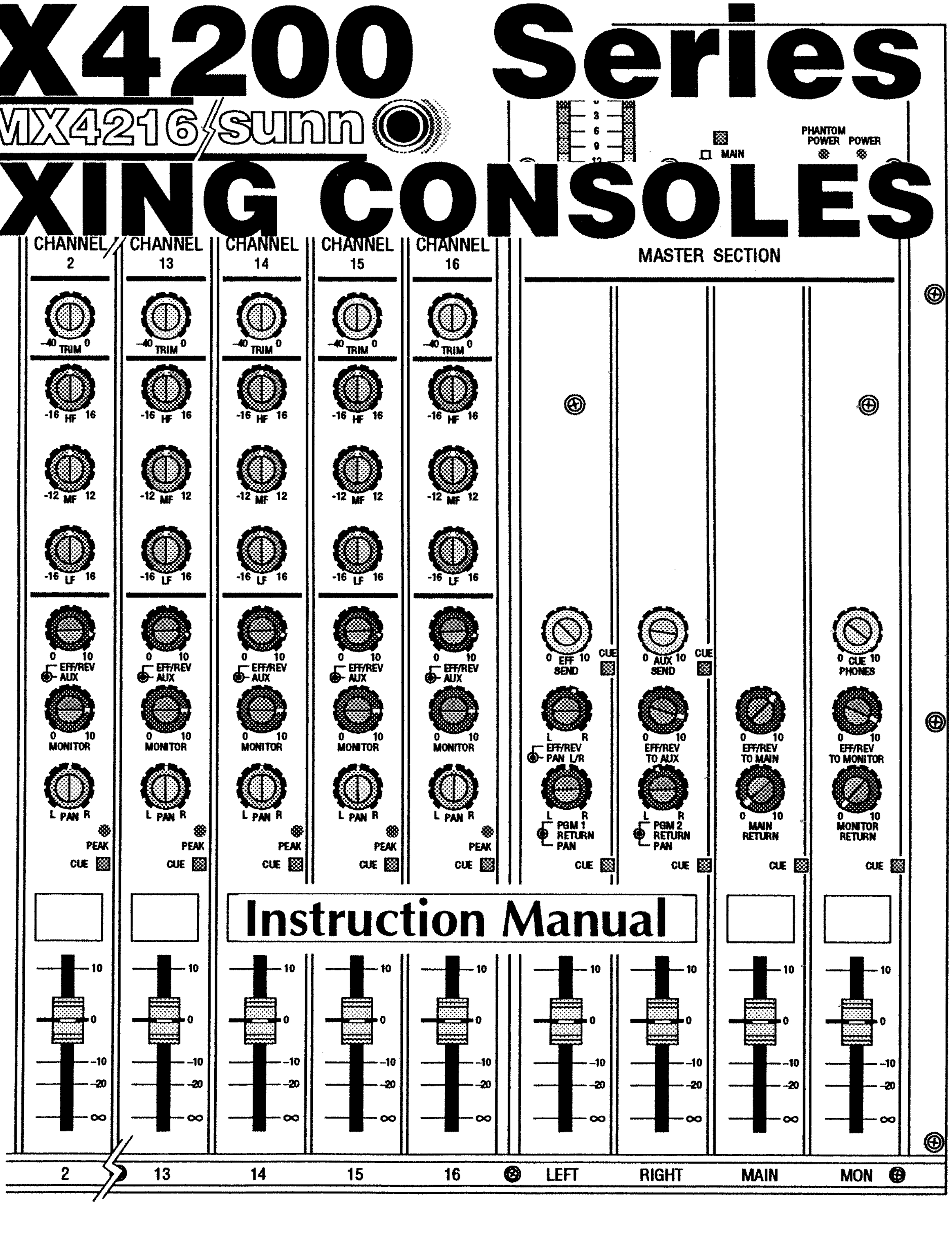
sunn

MX4208
MX4212
MX4216

MX4200 Series MIXING CONSOLES



The MX4208, MX4212 and MX4216 are virtually identical mixers, differing only in that they have 8, 12 or 16 input channels respectively. We refer to them herein as the MX4200 Series, or the MX4200 mixer to be brief. Their features are discussed in the BRIEF OPERATING INSTRUCTIONS section of this manual. Check this and the SPECIFICATIONS section, and you will see most of what you need to know to operate the console. The balance of this manual provides background information for optimum connection and better utilization of the console and auxiliary equipment.

There are internal jumpers within the console which can be configured to alter the signal paths in certain circuits. Refer to the OPTIONAL PRE-POST FUNCTIONS section for details.

TERMINOLOGY AND TYPOGRAPHIC CONVENTIONS

Generally, where we refer to a particular control or function as it is actually labeled on the console, we will use all upper case type. That is, if we refer to an input channel's gain trim control, we may print "the input TRIM control." On the other hand, if the feature is not labeled, we will use upper case type only on the first letter; for example, "observe there is no identification of the input Fader." Where the labels are not explicit, as for example the input channel equalizer's mid-frequency control labeled "MF", we may add a parenthetic reference "(EQ)". We use the terms "mixer," "console" and "mixing console" interchangeably.

Particularly important information is distinguished in this manual by the following notations:

NOTE: A NOTE provides key information to make procedures or functions clearer or easier.

CAUTION: A CAUTION indicates special procedures or guidelines that must be observed to avoid damage to the console or related equipment, or to avoid an undesirable result while using the console.

WARNING: A WARNING indicates special procedures or guidelines that must be observed to avoid injury to the operator or others using or exposed to the console or related equipment.

In the BRIEF OPERATING INSTRUCTIONS section of this manual, each feature is provided with a numerical reference. Elsewhere, if we are referring to that feature, we may cite the reference number in square brackets for clarity. For example, on the input section, the fourth item to be described is a pair of controls labeled EFF and AUX. In other places on the console there are other EFF controls. For clarity, then, if we are discussing this particular input control, we will describe it like this: "the EFF send control [4]."

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WARNING: to prevent fire or shock hazard, do not expose this equipment to rain or moisture.

SECTION 1 INTRODUCTION

The MX4200 Series are audio mixing consoles that offer solid performance in a very cost effective, no-frills package.

The same basic console is available with 8, 12 or 16 input channels. There is a stereo mixing bus, a monitor bus, an effects bus, and an auxiliary bus to which any of the input channels can be assigned. There is also a main mixing bus to which an external signal source and an effects return may be applied, providing in-console control and enhancement of a submixed source from another console or remote signal source.

The MX4200 mic inputs are differentially balanced XLRs, and are equipped with a continuously variable gain TRIM control so that literally any mic and many line level signals can be accommodated with channel faders set at nominal level. Separate 1/4" tip/ring/ sleeve phone jacks on each channel accommodate line level input signals. While the console has ample headroom throughout, it is always possible to incorrectly set controls. For this reason, the MX4200 is equipped with "PEAK" LEDs on each input channel, after the EQ section. A pair of 10-segment LED output VU meters can be switched to provide output stage indication for the left and right program buses, the main output or the monitor output.

The MX4200 is equipped with both auxiliary and effects buses, as well as a separate monitor bus. The effects mix is automatically routed to a built-in reverb unit, and back to the various effects return controls. It is also routed to an EFFECTS OUT jack so it can drive an external signal processor. If the signal from an external processor is plugged into the EFFECTS RETURN jack, it automatically replaces the signal from the built-in reverb. However, the built in reverb can be allowed to function in parallel with the external device driven by the same mix if the return is brought to one of the two program return inputs, the monitor return or the main return (which apply signals to the respective buses). The auxiliary mix can be used to drive a separate signal processor, or as an auxiliary program or monitor mix.

All input channels have internal jumpers which enable you to customize the console's functions to suit your specific requirements. For example, the aux mix can be derived post-fader and post-EQ (factory setting) or pre-fader post-EQ. The monitor mix, which is discrete from the aux and effects mixes, can be jumpered to be post-EQ and pre-fader (the factory setting) or pre-EQ and pre-fader.

A high degree of flexibility is afforded by the multiple return paths, each with separate level controls. The EFFECTS DIRECT INPUT, in conjunction with the PROGRAM, MAIN and MONITOR RETURN INPUTS, may be used to link another mixer's output to the MX4200 for an expanded system.

An important feature of the MX4200 is its extensive cue capability. There is a CUE switch on every input channel and master output. Cue applies the corresponding signal to the stereo headphones and the mono cue output jack. Cue may be used for troubleshooting, previewing a channel before applying it to the mix, or "touching up" the EQ on a channel during a performance. It is also used by the console operator for listening to the mix during an event.

When operated in accordance with these instructions, your MX4200 will yield very low-distortion, wide bandwidth audio. Noise is very low, thanks to well-engineered circuitry and a sensible internal grounding scheme that minimizes hum due to *ground loops*.

Take your time studying the panel, read the descriptions in this manual, and you'll find operating this console to be very straightforward.

SECTION 2
BRIEF OPERATING INSTRUCTIONS

2.1 MX4200 Front Panel Features:

2.1.1 The Input Section

The input section consists of either 8, 12 or 16 input strips — identical groupings of controls which each process a discrete pair of rear-panel mic and line inputs. The following descriptions of one input strip are applicable to all of them.

1. **TRIM**
   The knob provides 40 dB of continuously variable adjustment for the input preamplifier gain. A setting of "–40" is least sensitive, corresponding to 40 dB of attenuation, whereas a setting of "0" is most sensitive, corresponding to 0 dB of attenuation. It is always a good idea to begin with the TRIM set to –40 position, and to turn it up from there to avoid input overdrive.

2. **Equalizer**
   Each input channel equalizer is divided into three bands. The level (gain) is adjustable over a wide range of boost and cut.

   HF (High Frequency EQ)
   Provides ±15 dB EQ at 10,000 Hz.

   MF (Mid Frequency EQ)
   Provides ±12 dB EQ with the band center at 3,000 Hz.

   LF (Low Frequency EQ)
   Provides ±15 dB EQ at 50 Hz.

   The high and low bands have shelving EQ curves, which means that the applied boost or cut does not change appreciably with frequency above the HF knee (or hinge point) or below the LF knee. The mid band has a peaking curve, which means that the amount of boost or cut is maximum (to the set value) only at the center frequency of that band; the effect of the EQ decreases as the frequency is further from the MF band center.

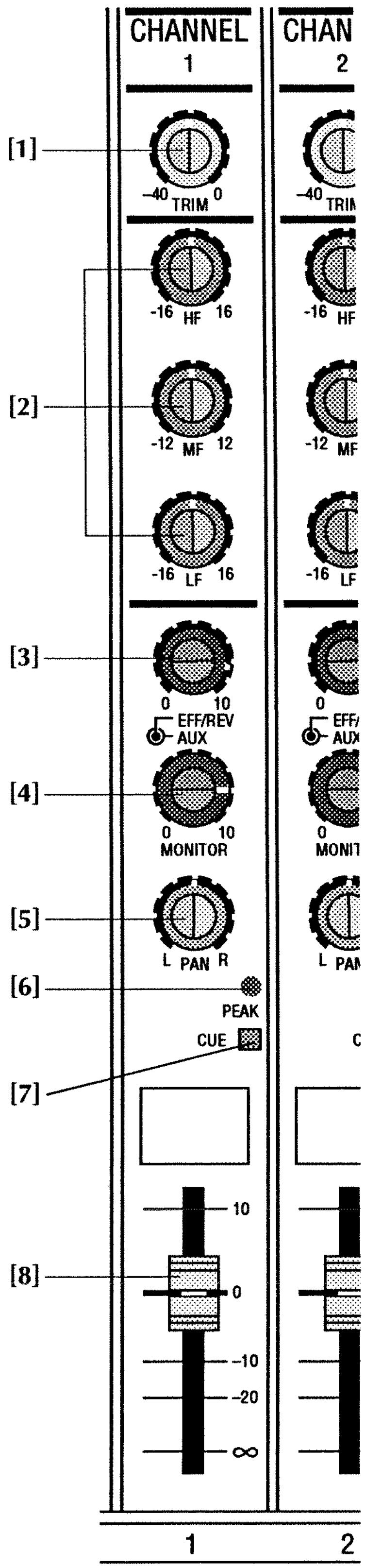


Figure 2-1. MX4200 Input Channel (Typical of all 8, 12 or 16 channels)

3. **EFF/AUX**
   These 2 rotary send level controls are concentric, the larger knob controlling the AUX bus send, and the smaller one the EFFects bus send. Each control determines how much of the channel's signal is applied to the corresponding mixing bus. Refer to items [9] and [10] for a description of the effects and aux output mixes.
   The EFFects send is always derived post-fader and EQ (i.e., its level also is affected by the channel's fader setting and its sound quality is altered by the channel EQ controls). The AUX send is factory-wired to be derived from the same point as EFF, so these two sends are identical, as delivered, and may be used for a pair of EFFects sends (for instance, EFF for the built-in reverb unit, and AUX for an external delay line — or as left and right sends to a full stereo effects unit). However, AUX may be internally rewired on a channel-by-channel basis, via a plug-in jumper change, to be derived pre-fader (i.e., ahead of the channel fader, but still affected by the EQ). If so rewired, it then functions very much like a second MONITOR level control [4]. Refer to page 15 for details of the jumper alteration, and the reasons why this may be desirable.

4. **MONITOR**
   The MONITOR control determines how much of the channel's signal is applied to the monitor mixing bus, typically used to drive stage monitor speakers. The monitor send is factory wired to be derived pre-fader and post-EQ. However, MONITOR may be internally rewired on a channel-by-channel basis, via a plug-in jumper change, to be derived pre-fader and pre-EQ. Refer to page 17 for details of the jumper alteration, and the reasons why this may be desirable.

5. PAN

This rotary control is known as a pan pot because it is a potentiometer (pot) which controls the stereo panorama (pan). It enables a mono input signal to be applied to the left and right program mixing buses in a continuously variable proportion from all left (full counterclockwise rotation) to all right (full clockwise rotation).

A center detent is provided for equal PAN signal assignment to both buses, which applies 3 dB less signal to each bus than the level obtained with full left or right assignment so that the combined stereo signal adds up to constant power at all PAN pot positions.

6. PEAK

This red light emitting diode (LED) turns on when the post-EQ, pre-fader signal is 3 dB below clipping, a warning to decrease the EQ boost and/or to turn down the signal level with the channel's TRIM control.

Clipping the equalizer section can occur even though the input signal is not clipping, due to boost (added gain) applied. Therefore, pay close attention to this LED, and if it flashes more than occasionally, either turn down the TRIM level, use an external attenuation pad, turn down the output level of the device connected to the input channel, or use some combination of these three techniques.

7. CUE

Engaging this input channel CUE switch causes the channel signal to be applied to the Cue output and the PHONES output [52]. Cue is derived after the EQ *and* channel patch point, but ahead of the channel fader. It may be used to preview a channel's incoming signal, pre-adjust the EQ and TRIM prior to raising the fader to apply signal to the program buses.

*NOTE: The console operator will generally be listening to one or more of the master cue mixes — the stereo buses, or possibly the main or monitor buses — by means of engaging their CUE switches. The MX4200 is set up to simply add any input cue signals to this output cue mix. (While the output program cue mix appears in the headphones as a stereo mix, the input cue signals are applied equally to left and right and appear as a center mono signal.) To **solo** one or more inputs, simply disengage all master CUE switches.*

8. FADER

This linear fader sets the level applied to the left and right program mixing buses, the effects bus, and the AUX bus (assuming the internal pre–post jumper is set as factory shipped). It only affects the channel's monitor send if the internal pre–post jumper has been reset to post–fader position.

2.1.2 Master Section

9. EFF SEND

This rotary control adjust the overall signal level from the effects mixing bus applied to the EFFECTS OUTPUT jack [43]. It does not affect the level applied to the build-in reverb unit.

10. AUX SEND

This rotary control adjust the overall signal level from the aux mixing bus applied to the AUX OUTPUT jack [42].

11. CUE/PHONES

This 2-gang rotary control adjust the output level to both sides of the stereo PHONES output jack [52]. It also simultaneously adjusts the level at the mono CUE OUTPUT jack [46].

12. EFF/REV & PAN L/R (Concentric Controls)

These 2 rotary controls are concentric. The smaller knob controls the level of the signal from the built-in reverb or the EFFECTS RETURN input jack [35] applied to the PAN L/R pot, which is the larger knob. PAN L/R applies that signal to the left and right program mixing buses in a continuously variable proportion from all left (full counterclockwise rotation) to all right (full clockwise rotation).

These controls normally process the signal from an effects unit, such as a delay line, reverb, or other specialized signal processor. The effects return input may be used as an auxiliary line input, in which case these controls act much like the fader [8] and pan pot [5] on a standard input channel.

13. EFF/REV TO AUX

This control sets the level of the signal from the built-in reverb or the EFFECTS RETURN input jack [35] applied to the aux mixing bus. If aux is used for a stage monitor (foldback) mix, then this control may be used to adjust the reverb (or other effect) heard in that mix.

CAUTION: If the aux mix is used as an effects mix, with the AUX output [42] driving an external effects unit and brought back to the EFFECTS RETURN input [35], then make sure this control is turned all the way down to 0 (full counterclockwise). Otherwise, a feedback loop will be established. The consequences will be a powerful howl or squeal, and possibly damage to human hearing, electronic circuits and/or loudspeakers.

14. EFF/REV TO MAIN

This control sets the level of the signal from the built-in reverb or the EFFECTS RETURN input jack [35] applied to the main mixing bus without having to put the reverb into the stereo L/R buses.

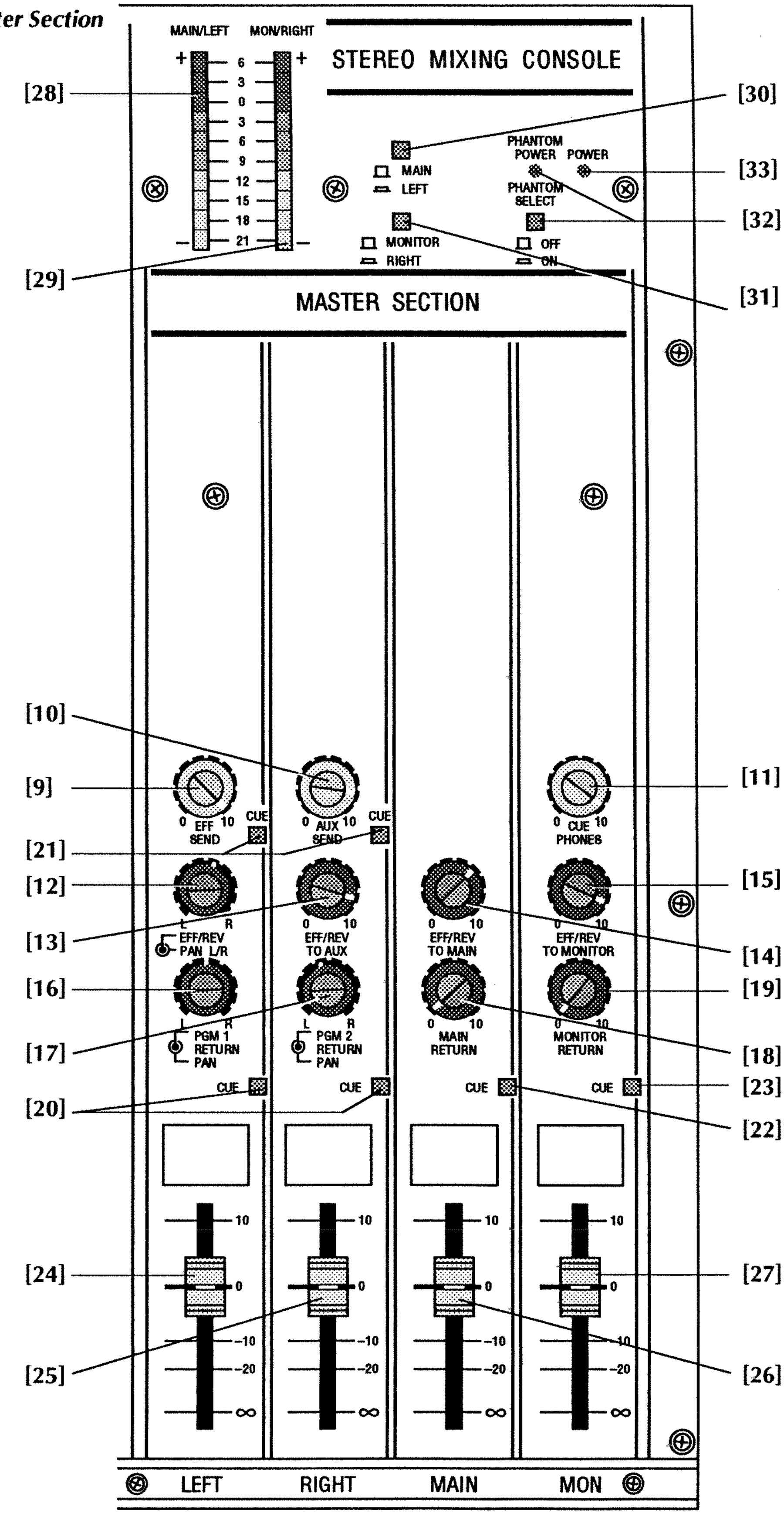
15. EFF/REV TO MONITOR

This control sets the level of the signal from the built-in reverb or the EFFECTS RETURN input jack [35] applied to the monitor mixing bus.

16. PGM 1 RETURN & PAN (Concentric Controls)

These 2 rotary controls are concentric. The smaller knob controls the level of the signal from the PROGRAM RETURN 1 input jack [41] applied to the PAN pot,

Figure 2-2. MX4200 Master Section



which is the larger knob. The PAN control applies that signal to the left and right program mixing buses in a continuously variable proportion from all left (full counterclockwise rotation) to all right (full clockwise rotation).

17. PGM 2 RETURN & PAN (Concentric Controls)

These 2 rotary controls are similar to the adjacent PGM 1 controls [16], except they control the signal from the PROGRAM RETURN 2 input jack [40].

18. MAIN RETURN

This control sets the level of the signal from the MAIN RETURN input jack [37] applied directly to the main mixing bus.

19. MONITOR RETURN

This control sets the level of the signal from the MONITOR RETURN input jack [36] applied directly to the monitor mixing bus.

20. Left and Right Program CUE

Pressing either of these switches applies Cue signal from the corresponding left or right program output (pre-master fader [24, 25]) to the respective left and right sides of the PHONES output jack [52], and to the monaural CUE OUTPUT jack [46].

The two CUE switches [20 and 21] are generally used by the operator to monitor the stereo program output of the console. Since they are pre-fader, they may be used to check the output program before bringing up the faders to feed the house. To solo an input in the cue mix, disengage these switches and press the input channel(s) CUE switch.

21. EFF or AUX SEND CUE

These switches are virtually identical to one another, and apply signal from the EFFects send output or AUX send output to the cue bus. The signal is thus applied equally to both sides of the PHONES output jack [52] (it is center mono) and to the CUE OUTPUT jack [46].

22. Main Bus CUE
Pressing this switch applies Cue signal from the main program output (pre-master fader [26]) to the cue bus. The signal is thus applied equally to both sides of the PHONES output jack [52] (it is center mono) and to the CUE OUTPUT jack [46].

23. Monitor Bus CUE
This switch is nearly identical to the adjacent main bus CUE switch [22], except it applies signal from the monitor bus to the cue bus and subsequent outputs.

24. Left Program Master Fader
This linear fader adjusts the level applied from the left program mixing bus to the LEFT OUT jack [45].

25. Right Program Master Fader
This linear fader adjusts the level applied from the right program mixing bus to the RIGHT OUT jack [44].

26. Main Master Fader
This linear fader adjusts the level applied from the main mixing bus to the MAIN OUT jack [39].

27. Monitor Master Fader
This linear fader adjusts the level applied from the monitor mixing bus to the MONITOR OUT jack [38].

28. MAIN/LEFT (Meter)
This array of 10 LEDs constitutes a bar-graph type VU (volume unit) display of the average output level at the MAIN or the LEFT PROGRAM output, depending on the setting of the adjacent MAIN/LEFT meter switch [30]. Ideally, the display should operate in the range from –6 to 3 during typical program passages, and should only occasionally reach a level of 6. If it is regularly pegged at 6, then the bus level is too high, and either the corresponding master fader should be pulled down, or the various channel controls throughout the console which feed the corresponding bus should all be turned down. The meter is calibrated so that a zero indication represents an open circuit output level of 0 dBu.

29. MON/RIGHT (Meter)
This meter is similar to the adjacent MAIN/LEFT meter, except it displays the output level at the MONITOR or LEFT PROGRAM output, depending on the setting of the adjacent MONITOR/RIGHT meter switch [31].

30. MAIN/LEFT (Switch)
When this switch is engaged (down), the MAIN/LEFT meter [28] displays the level at the LEFT OUTPUT jack [45]. When the switch is disengaged (up), the meter displays the level at the MAIN OUTPUT jack [39].

31. MON/RIGHT (Switch)
When this switch is engaged (down), the MON/RIGHT meter [29] displays the level at the RIGHT OUTPUT jack [44]. When the switch is disengaged (up), the meter displays the level at the MONITOR OUTPUT output jack [38].

32. PHANTOM POWER SELECT (Switch) and PHANTOM POWER (Indicator)
Engaging the switch applies 15 volts to the phantom power bus, and thereby to all XLR microphone input connectors [50] for powering electret condenser microphones. The LED is illuminated to indicate the phantom power is on. This voltage should not damage 12 volt mics (though you should be sure they are not A–B type mics, which have different pin configurations), and should be adequate for microphones rated at 48 volts (the primary drawback to using a lower voltage is slightly less maximum level capability in such mics; the advantage is that there is no conflict with UL listing requirements).

33. POWER indicator
This LED is illuminated when the console is turned on with the rear-panel POWER switch [51].

2.1.3 Rear Panel

34. EFFECTS DIRECT IN
This unbalanced phone jack applies nominal –6 dBu (388 mV rms) signal directly to the effects (EFX) mixing bus. It may be used as a sub input to link the effects bus from a submixer (such as another MX4200 series console's EFFECTS OUT jack) to create a larger mixing system. Generally, the output of an effects device would not be connected to this input because the output of this bus is typically fed to the input of the effects device, and such a loop would create feedback.
If, however, the MX4200 AUX OUT [42] or MONITOR OUT [38] is used to drive the effects unit (signal processor), and the effects unit has an output level control, then that signal may be applied to this input. This makes sense in cases where you wish to loop the output of one signal processor into the input of another, using the console as a control center. For example, the aux mix could be fed to a delay line via the AUX OUT jack [42]. The output of the delay line could then be brought into the EFFECTS DIRECT IN jack [34]. The signal is then automatically applied to the built-in reverb unit (or to any external reverb connected to the EFFECTS OUT jack [43]), creating a slap-echo effect which is subsequently controlled by the various effects return controls [12], [13], [14] and/or [15].

35. EFFECTS RETURN
This unbalanced phone jack applies nominal -6 dBu signal directly to four different EFF/REV controls, replacing any signal which may have been present from the built-in reverb unit. It is generally used for bringing the

output of a signal processor back into the console. The signal appears at the EFF/REV–PAN L/R controls [12], EFF/REV TO AUX control [13], EFF REV TO MAIN control [14], and EFF/REV TO MONITOR control [15], where it can be independently assigned to the respective buses.

36. MONITOR RETURN

This unbalanced phone jack applies nominal –6 dBu signal directly to the MONITOR RETURN control [19], and then to the monitor bus. It may be used to apply an effects signal to the monitor mix, effects which are independent of the built-in reverb or of any external signal processing patched between the EFFECTS OUT [43] and EFFECTS RETURN [35] jacks.

This jack may be used as a sub input to link the monitor bus from a submixer (such as another MX4200 series console's MONITOR OUT jack) to create a larger mixing system.

37. MAIN RETURN

This unbalanced phone jack applies nominal –6 dBu signal directly to the MAIN RETURN control [18], and then to the main bus. It may be used to apply an effects signal to the main mix, effects which are independent of the built-in reverb or of any external signal processing patched between the EFFECTS OUT [43] and EFFECTS RETURN [35] jacks.

This jack may be used as a sub input to link a primary mixing bus from a submixer (such as another MX4200 series console's MAIN OUT jack) to create a larger mixing system.

38. MONITOR OUTPUT

This unbalanced phone jack delivers the post master fader [27] line-level output of the monitor mixing bus to an external power amplifier, mixer or signal processor. The nominal output level is –6 dBu.

39. MAIN OUTPUT

This unbalanced phone jack delivers the post master fader [26] line-level output of the main mixing bus to an external power amplifier, mixer or signal processor. (The Main bus is a mono combination of the Left and Right busses.) Nominal output level is +4 dBu (1.23V rms).

40. PROGRAM 2 RETURN

This unbalanced phone jack applies nominal –6 dBu signal directly to the PGM 2 RETURN level control and PAN pot [17], and then to the program left and right mixing buses. It may be used to apply an effects signal to the stereo mix — effects which are independent of the built-in reverb or of any external signal processing patched between the EFFECTS OUT [43] and EFFECTS RETURN [35] jacks.

41. PROGRAM 1 RETURN

This jack is identical to the PROGRAM 2 RETURN [40], except it applies signal to the PGM 1 RETURN controls [16].

Panned full left and full right, the PROGRAM 1 & 2 RETURN jacks may be used as sub inputs to link the stereo mixing buses from a submixer (such as another MX4200 series console's LEFT OUT and RIGHT OUT jack) to create a larger mixing system.

42. AUX OUTPUT

This unbalanced phone jack delivers the post AUX SEND control [10] line-level output of the aux mixing bus to a signal processor, mixer or power amplifier. The nominal output level is –6 dBu.

43. EFFECTS OUTPUT

This unbalanced phone jack delivers the post EFF SEND control [9] line-level output of the effects mixing bus to a signal processor, mixer or power amplifier. The nominal output level is +4 dBu.

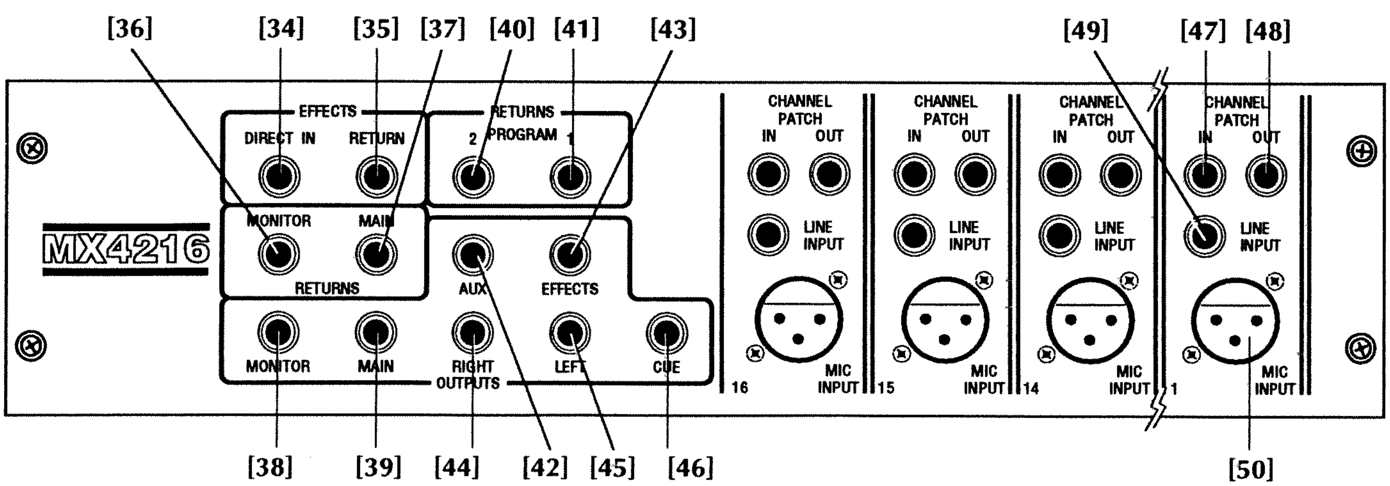


Figure 2-3. MX4200 Connector Panel Detail

44. RIGHT OUTPUT
This unbalanced phone jack delivers the post master fader [24] line-level output of the right program mixing bus to an external power amplifier, mixer or signal processor. The nominal output level is +4 dBu.

45. LEFT OUTPUT
This unbalanced phone jack delivers the post Master Fader [24] line-level output of the left program mixing bus to an external power amplifier, mixer or signal processor. The nominal output level is +4 dBu.

46. CUE OUTPUT
This unbalanced phone jack delivers the post CUE/PHONES control [11] line-level monaural output from the cue circuitry to an external power amplifier or another mixer (for expansion of the system). The nominal output level is +4 dBu.

NOTE: Items 47 through 50 are repeated for each of the 8, 12 or 16 input channels in the console.

47. CHANNEL PATCH IN
This unbalanced phone jack applies signal to the input channel just ahead of the fader. Nominal input level is -6 dBu. Generally, the CHANNEL PATCH IN is used for the return from a signal processor (such as a compressor or noise gate) which is used to process the channel's signal. Alternately, the jack may be used as a special line input to replace any signal applied to the channel's balanced LINE INPUT [49] or MIC INPUT [50]. (Inserting a phone plug in this jack breaks the internal signal flow from points ahead of it in the channel circuitry.)

48. CHANNEL PATCH OUT
This unbalanced phone jack outputs the signal from the input channel (just after the EQ and PEAK LED, but before the fader). Nominal output level is -6 dBu (388 mV). The CHANNEL PATCH OUT jack may be used as an auxiliary output to another console or as a direct out to a multitrack tape machine. It is intended primarily for sending the input channel signal to an auxiliary signal processor (compressor, graphic EQ, noise gate, etc). CHANNEL PATCH OUT is always *live* regardless of the setting of the channel fader [8], although it is affected by the TRIM control [1].

NOTE: These CHANNEL PATCH OUT and IN jacks are normalled so that when nothing is plugged in, signal flows across the jacks internally. Inserting a plug into the OUT jack does not alter the internal signal flow, but merely splits signal out in parallel. However, inserting a plug into the IN jack interrupts the internal signal flow through the channel, instead bringing in the return from an auxiliary signal processor.

49. LINE INPUT
This electronically balanced tip/ring/sleeve phone jack accepts a line-level signal and applies it, via a 20 dB pad, to the channel preamplifier. With the 40 dB of trim range, it is thus possible to accommodate signals of nominal levels from approximately –32 dBu (*very low* line level or *very hot* mic level) to +8 dBu (broadcast high line level). Maximum input level is +32 dBu with 40 dB of attenuation.

50. MIC INPUT
This female XLR accepts a microphone-level signal and applies it directly to the channel preamplifier. The sensitivity (i.e., the nominal input level required to produce a nominal output level) may vary from -52 dBu to -12 dBu depending on the setting of the channel's TRIM control [1]. This provides 26 dB of headroom.

51. POWER
This rocker switch turns on the AC input to the console's built-in power supply, and thereby provides the necessary voltages to the console circuitry. The front-panel POWER LED [33] is illuminated when power is on (if the LED is not illuminated, check the AC outlet to be sure it is live, or check for a blown fuse [53] in the console if the outlet is live and power still does not appear to be on.)

52. PHONES
(Not illustrated) This 1/4" tip/ring/sleeve (stereo) phone jack (located on the right, front edge of the console) can accommodate a pair of 600–ohm or higher impedance stereo headphones. Do not use with headphones having an actual load impedance of 8 ohms, as this will overload the output driver stage.

53. FUSE
A fuse protects the primary of the MX4200 power supply. It should be replaced only with a fuse of the same current rating and type. All three MX4200 models which operate on 120V AC utilize a 1A, Slo-Blo type fuse (120 or 240 volt fuse rating). Models intended to operate at different AC line voltages will use different fuse ratings; be sure to replace a blown fuse only with one of the identical type and current rating.

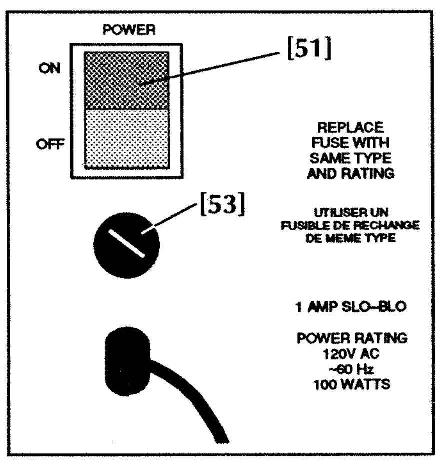


Figure 2-4. Detail of Rear Panel Power Switch and Fuse Area

SECTION 3 SPECIFICATIONS

Frequency Response	20 Hz to 20 kHz, ±1 dB
THD	Less than 0.05% from 20 Hz to 20 kHz @ +8 dBu output
Noise	Less than –90 dBu with Master Faders down –122 dBV Equivalent Input Noise
Common Mode Rejection Ratio	Greater than 60 dB @ 1 kHz
Crosstalk	–60 dB @ 1 kHz
Voltage Amplification	76 dB ± 2 dB (maximum) @ 1 kHz Lo–Z Input to Left, Right, Effects output 86 dB ±2 dB (maximum) @ 1 kHz Lo–Z Input to Main output 66 dB ±2 dB (maximum) @ 1 kHz Lo–Z Input to Monitor and Cue Output
Maximum Input Level	
Lo–Z Input	+14 dBu (3.88 V rms)
Hi–Z Input	+35 dBu (43 V rms)
Equalization	
Input Low EQ	±15 dB @ 50 Hz, shelving
Input Mid EQ	±12 dB @ 3 kHz, peaking
Input High EQ	±15 dB @ 10 kHz, shelving
Faders	60mm throw, carbon–type
Indicators	
Peak Indicator	Each input channel
VU Ladder Displays	Switchable Main/Left, Monitor/Right; 0 VU = 0 dBu output level (open circuit).
Phantom Power	LED
Power	LED
Inputs	
Lo–Z	Balanced (transformerless) 6 kohm actual impedance. –52 dBu sensitivity (TRIM at 0), +12 dBu maximum input level (TRIM at –40)
High–Z	Unbalanced 10 kohm actual impedance. –32 dBu sensitivity (TRIM at 0), +32 dBu maximum input level (TRIM at –40)
Outputs	
Left, Right, Main Monitor, Effects, Aux, Cue	Unbalanced, 150 ohm source impedance. +4 dBu nominal level (except Monitor and Aux are –6 dBu), +18 dBu maximum output level
All Patch Points	Unbalanced, 150 ohm source impedance, –6 dBu nominal level
Headphones	For 600 ohm or higher impedance stereo headphones
Mixing Buses	Main, Monitor, Left, Right, Effects, Auxiliary, Cue
Phantom Power	+15 volts DC on pins 2 and 3 of each Lo–Z Mic input (pin 1 common)
Reverb	Built-in spring type unit, as well as provisions for external signal processing.
Connectors	
Channel Mic & Line Inputs	Mic: 3–pin female XLR (pin 2=high), Line: TRS Phone (Tip=high, Ring=low)
Headphone	TRS Phone; Tip=left, Ring=right, Sleeve=common (shield)
All other connectors	TS Phone; Tip=high, Sleeve=low (shield)
Power Consumption	120 volts (±10%), 60 Hz, 1 amp slo-blo fuse (U.S. & Canadian Models)

Dimensions & Weight	**MX4208**	**MX4212**	**MX4216**
Depth	24.0 inches	24.0 inches	24.0 inches
Width	22.25 inches	28.25 inches	34.25 inches
Height	6.25 inches	6.25 inches	6.25 inches
Weight	28 pounds	32 pounds	36 pounds

Finish	Solid wood side panels, gun–metal grey, color–coordinated controls

0 dBu is referenced to 0.775 volts rms. 0 dBV is referenced to 1.000 volts rms.

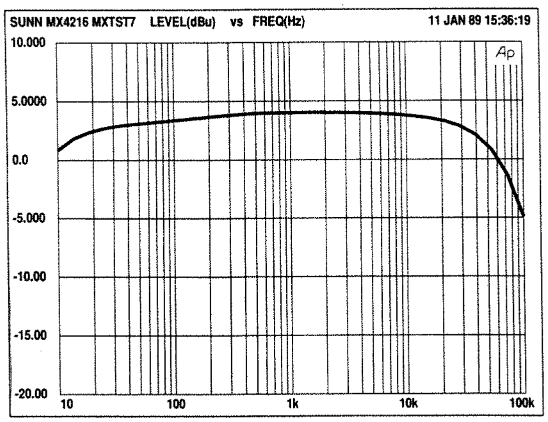


Figure 3-1. Amplitude vs Frequency Plot

Output amplitude set to +4 dBu.
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 20 dB attenuation.

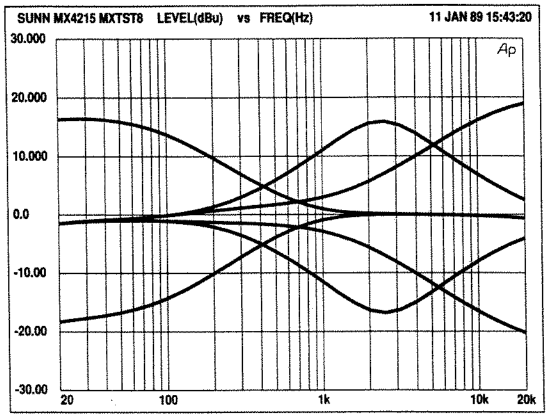


Figure 3-2. EQ Frequency Response Curves

Output amplitude set to 0 dBu.
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 20 dB attenuation.

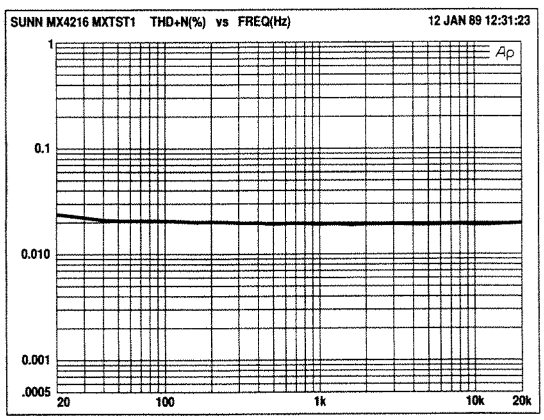


Figure 3-3. T.H.D. vs Frequency (20 Hz to 20 kHz)

Output amplitude set to +8 dBu.
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 0 dB attenuation.

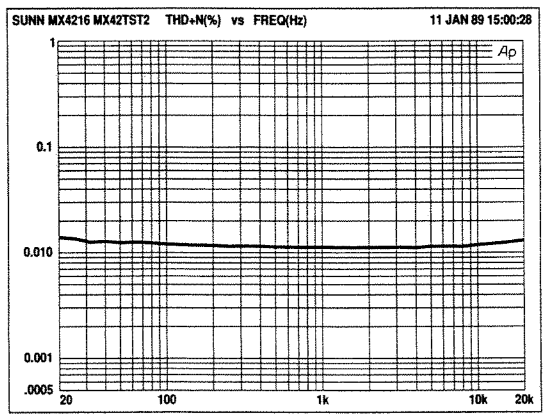


Figure 3-4. T.H.D. vs Frequency (20 Hz to 20 kHz)

Output amplitude set to +8 dBu.
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 20 dB attenuation.

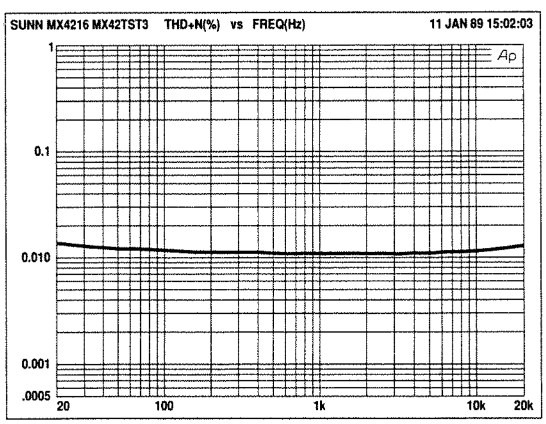


Figure 3-5. T.H.D. vs Frequency (20 Hz to 20 kHz)

Output amplitude set to +8 dBu.
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 30 dB attenuation.

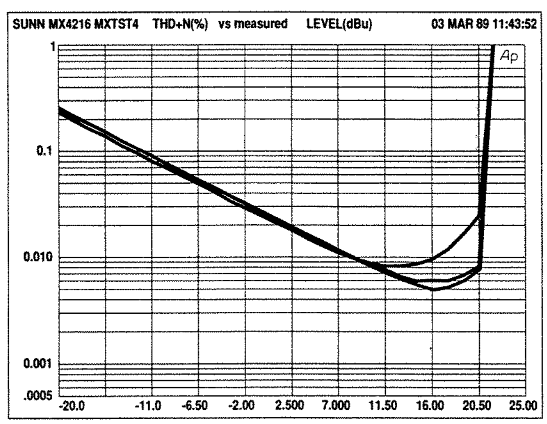


Figure 3-6. T.H.D. vs Amplitude (at 20 Hz, 2 kHz and 20 kHz)

Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 20 dB attenuation.

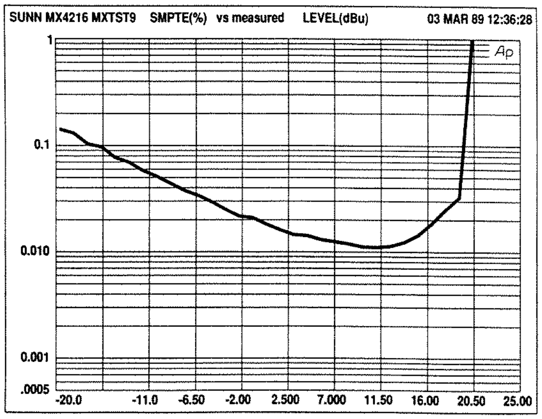


Figure 3-7. SMPTE I.M. Distortion vs Amplitude
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 20 dB attenuation.

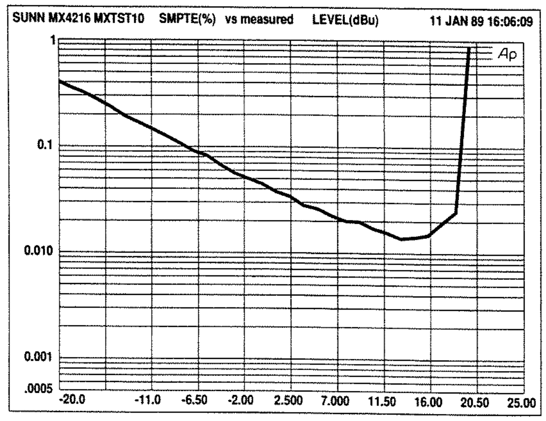


Figure 3-8. SMPTE I.M. Distortion vs Amplitude
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 0 dB attenuation.

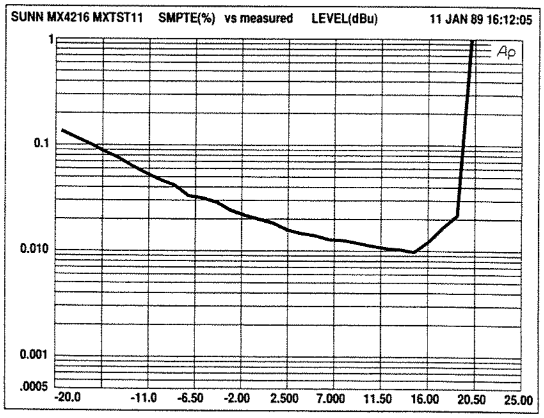


Figure 3-9. SMPTE I.M. Distortion vs Amplitude
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 40 dB attenuation.

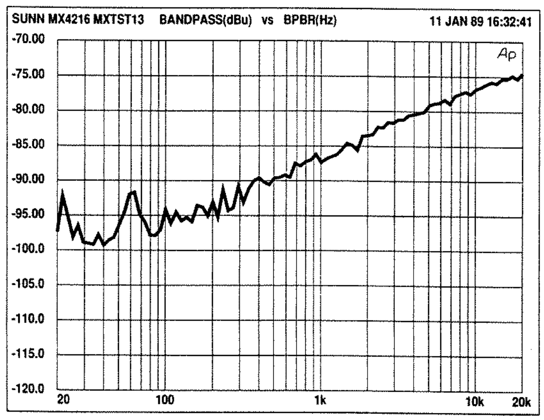


Figure 3-10. Output Noise Spectrum
Channel 1 MIC INPUT assigned to LEFT OUTPUT.
Channel 1 MIC INPUT terminated with 150 ohms.
Channel 1 Fader set at 10 dB attenuation.
LEFT Fader set at 10 dB attenuation.
Channel TRIM set at 0 dB attenuation.

SECTION 4 INSTALLATION

4.1 PLANNING AN INSTALLATION

In preparation for installing the MX4200, consider how it will be used, how it is going to be connected, and what is the best way to implement the installation.

The desk or table top on which support the console is placed should be capable of supporting at least the weight of the console plus a human console operator leaning on the wrist rest. Allow adequate access behind the console for cable connections and for service loops (extra cable) so that the console can be moved without disconnecting everything. It is advisable to prepare a detailed block diagram of the entire sound system prior to installation. From this you can figure out all the necessary cables, where they run, and the required length so that the cables can be prepared ahead of time.

4.2 POWER MAINS VOLTAGE

The power supplies for MX4200 consoles sold in the U.S.A. and Canada are designed to operate with 110 to 120 volt, 50 or 60 Hz AC power mains. Export models operate on 220 or 240 volt, 50 or 60 Hz AC mains. If you are traveling with this equipment, be sure to test the power mains, and to use the appropriate power supply (consult your Sunn dealer for assistance). If the power line voltages do not fall within the allowable range, do not connect the MX4200 to the mains. Instead, have a qualified electrician inspect and correct the condition. Failure to observe this precaution may damage the console, and will void the warranty.

4.3 EARTH GROUND

The console must be grounded for safety and proper shielding. A 3–wire power cable is provided for this purpose. Use a special circuit tester to insure that the outlet is properly grounded, and that the AC 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. Beware of using cold water pipes (water meters are often electrically isolated by plastic pipe), and avoid using gas pipes as a ground since a loose or broken pipe and a spark could spell disaster.

4.4 AC SAFETY TIPS

1. If you are going to verify the quality of AC wiring, there are two inexpensive items you should carry. One of these is a commercial outlet tester, the other is a neon lamp type AC voltage tester. These items are inexpensive and available at most hardware stores, electrical supply houses and some lighting stores. It is advisable to also have an rms (or averaging) voltmeter to measure the exact AC line voltage.

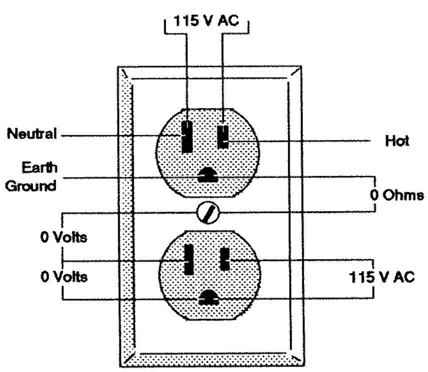


Figure 4-1. Testing an AC Outlet for Proper Voltage & Wiring

2. Use the outlet tester to examine all power outlets. The neon voltage tester should be used to check for voltage differences between microphone and guitar amps, microphones and electric keyboard chassis, and so forth.
3. If you're not sure whether an outlet is good, don't use it. In anticipation of finding a bad outlet, carry a long, heavy duty extension cord. A good extension should be made of #12-3 (12 gauge, 3 wires), and no longer than 15 meters (about 50 feet).
4. If there is no suitable power source at a venue, don't plug in your equipment. Any fault in the wiring of the AC outlet is potentially hazardous. Rather than take a chance with damage to equipment and possibly lethal shock, it is best to refuse to use a faulty outlet until it has been repaired by a licensed electrician. *Don't take unnecessary risks.*

4.5 BALANCED AND UNBALANCED LINES, AND GROUND LIFT SWITCHES

In certain situations you can lift (disconnect) the shield at one end, usually the output, of an audio cable and thus eliminate the most likely path that carries hum-inducing ground loop currents. In a balanced line 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. Some professional audio equipment is equipped with ground lift switches on the balanced inputs so ground lifting can be used when multiple unbalanced audio cables join two pieces of equipment. In this case, all but one of the shields can be lifted, thus maintaining the low side of the audio connection without unnecessary duplication, which avoids the ground loops and induced noise. If you want to avoid ground lifting, try tightly bundling the cables.

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

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

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

4.6 AUDIO CONNECTORS AND CABLES

Use of low quality or 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 your sound equipment is to be realized.

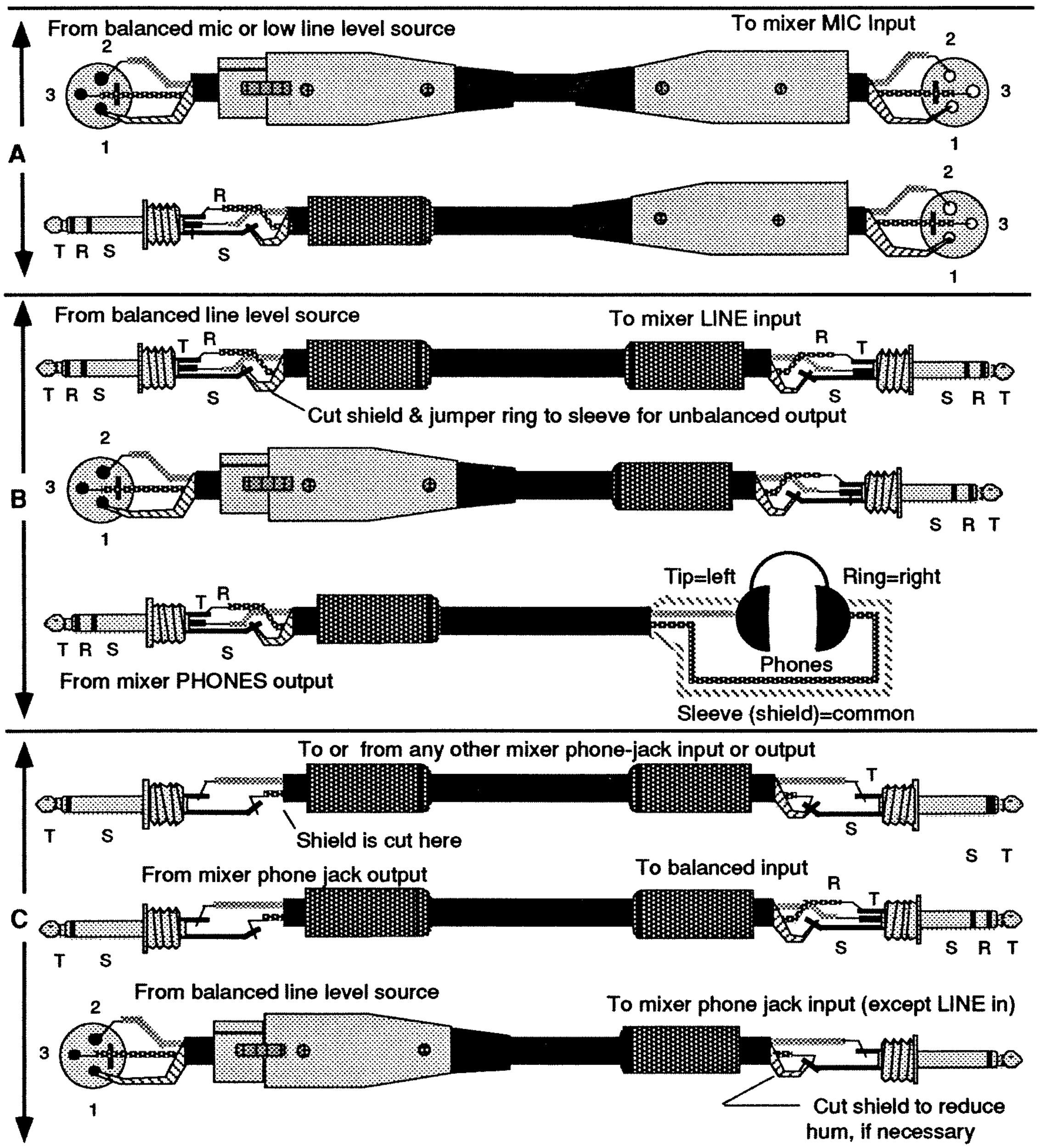


Figure 4-2. Connector Wiring for MX4200: (A) XLR-3 (Microphone Inputs) (B) T/R/S Phone Plug (Line Inputs and Phones Output) (C) T/S Phone Plug (all other Inputs and Outputs)

The MX4200 is fitted with only three types of audio connectors: 3-pin female XLRs, 2-circuit (tip/sleeve) 1/4" phone jacks, and 3-circuit (tip/ring/sleeve) 1/4" phone jacks (also known as stereo phone jacks).

4.6.1 About Balanced Cables

Two-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 equipment confined to portable racks or cases, lighter duty cables such as Belden 8451, Canare L-2E5AT or an equivalent are suitable. Snakes (multi-core cables containing multiple shielded pairs) must be handled very carefully because the leads tend to be fragile, and a broken conductor cannot be repaired. If you are using a snake, allow at least one or two spare channels that can be used in case of breakage in one of the channels in use.

4.6.2 Layout

Never run AC power lines in the same conduit, or even closely bundled, with audio cables. Hum can be induced from the relatively high voltage AC circuits into the lower voltage audio circuits. A heavy 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 the AC cables and audio lines by as wide a distance as is practical, and where they must cross, try to have them do so as close to a right angles as possible, which minimizes induced noise.

Similarly, avoid closely bundling the line-level outputs from the MX4200 with any mic-level inputs to the console. We recommend that you avoid using a snake 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 will be established. This will not always be manifest as audible howling, but instead may be manifest as very high frequency oscillation that indirectly causes distortion of the signal and that can precipitate premature component failure. Again, 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 the cables run between the power amp output and the speakers be separated from mic- or line-level cables. Speaker cables should be treated much like AC cables. If they cross other audio cables, they should do so at right angles. If they must be run along the same path, they should not be bundled tightly.

4.6.3 Balanced and Unbalanced Wiring

There are two basic 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 has no particular relationship to whether or not the line is balanced.

An 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.

A 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.

Balanced wiring is more expensive to implement than unbalanced wiring. It offers advantages, especially in portable sound systems, that may offset the added cost. There is nothing inherently better or more professional about balanced wiring so the nature of the application should guide the selection.

Unbalanced wiring works best when high quality cable is used, the cable runs are relatively short, and one branch of the AC power system feeds all the interconnected audio equipment. Such wiring is commonplace for broadcast signal transmission and laboratory test equipment.

Balanced wiring is useful for excluding some types of externally-generated noise. The two wires of a balanced cable both 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 (hence the term *balanced differential input*). Should any electrostatic 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, which is sensitive to differing voltages across the two wires. (This is why the term *common mode rejection* applies; signals in common to the two center wires are rejected.) The MX4200 utilizes balanced circuits for the channel mic and line inputs. These preserve the advantages of balanced sources, yet will accept unbalanced sources (though if you connect an unbalanced source you will lose the noise immunity advantages of a fully balanced circuit).

SECTION 5 OPERATING LEVELS

5.1 WHERE TO BEGIN

Level control, trim control and fader settings throughout a sound system can dramatically affect the signal-to-noise ratio and distortion of the system. These settings should be optimized for each component in the system. A 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 channel. Set the input 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 red PEAK LEDs, and in some cases it can be heard by listening for distortion while making TRIM adjustments. The next step is to set the level of the console input channel (the channel fader and/or the appropriate send control) so that it properly drives the mixing buses. ***Begin with the faders set to 0, and other level controls set at about 8 (roughly at 4 o'clock knob rotation).*** You can refer to the VU bar-graph meters to examine the bus levels.

If line amplifiers, electronic cross-overs, 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.

5.2 SETUP PROCEDURE

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

Let's go through an actual setup procedure for a high quality, music reproduction system. First choose a headroom figure. To balance reasonable fidelity with moderate system costs, we will allow 16 dB of headroom above the average system output. While some extreme musical peaks reach or exceed 20 dB, the 16 dB figure is reasonable for most programs. A 16 dB headroom figure represents a peak level that is forty times as powerful as the average program level.

Remember that with a 16 dB headroom figure, a power amplifier as powerful as 250 watts will operate at an average 6.25 watts output power. In a surprising number of sound systems, given reasonably sensitive loudspeakers, this low average power will prove to be adequate. If you need 20 dB of headroom, you will have to increase to over 600 watts worth of power amplifier, or keep the 250 watt amp and cut the average power to a quarter of a watt (or use 4 dB more sensitive loudspeakers, or use some combination of these techniques).

After choosing a headroom figure, adjust the incoming and outgoing signal levels at the various devices in the system to achieve that figure. The adjustments for our chosen 16 dB headroom figure would be made as follows:

1. Initially, set the attenuators on the power amp at maximum attenuation (usually maximum counterclockwise rotation). Feed a sine wave signal at 1000 Hz to the console MIC INPUT [50] at an expected average input level (approximately –50 dBu (2.45 mV), or to the LINE INPUT [49] at from –20 dBu (388 mV) to +4 dBu (1.23 volts). The exact voltage is not critical, and 1000 Hz is a standard reference frequency, but any frequency from 400 Hz to about 4 kHz may be used.
2. With the channel EQ controls [3] centered, adjust the channel TRIM control [1] until the PEAK indicator [6] just begins to flash. Then turn down the TRIM 13 dB. (Note, since full rotation of the TRIM control represents a 40 dB range, a bit more than 1/4-turn reduction is equal to 13 dB). You can also use the output VU meters to get a good approximation by setting the channel and Left or Right Faders to give a good on-scale reading on the VU meter. Since there remains 3 dB of headroom above the point where the PEAK LED just flashes, this establishes 16 dB of headroom at the input stage of the console.
3. Set the input channel FADER [8] on the console at its heavy lined nominal setting of 0, and adjust the PAN control [5] all the way to one side (let's use the left bus for now, so turn PAN fully counterclockwise).
4. Set the LEFT program master fader [24] to nominal 0 dB position (heavy line).
5. Make sure the left VU meter [28] is set to indicate the left output signal by ensuring the MAIN/LEFT meter select switch [30] is engaged. Examine the meter. It should

indicate a level which is 16 dB below the rated maximum output level for the console.

The maximum rated output level is +18 dBu (6.16 volts) so for 16 dB headroom, the output level should be adjusted to +2 dBu (0.97 volts). Remember, this is a steady-state sine wave signal, not random noise or program material which varies in peak level. If necessary, adjust the channel TRIM control [1] to achieve this meter indication. While you could adjust the master fader, the whole idea of this procedure is to establish an initial setup with the faders and other level controls set at their nominal positions.

6. Finally, starting with the input 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 1/40 of the maximum rated power, the amplifier has 16 dB headroom left before clipping. A 250 watt amplifier would operate at nominal 0.25 watts on average level passages in order to allow 16 dB for the loud peaks.

7. If you connect an external signal processor in the channel's PATCH OUT/IN loop [47, 48], its levels should be adjusted correctly to ensure proper headroom, too. The nominal level at the patch output is –6 dBu (388 mV), but since the trim is adjusted 16 dB below nominal, the PATCH OUT jack [48] will be carrying a nominal -22 dBu (616 mV) signal level.

   A number of signal processors have nominal input sensitivity of –10 dBu to –20 dBu, as they are typically operated at unity gain. If the sensitivity is nominally –10 dBu, then the –22 dBu signal from the console with not be driving the processor adequately, and excess noise will result. In this case, where possible, turn up the external signal processor's input gain to compensate for the lower input level.

   If there is no input gain adjustment on the signal processor, you can turn up the console's TRIM control somewhat (perhaps 6 to 8 dB), allowing the higher level to enter the signal processor, and then turn down the signal processor's output level by a corresponding amount (6 to 8 dB) so that the signal returned to the console's CHANNEL PATCH IN jack [47] does not exceed the desired headroom figure. Bear in mind that this latter technique sacrifices headroom in the console's channel equalizer, and that if you apply much EQ boost, the equalizer may introduce clipping distortion even when no other stage in the equipment is overdriven.

   If the signal processor operates closer to –20 dBu nominal input sensitivity, then its input and output levels should be suitable without further adjustment of the console. Bear in mind that many signal processors operate at +4 dBu nominal level, and exhibit 0 dB gain (i.e., they do not change the average signal level). While such signal processors will work OK when patched into the nominal –22 dBu CHANNEL PATCH circuit described here (assuming, of course, the impedances are correct), you may hear more noise than you would like because the signal is 26 dB closer to the noise floor of the signal processor than it would be if it were at the +4 dBu nominal level of the signal processor. This is why it is best to use a properly matched signal processor, one whose input sensitivity and output level are adjustable to function in the –20 dBu to 0 dBu region.

To operate this system, use primarily the Master Faders or master output level controls on the console, and avoid levels that consistently turn on the console's VU meter above the zero LED, 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, as you add more channels to the mix, you notice distortion, the chances are that the distortion will not be reduced by turning down the master fader(s), though you can certainly try that first. This is because, while individual channels may not be distorting (their PEAK LEDs are not lighting up much), the combined levels on the mixing bus(es) add up to the point where they overdrive the bus summing amplifier(s). Non-indicated (i.e., non metered) summing bus overload is only possible if the Master Fader is set below –10 dB on the Fader scale. At this setting, the gain is unity beyond the summing amp[lifiers, and the VU meters will be off scale. In the event of summing amp overdrive, the solution is to bring down all channels' controls which contribute to the offending mix. For example, bring down all the contributing channels' faders a couple of dB. This will reduce the bus levels, and prevent the summing amp from clipping. You then will not have to change the master fader setting or the power amplifier setting.

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. While you can make an occasional TRIM adjustment during an actual performance, don't ride gain with the TRIM controls. Use the Faders or other level controls to adjust levels during a performance for greater control resolution. Compressors can be used in the CHANNEL PATCH IN/OUT loops of individual channels (say for a vocalist with widely varying levels), or after the console 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 retained while raising the *average* power level.

SECTION 6
OPTIONAL PRE-POST FUNCTIONS

CAUTION: These procedures are to be performed only by a qualified service technician.

The signal sources from which the AUX send control [3] and MONITOR send control [4] on each input channel are derived may be altered on a channel-by-channel basis. As shipped from the factory, these sends are derived *post* (post fader and EQ for the AUX send, post EQ and pre fader for the MONITOR send). Plug-in jumpers which slide onto 3-pin headers on the internal circuit boards may be moved to change the signal derivation. The AUX send can be changed to be pre fader but post EQ, making it the same as the factory-shipped source for MONITOR send and effectively creating two monitor mixes. The MONITOR send can be changed to be Pre fader and pre EQ, thereby creating a mix which is independent of all other channel settings except TRIM and MONITOR.

CAUTION: In order to move the jumpers, the console must be partially disassembled. This work should be done only by a competent technician, and only after carefully studying these instructions.

WARNING: Lethal voltages exist inside the MX4200 console. Be sure to unplug the power cord and wait at least 5 minutes before opening the unit. Follow instructions carefully, and if you do not understand something, contact your Sunn dealer or service facility for assistance... do not make assumptions.

You will need a work bench or table top large enough to easily accommodate the console. Spread out a towel or pad to protect the working surface and the console, since you will be turning the console upside down for part of this procedure. You will also need a 10mm nut driver, a 13mm (or 1/2") nut driver, and a #2 phillips screwdriver. To alter the functions as discussed above, turn off the console POWER, unplug its power cord, and then open the console as follows:

1. Identify those input channels you plan to modify. Internally, the MX4216 circuit boards are arranged in groups of four adjacent channels, as follows: 1-4, 5-8, 9-12, 13-16. (Obviously, the MX4212 does not have 13-16, and the MX4208 has neither those channels nor 9-12.)
2. Pull all the round control knobs and the fader knobs off of any group of four input channels on which even one channel will be altered. If you have difficulty removing a knob, protect the console front panel with a sheet of paper, slide a screwdriver under the knob, and gently pry up while pulling on the knob.

   NOTE: We strongly suggest that you make note of positions of the various knobs, or organize them in a way that matches the front panel layout, so later on you can reinstall them correctly. The TRIM knobs are light gray, the equalizer (HF/MF/LF) knobs are light blue, the EFF/REV-AUX and MONITOR knobs are dark blue, and the PAN knobs are tan.
3. Unscrew the identification trim strip located at the front edge of the console atop the wrist pad. On the MX4216 there are 6 screws #6 flat-head phillips machine screws; the MX4212 has 5 screws, the MX4208 has 4 screws. Set the trim strip and screws aside. The top edge of the wrist rest should now be exposed.
4. Remove all 6 (or 5 or 4) #8 screws on the bottom edge of the wrist rest, but ***do not yet attempt to move the wrist rest away from the console.***
5. Using the 13mm (1/2") nut driver, unscrew the retaining nut from the headphone jack on the right front corner of the wrist rest. Remove the nut and washer, and push the jack out of the wrist rest. Now you can set the wrist rest aside.

   NOTE: There is no need to remove the side trim panels. Also, leave the 3 screws along each side of the top panel and the 2 screws along each side of the rear connector panel intact. They will not be removed at any time during this procedure.
6. With the wrist rest removed, you can observe the circuit boards protruding from beneath the front panel, and the ribbon cables plugged into these boards. Just below the MASTER SECTION there is a 16-pin ribbon cable plugged into a connector which runs from left to right along the frontmost edge of the Master circuit board (the cable is folded under the board, and runs toward the back of the console). This is the power connector. Unplug it from the front of the board (pull it straight out).
7. A 16-conductor ribbon cable runs from left to right across most of the console. The cable has a series of plugs, one of which is inserted into a mating connector on right edge of each of the 4-channel circuit boards. For those channels on which you plan to make any modifications (and from which you should have removed the control knobs), unplug the ribbon cable by pulling the connector straight up. Do not pull on the ribbon cable, but instead grasp the edges of the connector itself. (Be careful not to grasp the larger mating connector, which is soldered to the circuit board.)
8. Turn the console upside down, and place it face-down on bench with the rear connector panel facing away from you.
9. Remove all 6 (or 5 or 4) screws along the back edge of the top panel (where it bends down) which is now farthest away from you.
10. Remove the 6 #8 phillips screws from along each side of the bottom panel. This would seem to allow the bottom panel to be lifted off, but it is not yet free. ***The reverb unit wiring must first be unplugged.***
11. Lift the bottom panel a little way so you have access to the reverb unit. As you are now facing the console, the reverb is located at the left front corner of the console (beneath the master fader area). Unplug the two phono-plug type cables from the front of the reverb pan (red input from the left, black output from right). Now you can lift the bottom panel of the console the rest of the way off and set it aside.
12. Turn the console over again (right side up). Use the 10mm nut driver to remove the nuts and washers for all 28 pots on the four channels for a given circuit board. Then remove the two #4-40 screws along the

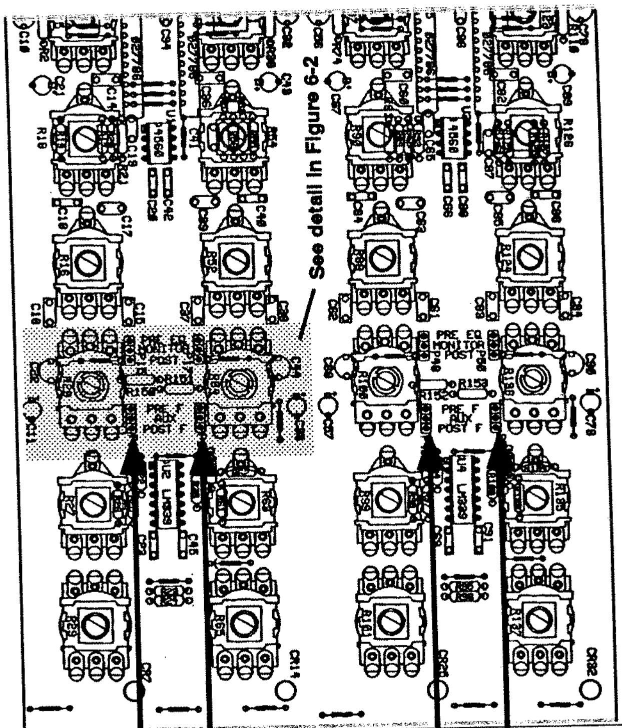
front edge of the console above this circuit board (these screws were exposed when you removed the trim strip with the channel labels). These screws loosen the bracket which holds the circuit board.

13. Use the 1/2" (13mm) nut driver to remove the nuts from the 12 rear panel phone jacks associated with the four channels. Also, use the phillips screwdriver to remove the 8 #4 thread-forming screws which secure the 4 XLRs to these channels. Now the input board should drop out onto the bench, though it is not entirely free of the console yet.
14. Reach behind the console, and you will find a green ground wire running from the circuit board to a point between the center pair of XLR connectors for the four channels. Remove the ground wire by lifting it straight off the board.
15. Now the circuit board is free. Lift the console up and withdraw the board.
16. Refer to Figure 6-1. Locate the 3-pin headers which are adjacent to the EFF/REV-AUX pot [3] on each channel. Headers P16, P18, P49 and P51 are labeled PRE F/AUX/POST F, and determine whether the AUX send control [3] is post fader (the factory preset) or pre fader. P15, P17, P48 and P50 are labeled PRE EQ/MONITOR/POST and determine whether the MONITOR send control [4] is post EQ (the factory preset) or pre EQ.
17. Once you have located the jumper corresponding to the channel and control you wish to alter, simply pull the jumper off, and slip it over the other pair of pins on the 3-pin header, as illustrated in Figure 6-2. When you are done, close up the console in the reverse order of these instructions before reapplying power.

NOTE: After removing the input module(s), if you have removed the circuit board for the master section, you will find it easier to reinstall if you first pull off the square pushbuttons from the various CUE switches, meter switches and so forth. Then, after the board is secured back onto the front panel, you can press these buttons into place through the front panel. Attempting to fit the board to the front panel with the pushbuttons in place can be very frustrating due to tight tolerances.

18. When you first turn on the console after these modifications, we recommend that the power amplifier(s) be turned off (or set at minimum volume) just in case you managed to make a wiring error. That way, even if you improperly reassembled the console, you will have an opportunity to discover the error and correct it before any loudspeakers are damaged.

Make note of any jumpers you move, and paste a label somewhere on the console so that any operator can determine how the console will function. A permanent marking pen on a strip of gaffers tape, applied to the rear panel or bottom panel, will suffice. If many different people use the console, or if only a few channels' functions are altered, you may wish to create front-panel labels with Dymo™ tape, Kroy™ lettering, or some neatly lettered, reasonably durable method.



CH 1	CH 2	CH 3	CH 4	MX4208, 12 & 16
CH 5	CH 6	CH 7	CH 8	
CH 9	CH 10	CH 11	CH 12	MX4212 & MX4216
CH 13	CH 14	CH 15	CH 16	MX4216 Only

Figure 6-1. Location of Headers & Jumpers for Pre-Post Functions.

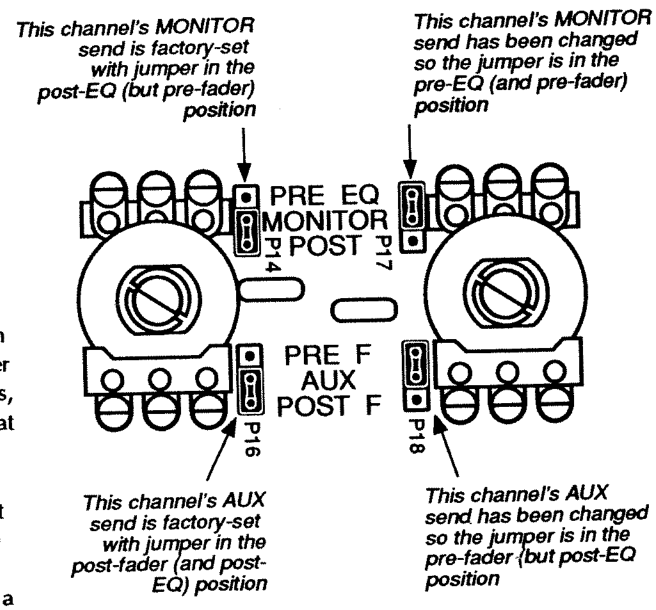


Figure 6-2. Detail of Header & Jumper Installation

NOTES

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