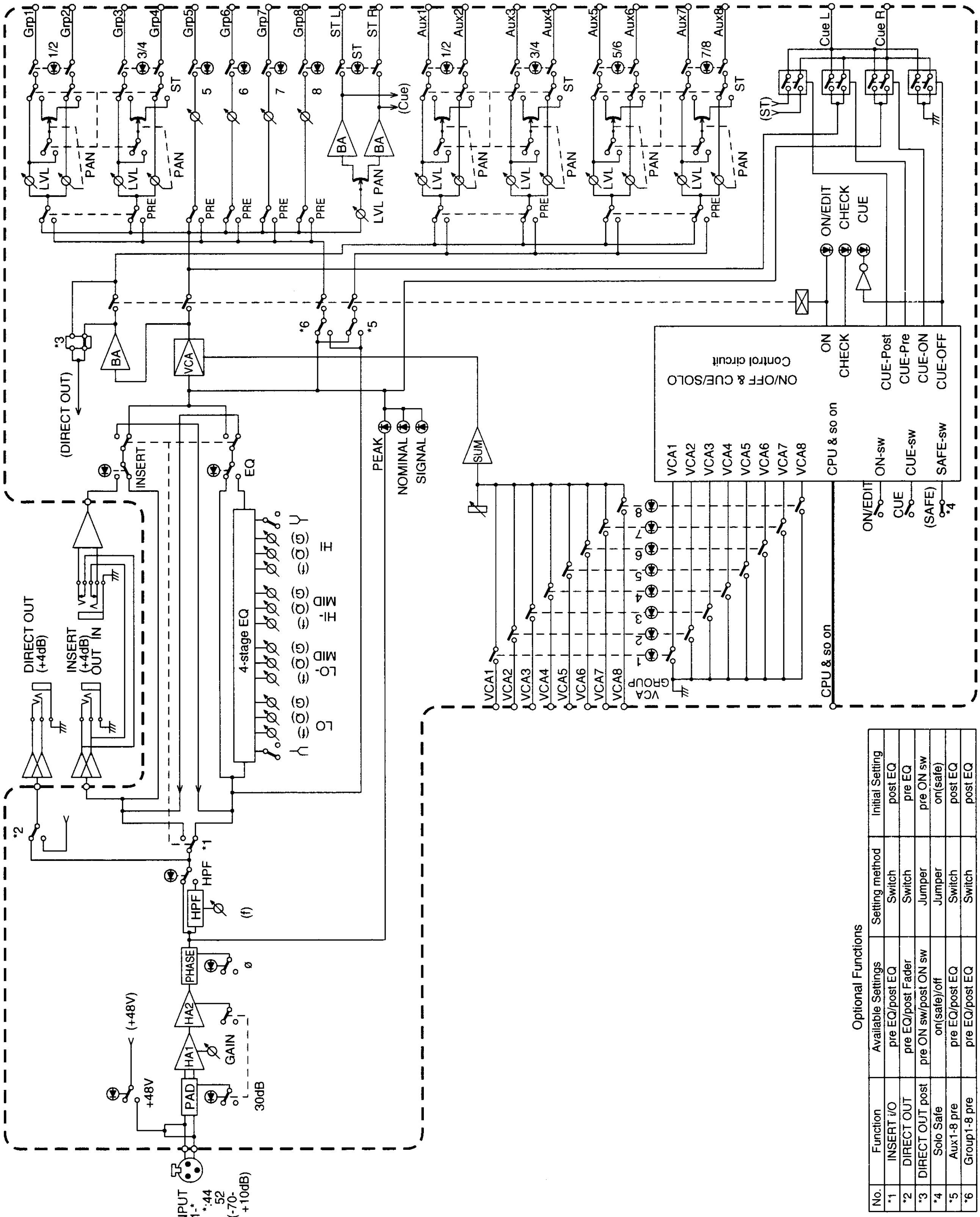
3.5 Optional equipment and parts for the PM3500M console

Model name	Model number
Monaural input module	MN3500M
Monaural input rear panel	MNRP3500
Group master module 1	GRM3500-1
Group master module 2	GRM3500-2
Stereo master module	STM3500
Monitor module	MON3500
Control module	CNT3500
Input transformer	IT3500
Blank module	BL3500

Contact your PM3500M supplier for details of availability of these parts.

3.6 Block diagrams

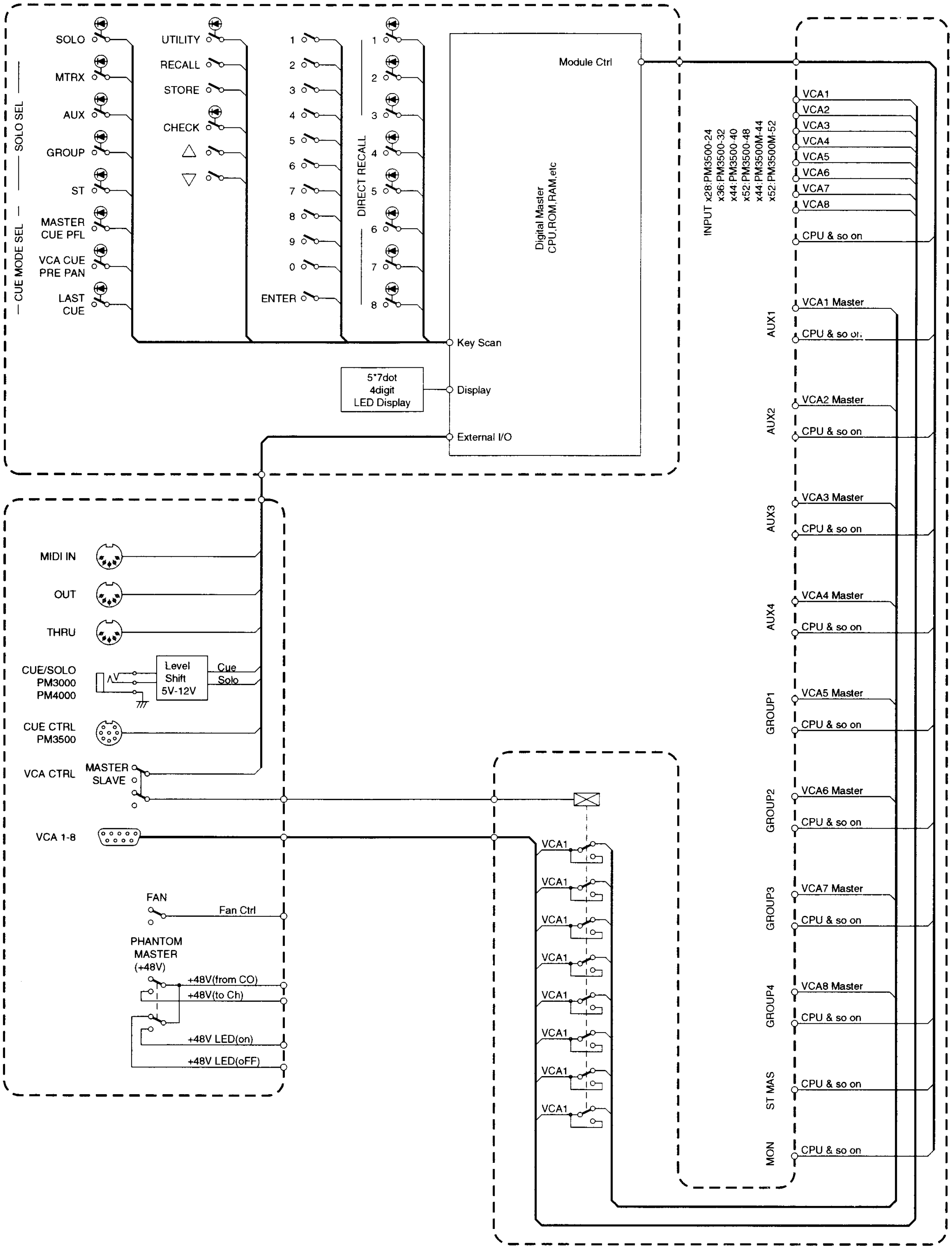
3.6.1 Monaural input module



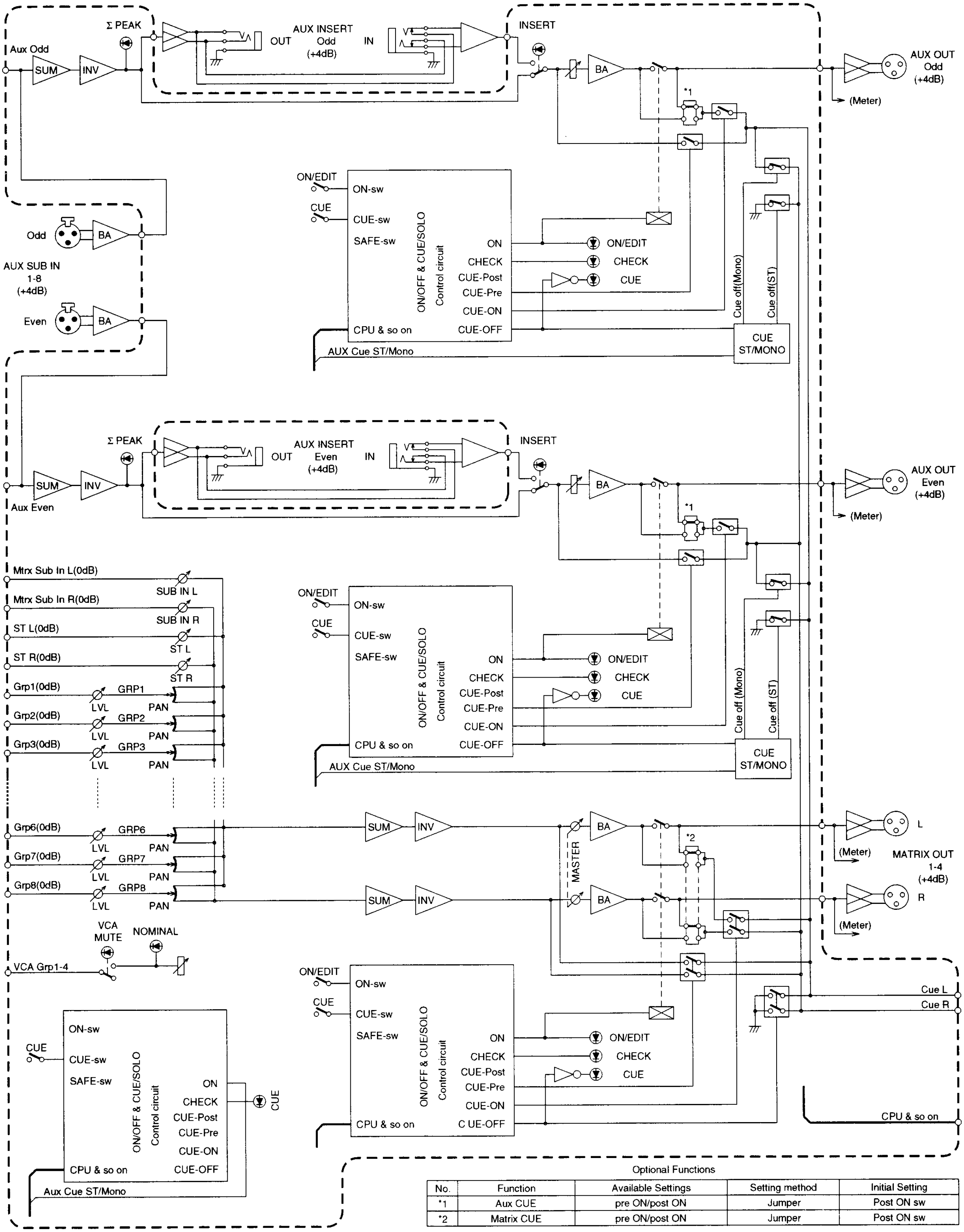
Optional Functions

No.	Function	Available Settings	Setting method	Initial Setting
*1	INSERT I/O	pre EQ/post EQ	Switch	post EQ
*2	DIRECT OUT	pre EQ/post Fader	Switch	pre EQ
*3	DIRECT OUT post	pre ON sw/post ON sw	Jumper	pre ON sw
*4	Solo Safe	on(safe)/off	Jumper	on(safe)
*5	Aux1-8 pre	pre EQ/post EQ	Switch	post EQ
*6	Group1-8 pre	pre EQ/post EQ	Switch	post EQ

3.6.2 SOLO and scene recall module



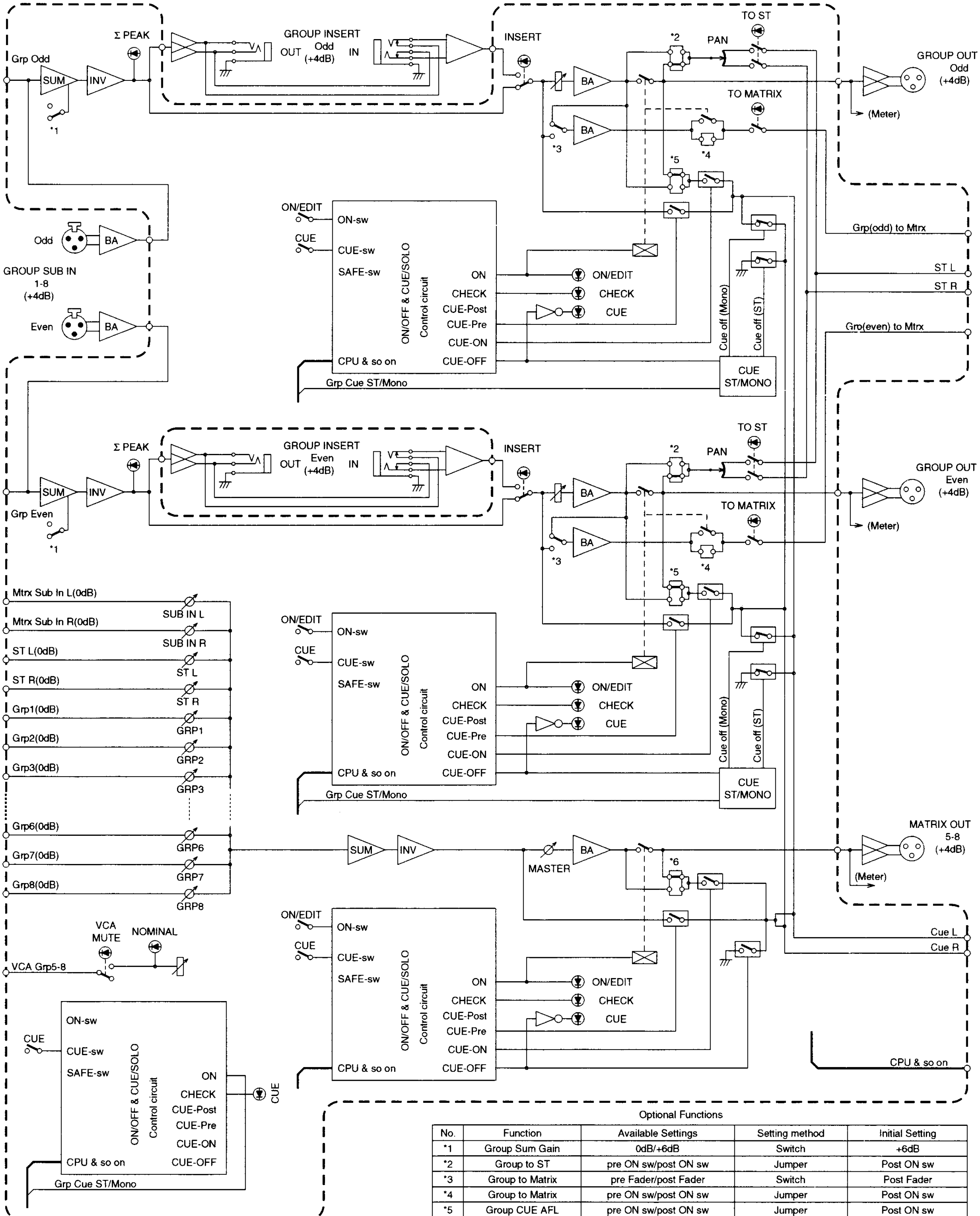
3.6.3 Group master module 1



Optional Functions

No.	Function	Available Settings	Setting method	Initial Setting
*1	Aux CUE	pre ON/post ON	Jumper	Post ON sw
*2	Matrix CUE	pre ON/post ON	Jumper	Post ON sw

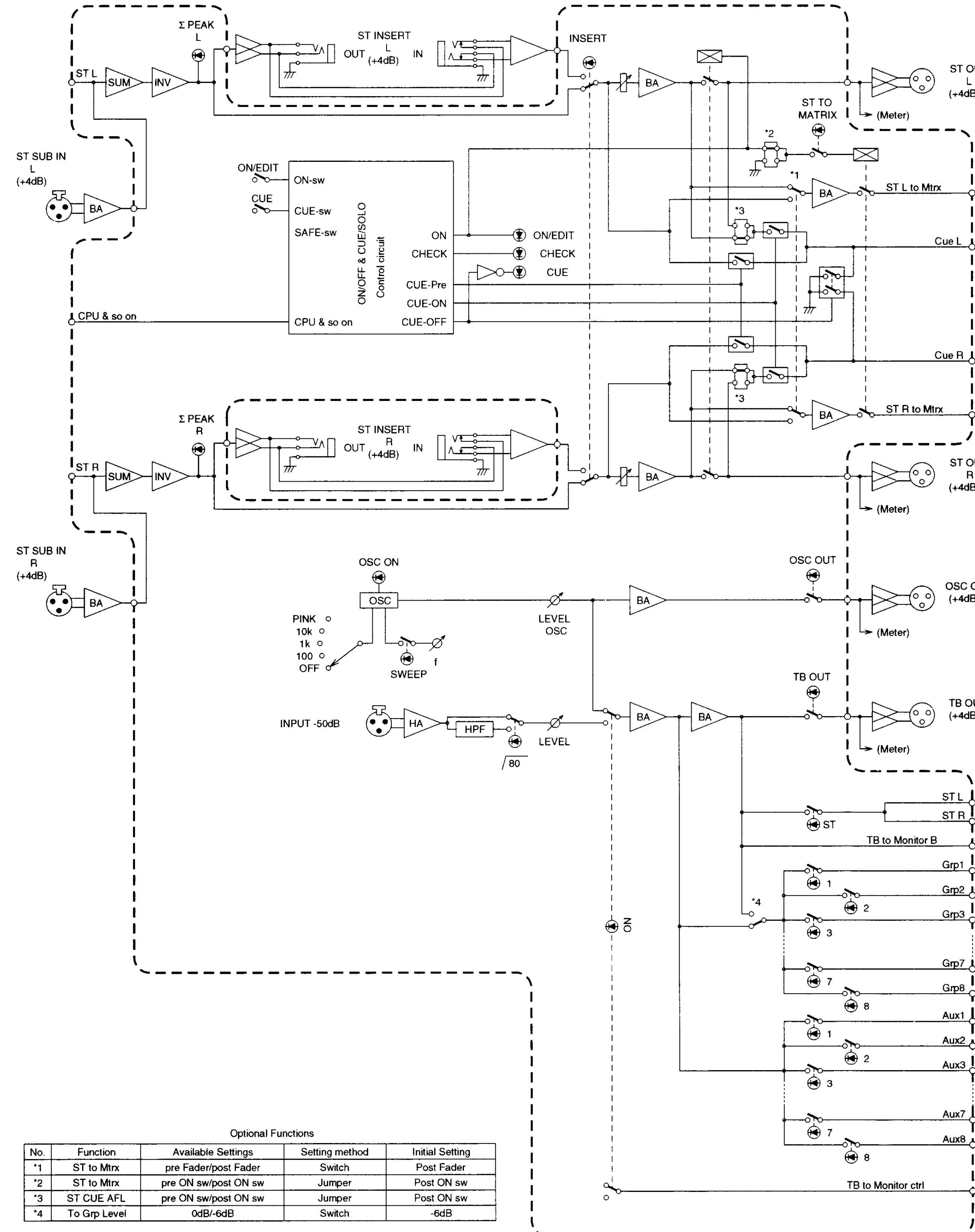
3.6.4 Group master module 2



Optional Functions

No.	Function	Available Settings	Setting method	Initial Setting
*1	Group Sum Gain	0dB/+6dB	Switch	+6dB
*2	Group to ST	pre ON sw/post ON sw	Jumper	Post ON sw
*3	Group to Matrix	pre Fader/post Fader	Switch	Post Fader
*4	Group to Matrix	pre ON sw/post ON sw	Jumper	Post ON sw
*5	Group CUE AFL	pre ON sw/post ON sw	Jumper	Post ON sw
*6	Matrix CUE AFL	pre ON sw/post ON sw	Jumper	Post ON sw

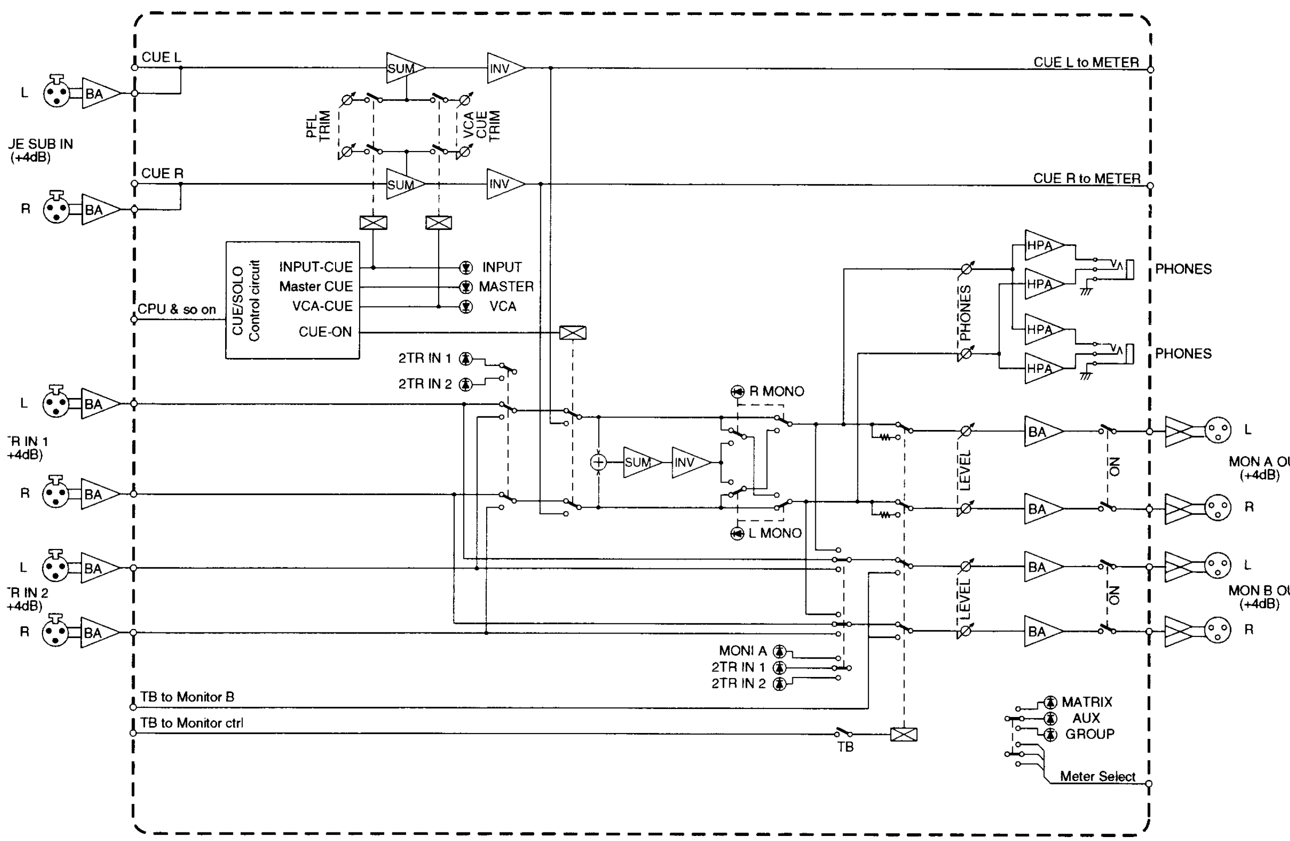
3.6.5 Stereo master module



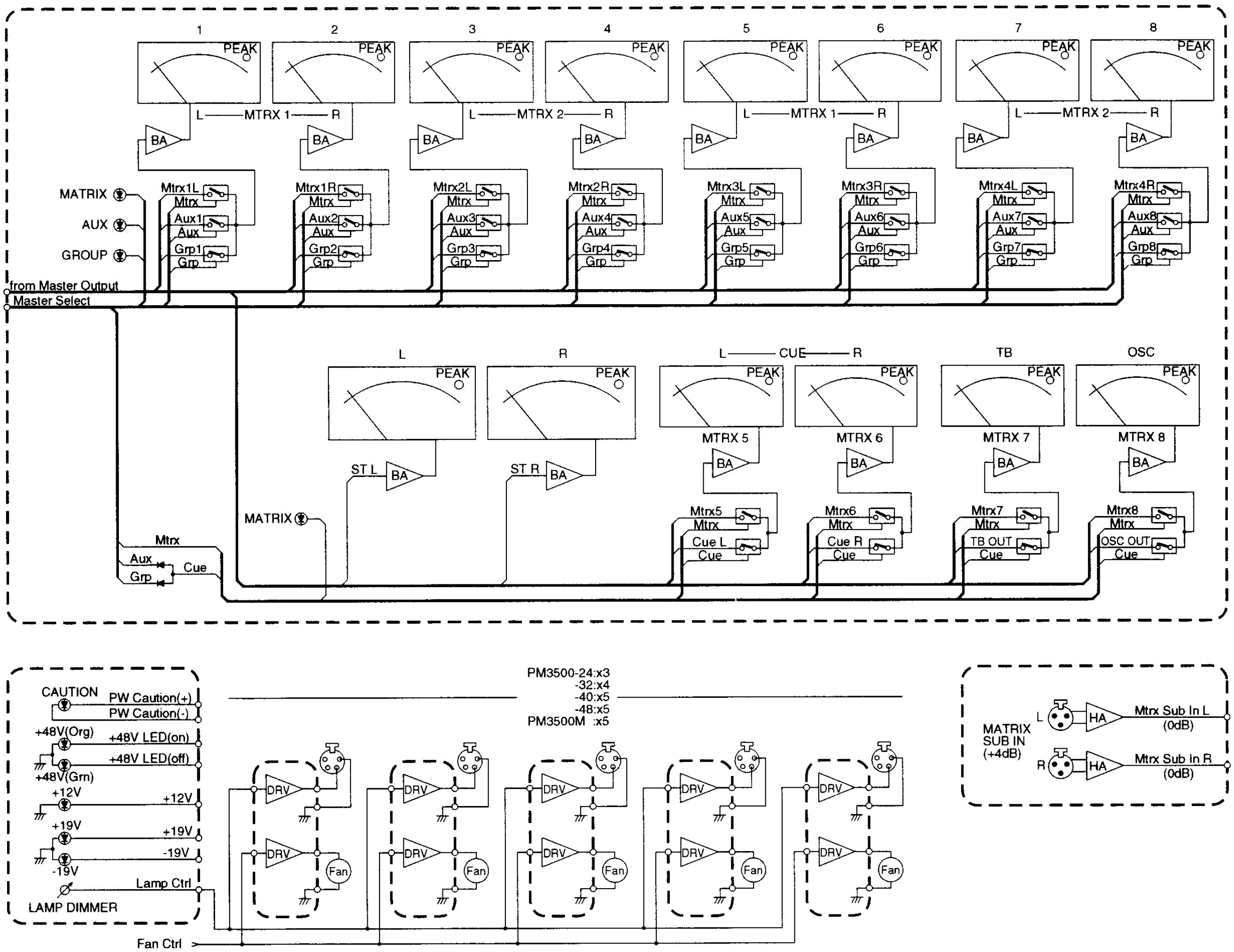
Optional Functions

No.	Function	Available Settings	Setting method	Initial Setting
*1	ST to Mtrx	pre Fader/post Fader	Switch	Post Fader
*2	ST to Mtrx	pre ON sw/post ON sw	Jumper	Post ON sw
*3	ST CUE AFL	pre ON sw/post ON sw	Jumper	Post ON sw
*4	To Grp Level	0dB/-6dB	Switch	-6dB

3.6.6 Monitor module

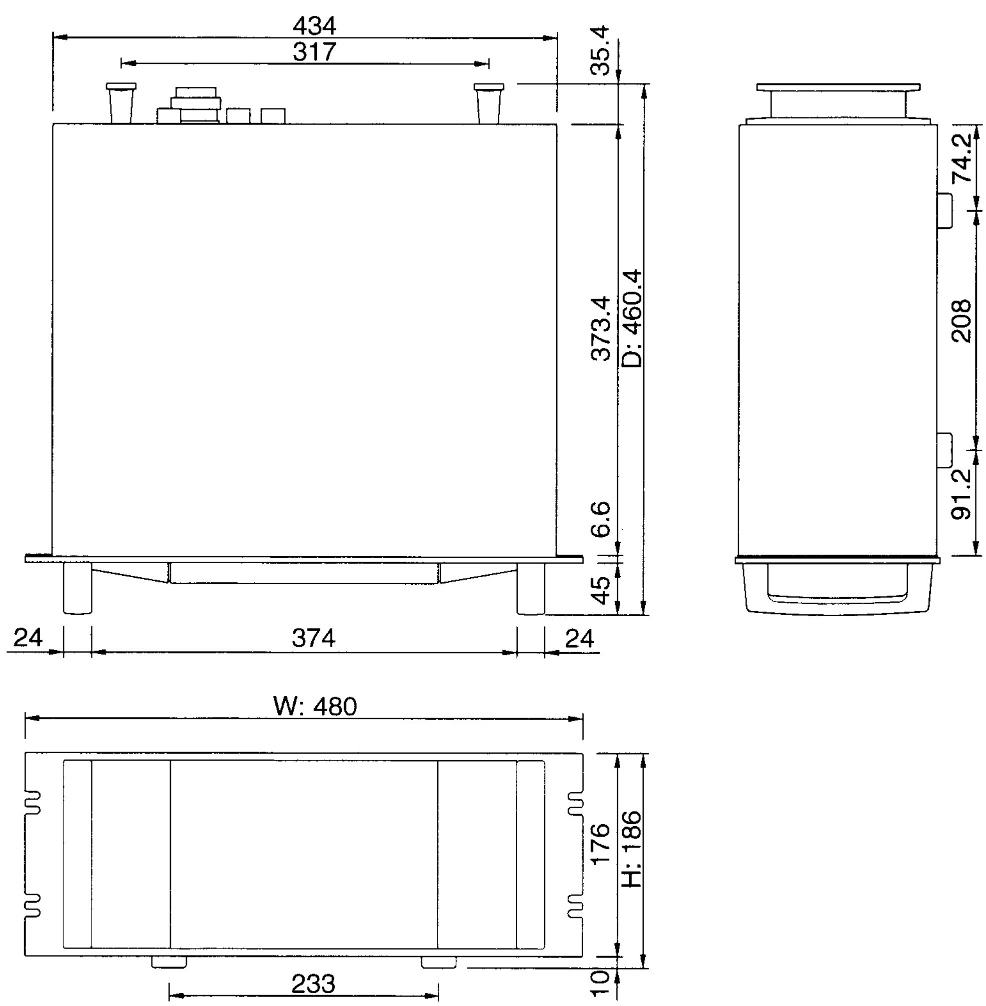


3.6.7 Meter bridge



3.7 PW4000 power supply specifications

Power requirements		120V 60Hz 1500VA 1250W	CSA/UL model
		240V 50Hz 1250W	BS model
		230V 50Hz 1250W	General model
Dimensions			
	Height	186mm (7.3 in)	
	Depth	461mm (18.1 in)	
	Width	480mm (18.9 in)	
	Weight	36kg (79.4lb)	
Secondary outputs		±19V	13A
		+12V	8A
		+48V	0.7A



All dimensions in millimeters (mm)

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 PM3500M, 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 [24] (page 19) through [31] (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 [24] (page 19) nor the CHECK key [28] (page 19) is lit. In this mode, the ON/EDIT switches [16] (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 [31] (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" (the memory number will be followed by a "/"). 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 54).

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 53).

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 56), 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 56).

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 57.

This operation is enabled or disabled with the UTILITY function (see "MIDI Control Change" on page 56).

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 53). 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 54), 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 56), 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 57) 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 58.

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 PM3500M, 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 [20] (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

4.6 MIDI implementation chart

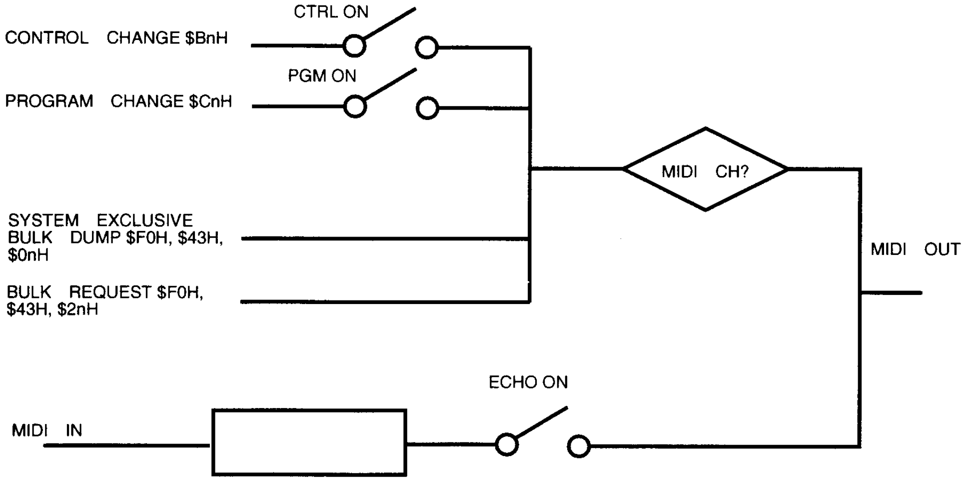
YAMAHA [Mixing Console] date:1994.11.24
Model PM3500M MIDI Implementation Chart Version : 1.0

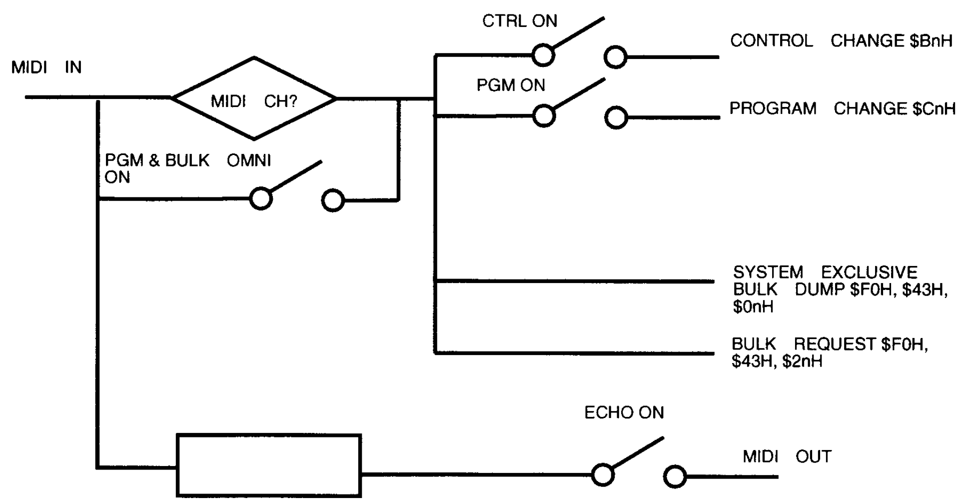
Function		Transmitted	Recognized	Remarks
Basic Channel	Default	1-16	1-16	memorized
	Changed	1-16	1-16	
Mode	Default	x	OMNI ON/OFF	memorized
	Messages	x	OMNI ON/OFF	
	Altered	**************	x	
Note Number		x	x	
	True voice	**************	x	
Velocity	Note ON	x	x	
	Note OFF	x	x	
After Touch	Key's	x	x	
	Ch's	x	x	
Pitch Bender		x	x	
Control Change	1-88	o	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 Messages	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 51) 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.

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.

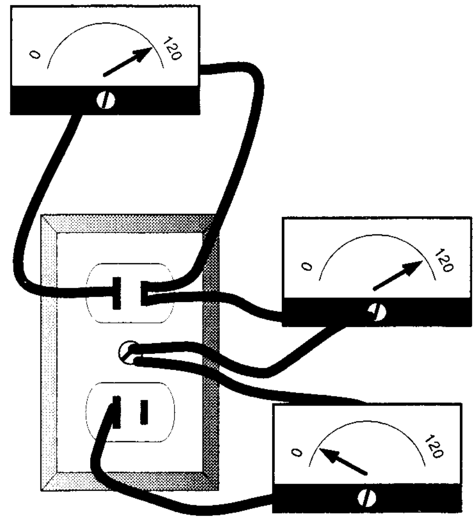
However, you must be sure that the screw is grounded.

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

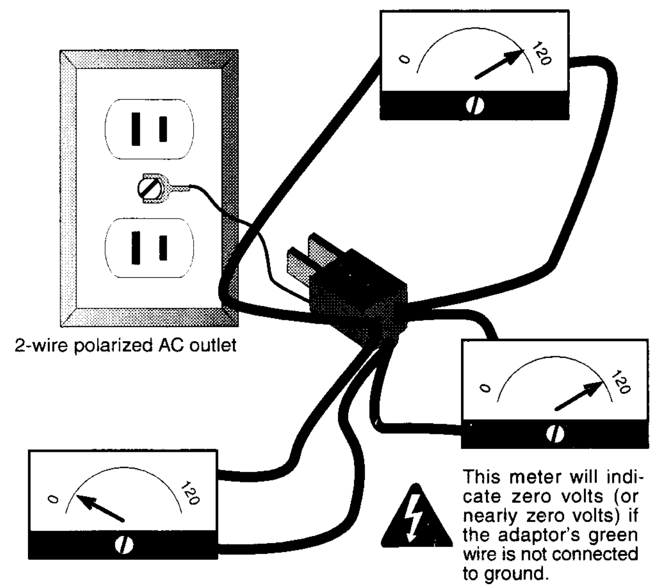
Plug the adaptor into the outlet.

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.

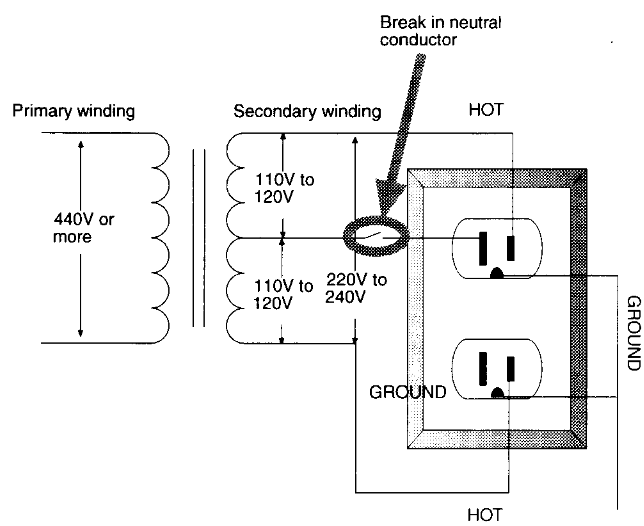
5.2.5 Improperly wired AC outlets: lifted neutral

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.

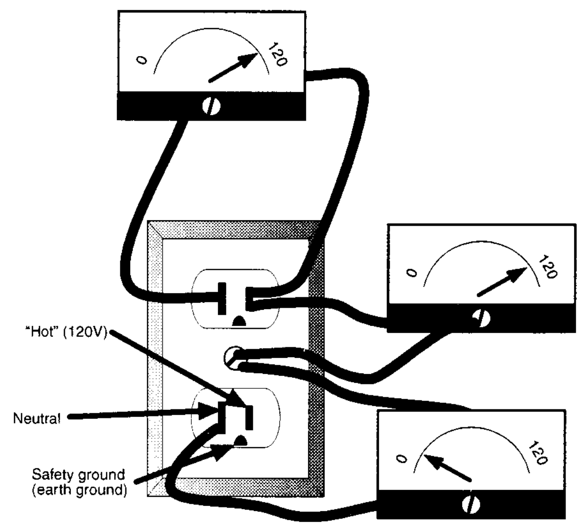
Such outlets may operate, but the voltage can swing from 0V to 220 or 240V AC (or the maximum voltage at the service entrance), creating a shock hazard and possibly damaging your equipment.

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.



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 connected and grounded.

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.

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.

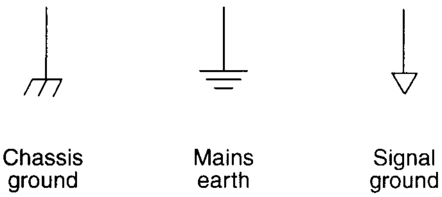
Ground is the electrical reference against which potentials (voltages) are expressed. In a practical audio system, a number of different independent references exist in various local subsystems. These may or may not be at the same electrical potential. If handled properly, they certainly need not be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

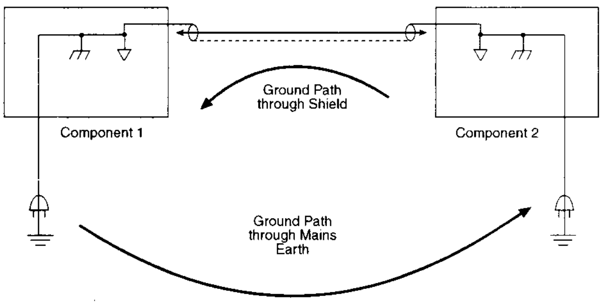
AC line-frequency hum is, without question, the single most common problem in sound systems, and the most common cause of hum is ground loops.

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.

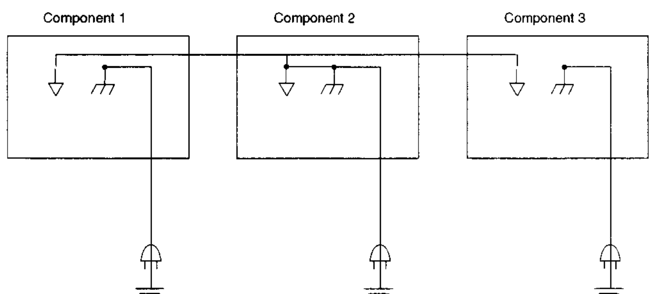
The figure below shows a typical ground loop situation. Two interconnected pieces of equipment are plugged into grounded AC outlets at separate locations, and signal ground is connected to earth in each of them. The earth ground path and duplicate signal ground path form a loop which can pick up interference. 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.



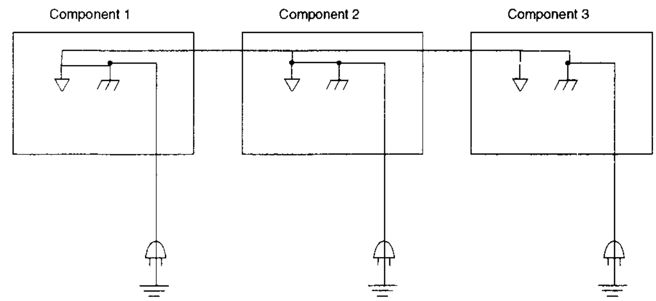
5.3.3 Basic grounding techniques

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.

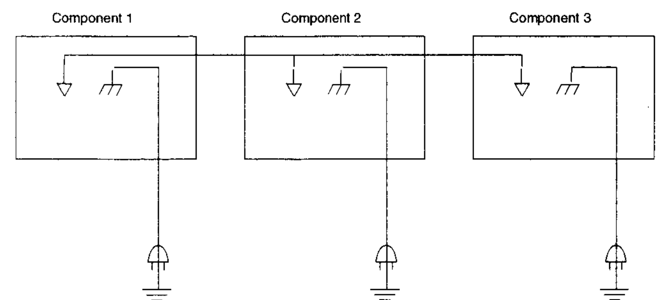
The figure below illustrates the single-point grounding principle. Chassis ground in each individual component is connected to earth; signal ground is carried between components and connected to earth at one central point. This configuration is very effective in eliminating line frequency hum and switching 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.



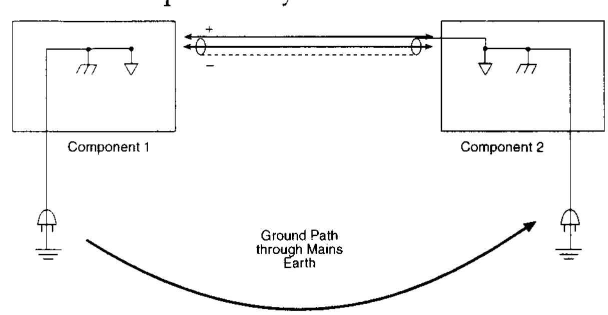
Multiple point grounding is shown in the next illustration. This situation is common in systems that use unbalanced equipment having the chassis connected to signal ground. It has the advantage of being very simple in practice, but it is not very reliable †± particularly if the connection configuration 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.

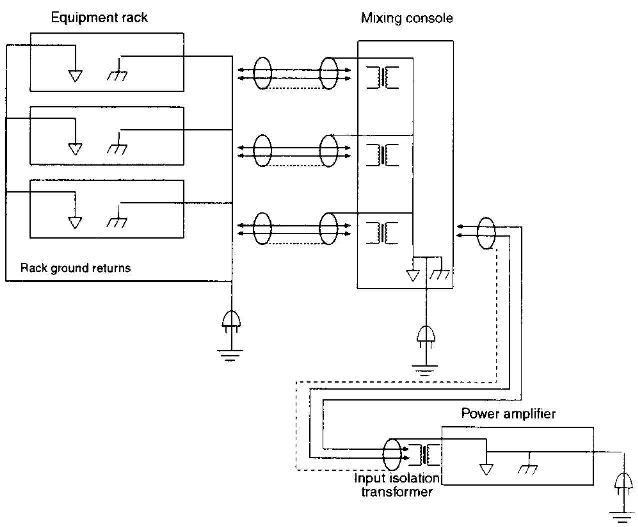


The principle of telescoping shields is shown below. This scheme is very effective in eliminating ground loops. If shields are connected only to earth, unwanted signals that are induced in them can never enter the signal path. Balanced lines and transformers are required to implement this approach, since 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.

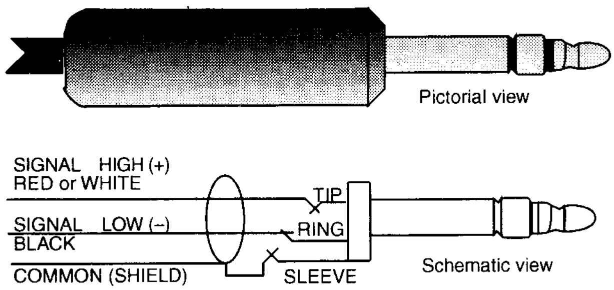
In those audio devices which anticipate ground loops by providing "ground lift" switches next to XLRs or three-wire phone jacks, the ground lift switch makes and breaks the connection between the connector's shield and the chassis of the particular device. Ground lift switches are usually found on "direct 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). In a monitoring console such as the PM3500M, direct boxes are not usually used, and their use is not discussed in this manual.

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.

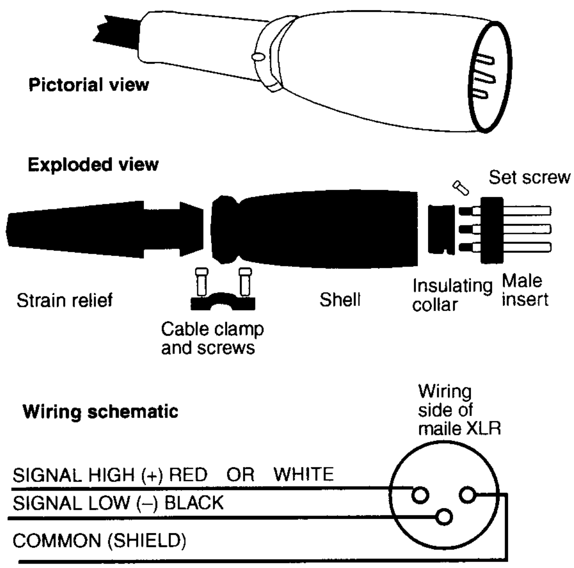
5.4 Audio connectors and cables

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.

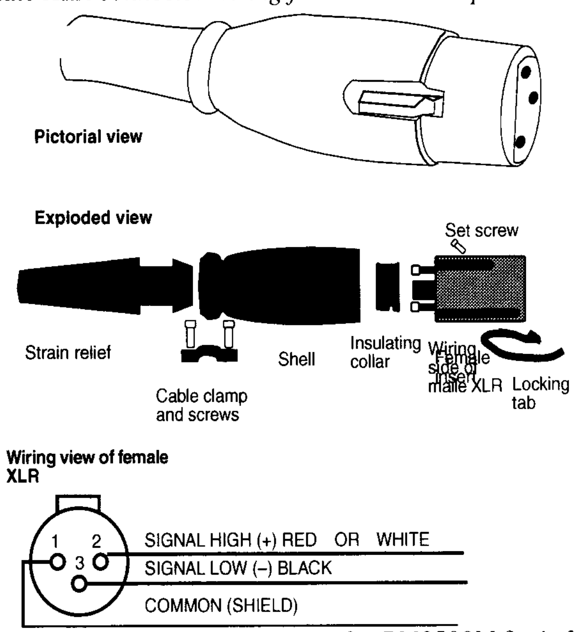
The PM3500M is fitted with only two types of audio connectors: 3-pin XLRs, both male and female, and 3-circuit (tip/ring/sleeve) 1/4" phone jacks, also known as stereo phone jacks, although their function is always (except in the case of the headphone jacks) to carry a balanced mono signal rather than a stereo signal.



TRS phone plug wiring for PM3500M insert IN/OUT jacks



Male XLR connector wiring for PM3500M 3-pin XLR inputs



Female XLR connector wiring for PM3500M 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.

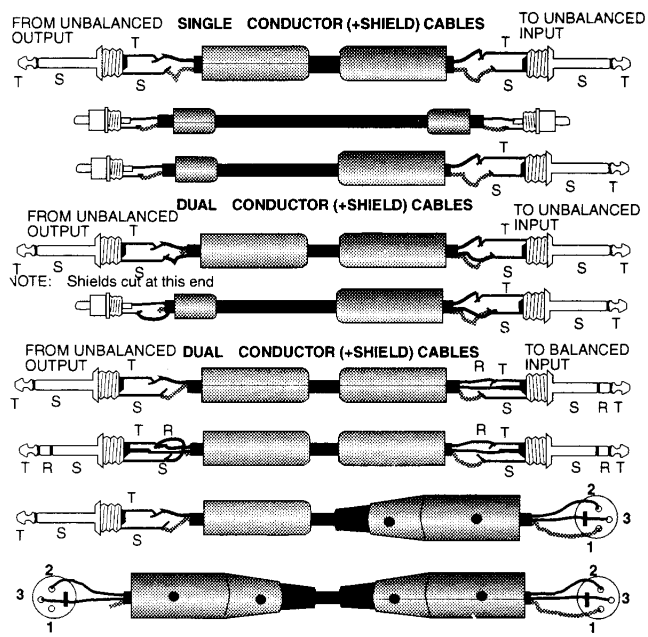
Balanced wiring is more expensive to implement than unbalanced wiring. It is often used, however, because it offers useful advantages, especially in portable sound systems. There is nothing inherently "better" or more "professional" about balanced wiring; the application dictates whether one system or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

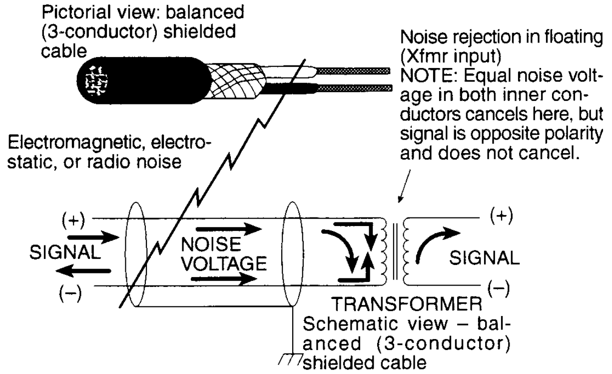
The RFI (radio frequency interference) cuts across both conductors, inducing equal voltages in the same direction. These voltages "meet" in the differential amplifier (or transformer), and cancel out, while the signals generated by the microphone flow in opposite directions in each conductor, and hence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



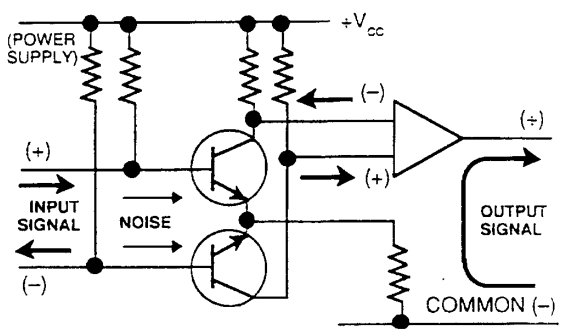
Connect pin 1 at both ends of a mic cable.
For a line cable, cut the shield here, as shown.

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 PM3500M, 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.

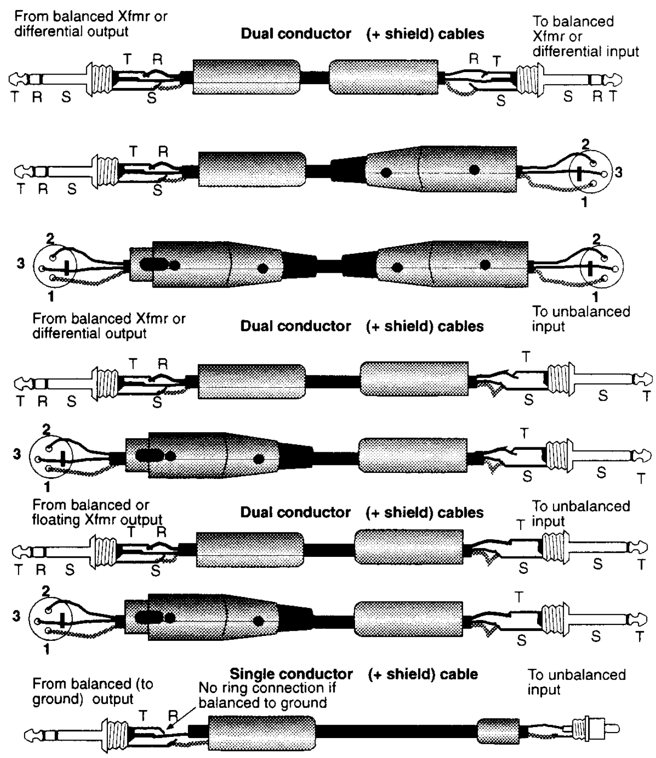
If the transformer is also used to prevent a low impedance input from overloading a high impedance output, it is known as a "bridging" transformer (not to be confused with the "bridged" connections of a stereo power amp output in mono mode).

In general, the PM3500M 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, PM3500M 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 PM3500M. 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 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 before-hand 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.

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

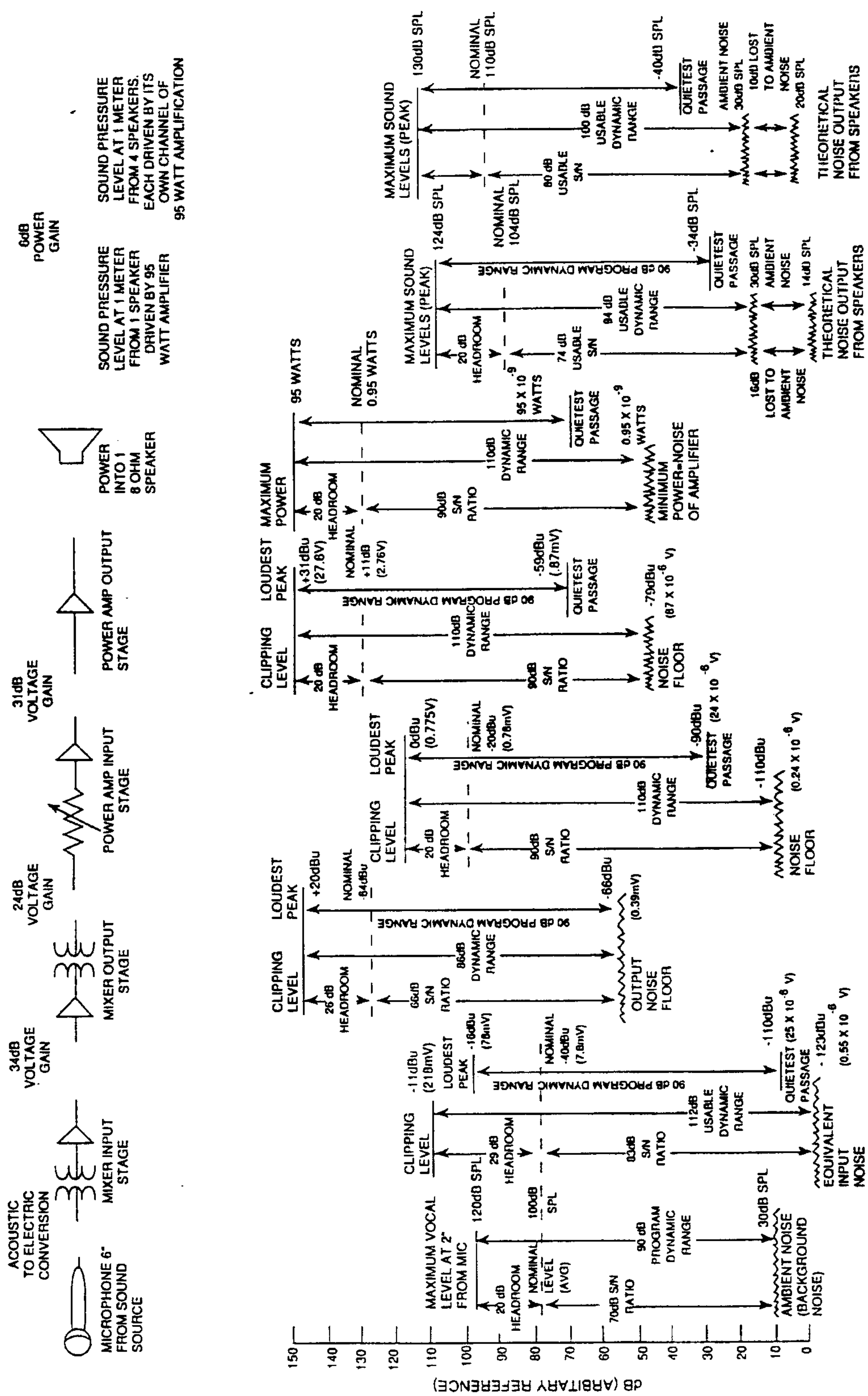
A piece of consumer sound equipment ("hi-fi") may operate at considerably lower nominal (average) line levels than +4dBu. This is typically around –16dBu (123 mV) to –10dBu (245mV) into 10,000Ω or higher loads. Peak output levels in such equipment may not go above +4dBu (1.23V). The output current available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

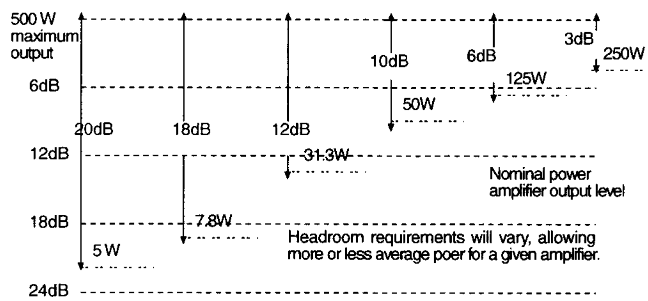
If line amplifiers, electronic crossovers, equalizers or other signal processing devices are inserted in the signal chain, signal levels at the input of these units should be set so the dynamic range of each unit is optimized. In other words, set the input level at each device as high as possible without 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 PM3500M 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 PM3500M 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 PM3500M 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 PM3500M 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.