

Mackie 8.b Audio Mixer Essentials

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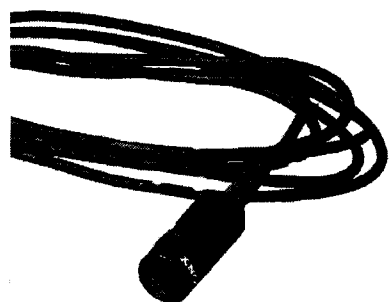
August 24, 2008

BCAT AUDIO MICROPHONES



Shure SM58

Transducer: Dynamic
Polar Pattern: Cardioid
Frequency Response: 50Hz- 15kHz
Output connectors: XLR-3M
Output impedance: 150 Ohms
Retail price: \$99.00



Sony-ECM 44B & 66B

44b

Transducer: Condenser
Polar Pattern: Omni-directional
Frequency Response: 40Hz- 15kHz
Output connectors: XLR-3M
Output impedance: 250 Ohms
Retail price: \$194.95

66B

Transducer: Condenser
Polar Pattern: Uni-directional
Frequency Response: 30Hz- 18kHz
Output connectors: XLR-3M
Output impedance: 100 Ohms
Retail price: \$370.00



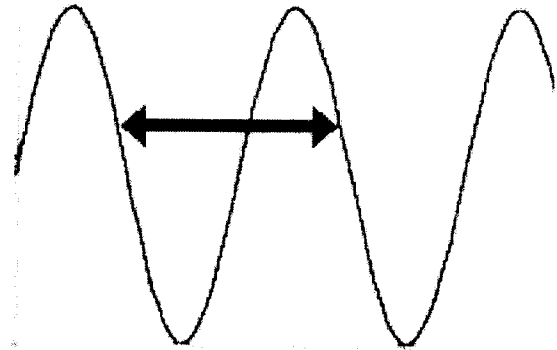
Shure SM89

Transducer: Condenser
Polar Pattern: Cardioid
Frequency Response: 60Hz- 20kHz
Output connectors: XLR-3M
Output impedance: 150 Ohms
Retail price: \$579.95

Sound wave properties

Wavelength

Wavelength: The distance between any point on a wave and the equivalent point on the next phase. Literally, the length of the wave.

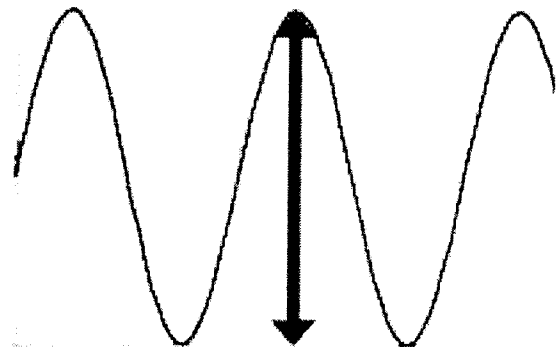


Amplitude

Amplitude: The strength or power of a wave signal. The "height" of a wave when viewed as a graph.

Higher amplitudes are interpreted as a higher volume - hence the name "amplifier" for a device, which increases amplitude.

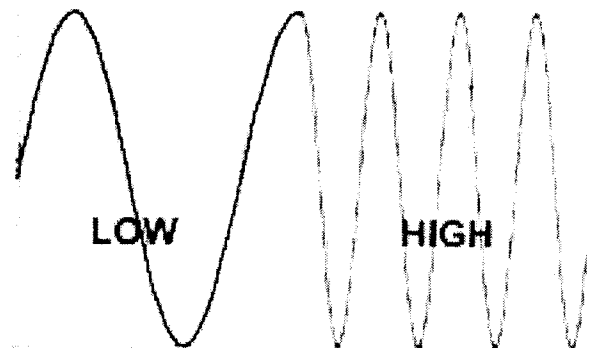
The opposite of amplitude is attenuate, which decreases the power of the signal.



Frequency

Frequency: The number of times the wavelength occurs in one second. Measured in Hertz (Hz) or cycles per second. The faster the sound source vibrates, the higher the frequency.

Higher frequencies are interpreted as a higher pitch. For example, when you sing in a high-pitched voice you are forcing your vocal chords to vibrate quickly.



**Brooklyn Community Access Television
Glossary of Audio Terms**

Amplitude: the strength or power of a sound wave signal

Attenuate: decreases the power of the signal

Assign: to select where a channel is to be sent

Audio: sound converted into electrical energy

Bandwidth: a measure of frequency range, measured in hertz

Center Frequency: frequency of the audio signal that is most boosted by an EQ

Channel: the single path that an audio signal travels through a device from an input to an output

Decibel (dB): a measure of the loudness of sound

Direct Box: device to directly wire the output of an electric instrument (such as an electric guitar) to a mic input

Distortion: garbled sound heard when an audio overloads an input

Equalization: processing specific sets of frequencies to increase or decrease more than other frequencies

Fader: sliding control on a channel of the mixer that determines the audio level for that channel

Feedback: often caused when a mic picks up its own signal from a speaker

Frequency: the number of cycles (or wavelength occurrences) that occur in a second, measured in Hz.

High Frequency (High-end) sounds are above 6000Hz or 6kHz

Low frequency (Low-end) sounds are below 1000Hz or 1kHz

Mid-range Frequencies are between 250Hz and 6000Hz

Gain: amount of audio boost, measured in dB.

Group: an array of output channels controlled by a set of sliders.

Hertz (Hz): unit of measurement for frequency or cycles per second

Headroom: The safe range between your optimum line level and the level of distortion

Impedance: measures the amount of opposition a device has to an AC current (like an audio signal). Generally, low impedance devices are better than those with high impedance

Input: refers both to the source and the connector that receives the source.

Level: the amount of a signals' strength.

Meter: device that measures and displays the level of an input or an output.

Microphone (mic): Device that transduces (converts) sound waves into electrical energy.

Mix: (verb) to blend various audio signals into one signal; (noun) the signal made by blending various signals together.

Mixer: both the console that blends together audio signals but also the person who mixes.

Monitor: to listen or the device used to listen.

Monophonic (mono): one channel audio can be either input or output.

Mute switch: turns off a channel; removes a signal from the monitor; or turns off the monitor signal.

Octave: in music, is the interval between one musical note and another with half or double the frequency. For example, if one note has a frequency of 400 Hz, the note an octave above it is at 800 Hz, and the note an octave below is at 200 Hz. The ratio of frequencies of two notes an octave apart is therefore 2:1.

Output: the jack or physical location where a signal is sent or the signal that is sent out by a device

Pan pot: rotating control used to distribute one audio signal to two channels; equivalent to the "balance" control in home stereo systems.

Parametric EQ: equalizer in which all of the parameters of equalization (center frequency, amount of boost or cut) can be adjusted.

PFL: abbreviation for pre-fader listen, which allows a channel signal to be heard and metered without sending it to the main mix

Phantom Power: supplies power to condenser mics through the mixer

Pick up pattern: area that a mic will evenly pick up sound from

Polar pattern: two-dimensional graphic representation of a mic's pick up pattern

Playback: reproduces music over loudspeakers so talent can "lip-sync" or perform to the music track

Roll-off: gradual reduction of sound level below a set of frequency, as in the "75Hz roll-off switch"

Rumble: low frequency noise, often made by the handling of a mic or a/c systems

Solo button: allows only one channel to be heard

Sound: vibrations in air between 20Hz and 20kHz

Sound track: audio recording on video, film, or non-linear edit system

Speaker: device that transduces electrical signals to sound (acoustical energy) by vibrating

Stereophonic sound (stereo): the reproduction of sound, using two independent audio channels, to create a pleasant and natural impression of sound heard from various directions as in natural hearing

Sub mix: mix of audio signals that is treated as one channel or two channels (stereo) in a mix

Talkback: system that allows communication between the control room and the studio floor

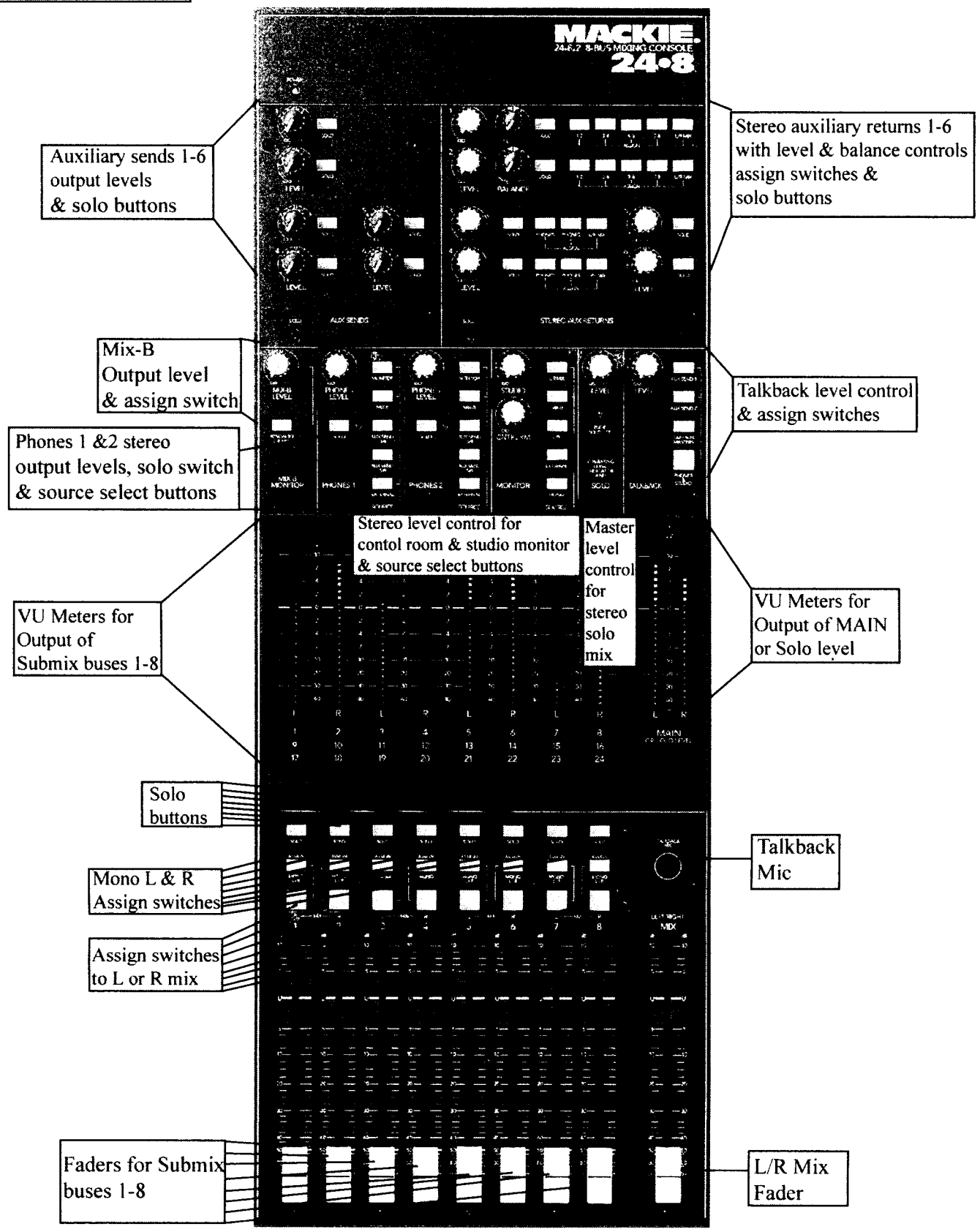
Tone: a fixed signal usually 1Hz

Wavelength: the distance between any point on a sound wave and the equivalent point in the wave's next phase

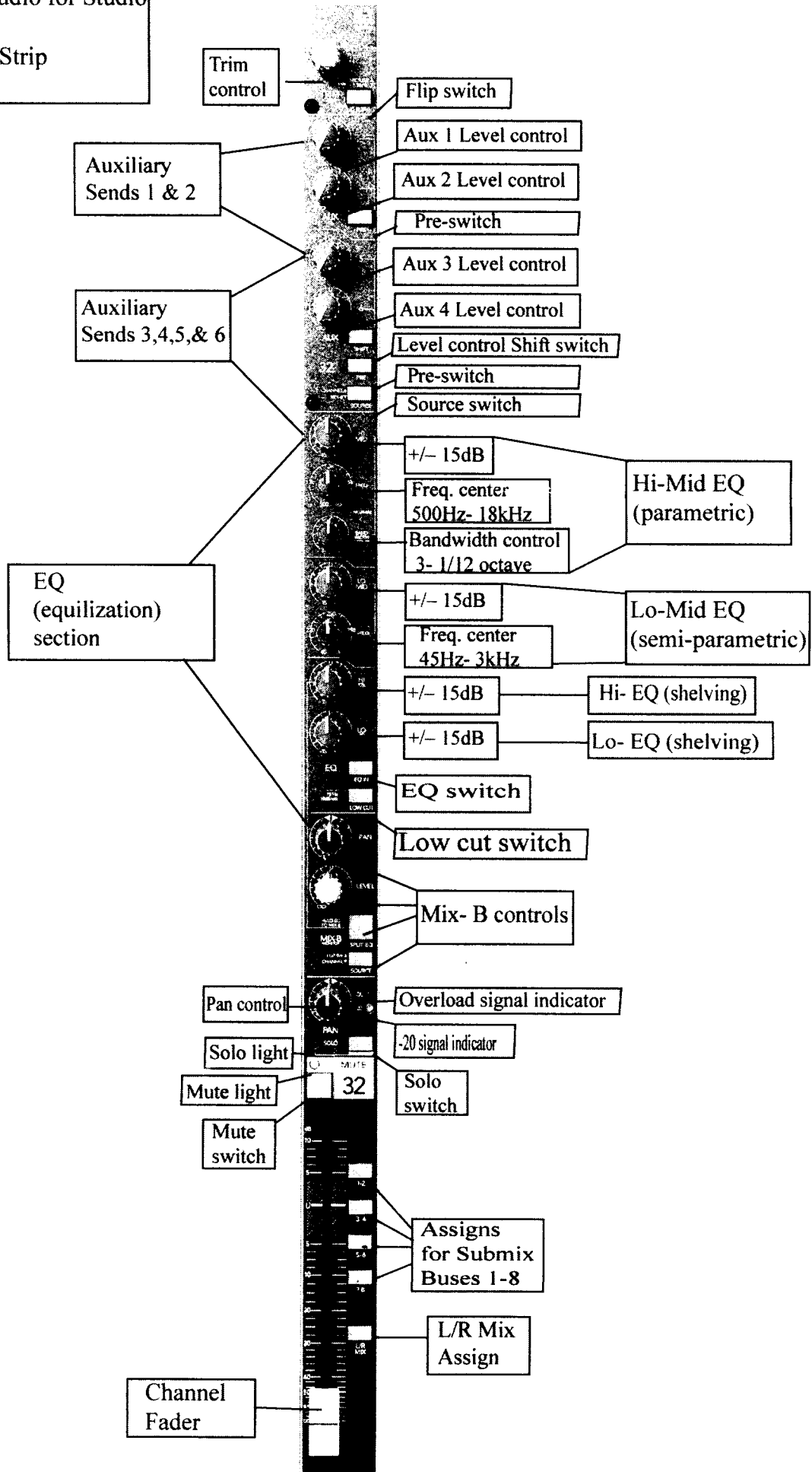
VU: short for Volume Unit, which measure loudness changes

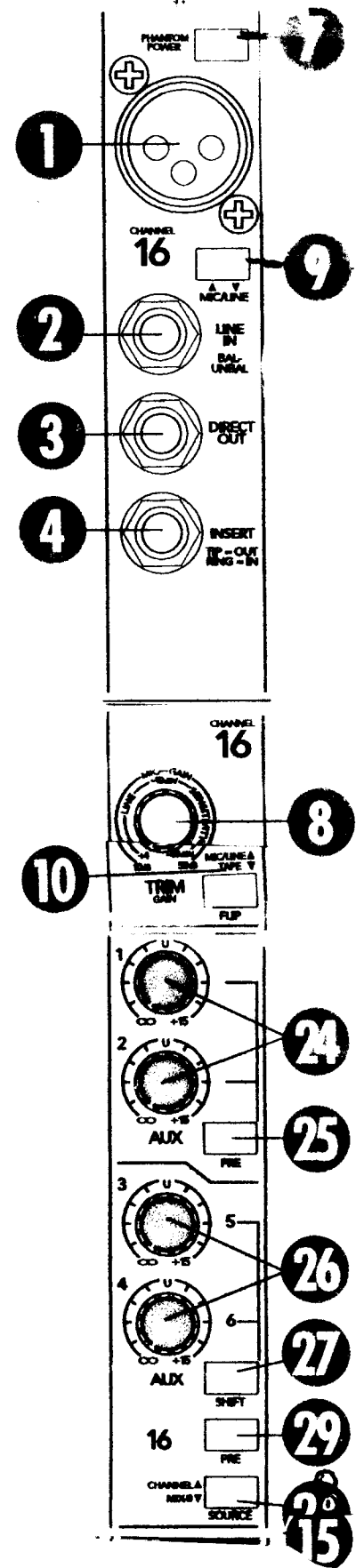
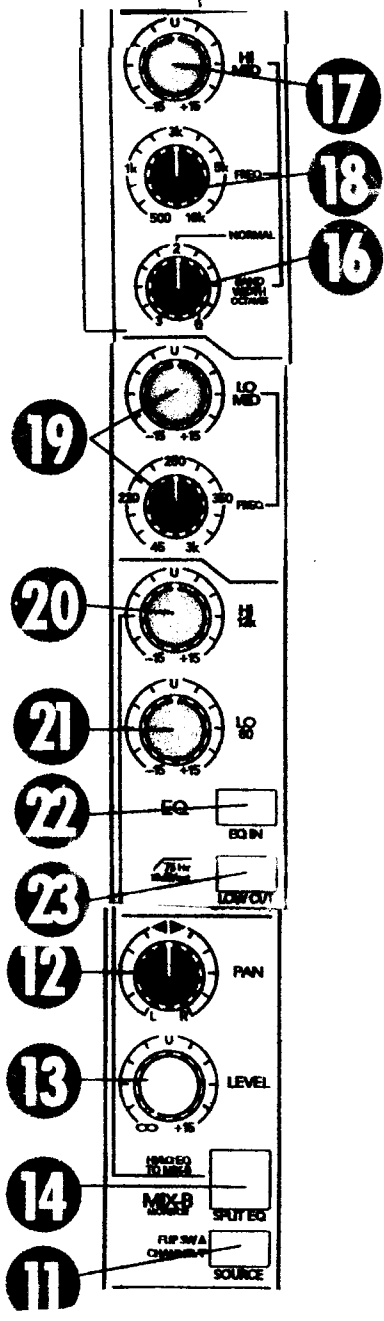
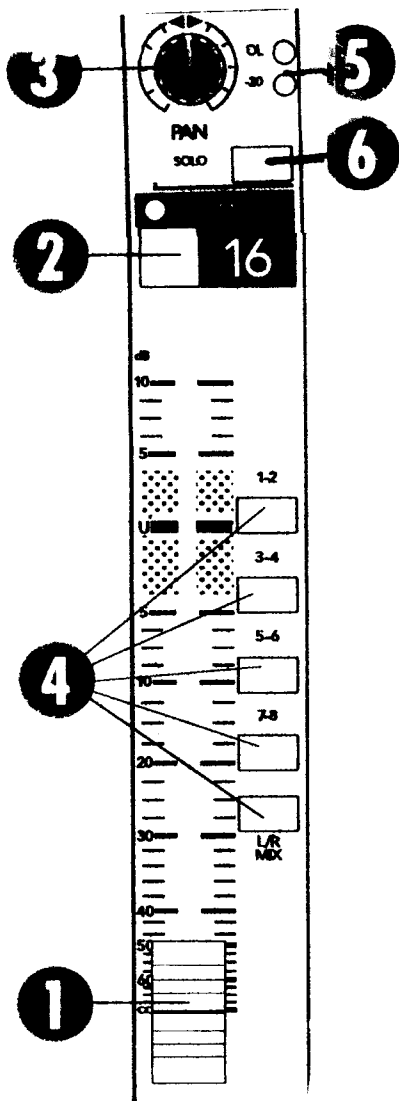
XLR connector: 3-pin grounded connector used in audio.

BCAT Audio for Studio
Mackie
Master Output
Diagram

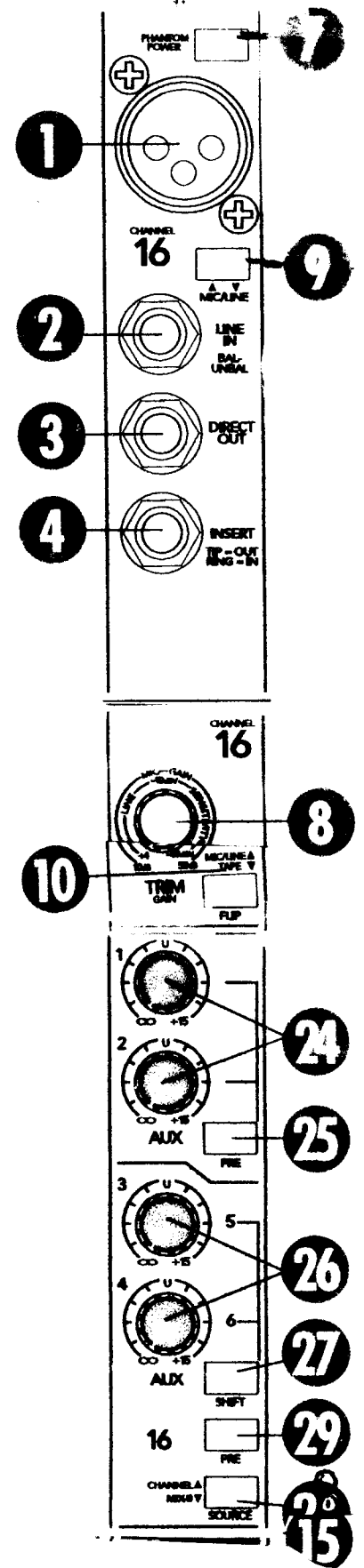
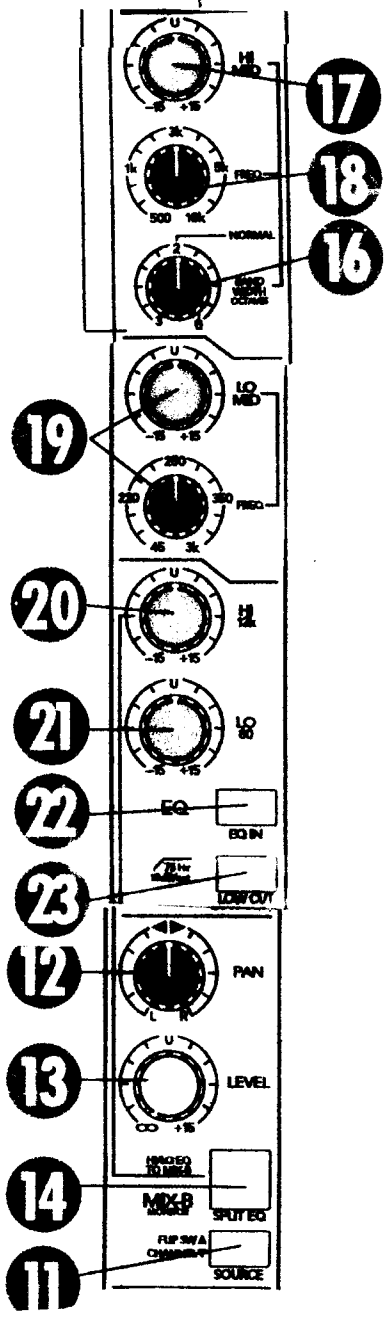
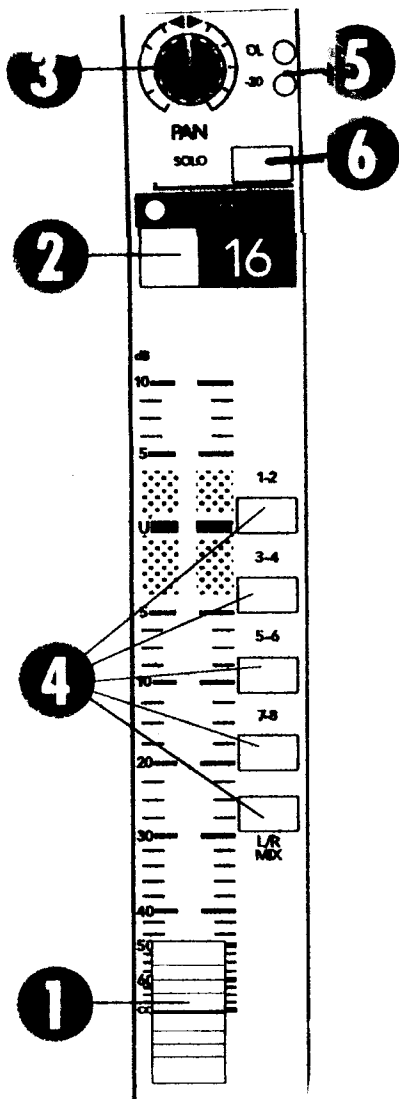


BCAT Audio for Studio
Mackie
Channel Strip
Diagram

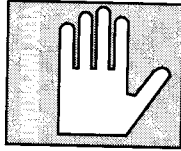




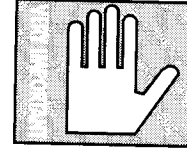
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IMPORTANT SENSITIVITY ADJUSTMENT PROCEDURE!



To fully achieve the Mackie 8•Bus console's impressive headroom and specs, you should "tune" channel sensitivity for each channel.

FOLLOW THIS PROCEDURE FOR EACH CHANNEL IN USE:

1. Assign signal to channel fader:
 - If channel will be used with a microphone, MIC/LINE switch should be *up* & FLIP switch should be *up*.
 - If channel will be used with line input, MIC/LINE switch should be *down* & FLIP switch should be *up*.
 - If channel will be used with a tape input keep the FLIP switch *down*.
2. Set channel strip controls as follows:
 - TRIM pot all the way counterclockwise (+4dB)
 - AUX SEND controls all the way counterclockwise (*off*)
 - EQ switch *up*
 - LOW-CUT switch either *on* or *off* (*on* recommended for mic inputs)
 - Pan hard left or right
 - Channel fader at UNITY
 - SOLO switch *down*
3. Make appropriate "noise" into the channel input. For example, have a performer play/sing/strike something or someone, etc., at the level at which they're going to record or perform. Don't just play a single sustained note, but rather, jam away as you would be during recording or performance. If the channel is being used for a tape input during mixdown, roll an already-recorded track from your recorder.
4. The channel's -20dB LED may light. The L/R main meters will show the actual internal operating level of soloed signals. Now you will optimize levels.
5. For mic or line inputs, adjust the TRIM control clockwise to get peaks that regularly hit 0dB on the L/R meters. For mic inputs this may require full CCW rotation depending on the sensitivity of the mic.
6. If desired (optional):
 - Press the EQ switch *in*.
 - Adjust the channel strip's EQ to about what you will be using during the session.
 - Re-perform Step 5.
7. Return the channel strip's SOLO button to its *up* position.
8. Repeat Steps 1-7 on the next channel that is being used.

1

The channel fader ① is 100 millimeters long, with a precise logarithmic taper and attenuation in dBs printed along the slot for exact and repeatable level adjustments. The fader affects either the mic or line input to the channel (for recording) or the tape return to the channel (for mixing), depending on the position of the FLIP switch.

2

The MUTE switch, located at the top of the channel ②, turns off the primary outputs of the channel: the eight buses, the L & R buses, the channel solo, the direct output and the post-fader AUX sends. Pre-fader aux sends are not muted. With the exception of lighting the mute LED, pushing the MUTE switch is the same as pulling the fader all the way down.

3,4

The PAN control ③, immediately above the fader, pans the channel signal between the two sides of the L/R Mix buses, and also between odd and even pairs of buses 1 through 8.

The actual bus assignment of the PAN control depends on the positions of the five assignment switches located along the length of the fader. With no switches depressed, the PAN control has no effect (well, unless you solo the channel; it still pans the solo).

Pushing the L/R MIX switch ④ assigns the PAN to the main L/R Mix buses. Panning from L to R moves the sound smoothly (with constant loudness) from the left channel to the right channel. Assigning the PAN to a pair of the 8 buses has a similar effect. For example, pushing the 1-2 switch assigns the PAN to buses 1 and 2, and panning L to R will move the sound from bus 1 to bus 2 (from odd to even).

If you want to equally assign a channel to both buses 1 and 2, leave the PAN control at the top, or center, of its travel. If you only want bus 2, turn the PAN control fully clockwise (to the right).

Other comparably-priced consoles provide as little as 50dB attenuation/separation. We use active, buffered circuitry and a custom-taper potentiometer to achieve 87dB attenuation. You get far better channel separation plus freedom from level shifts caused by channel assignment and panning. In addition, our pan pots are *constant loudness*. When you sit between a pair of monitors and pan from side to side, the apparent volume at your ears should stay the same, no matter where the signal is positioned. Our special pan circuitry maintains consistent apparent energy whether the pot is dead center, hard left or hard right.

5

The two LEDs ⑤ next to the PAN control check the channel strip signal level at three important circuit points: at the output of the mic/line preamp, after the EQ and after the channel fader amplifier.

The green LED marked -20 is there to assure you that, yes, something is plugged into the channel (and yes, it does have some output). Most signals more interesting than tape noise will cause the green LEDs to flicker, so they give you a good visual indication of which channels are active. Any peaks higher than -20dBu (@ 1kHz) trigger the indicator. When we say "channel", we mean the signal going through the channel fader... but not the signal going through the MIX B Section. Please refer to the MIX B section of this manual, starting on the next page, for more details.

The red LED, labeled OL for overload, lights when the signal level is high enough to cause clipping at any of the three test points. In normal operation it will almost never light. If it is flashing at you, your level in that channel is much too high. You need to turn something down.

- First try the mic/line trim. If that has no effect,
- Turn down the EQ and/or the insert device, and if that doesn't fix it,
- Turn down the channel. If this doesn't fix it, your input signal is too hot (gasp). Use an external pad to reduce the level (see the sidebar on page 25).

6

The channel SOLO switch ⑥ assigns the output of the channel PAN control to the stereo solo buses and disconnects all other sources from the monitor section. SOLO does not interrupt the eight Submasters, the L/R Mix or the AUX sends, and can be used at any time without affecting the recording process.

SOLO is handy for spot-checking the presence and quality of individual inputs while setting up, recording and mixing. More than one SOLO switch may be pressed at the same time, allowing you to listen to the blend of any combination of channels throughout the console in stereo.

On the Mackie 8•Bus console, the SOLO assignments are stereo except for the AUX sends. SOLO maintains the perspective set up with the PAN controls. When any SOLO button on the console is depressed, its associated SOLO LED will glow steadily, and the RUDE SOLO LITE above the 8•Bus LED meters blinks annoyingly, serving as a reminder with an attitude.

The channel SOLO function is normally post-fader/post-mute, but can be modified for PFL or Pre-Fade (and pre-mute) Listen. See Appendix B: Options, Add-Ons, and Extra Stuff.

Note: All the SOLO buttons on the 8•Bus Series operate in the same way (although they're not all stereo like the channel SOLO). SOLO does not interrupt recording; it only affects the control room monitor.



HIGHLY, MEGA-MONDO-IMPORTANT: SOLO is intended for more than just "soloing." It is THE way to set levels for best noise and headroom. Complete instructions on proper level setting using SOLO are in Section 3: General Information, starting on page 20.

7

CAUTION: After switching PHANTOM Power on or off, wait 1 minute before changing any mic/line switch settings in that 8-channel block.

At the top of every eight channels is a PHANTOM Power switch ⑦. Pressing it sends +48VDC to the eight XLR sockets to the switch's left. For instance, depressing the PHANTOM switch above Channel 8 sends phantom power to the XLRs on channels 1 through 8. **NOTE:** It is always a good idea to check with the Mic manufacturer to verify phantom power requirements.

8

The TRIM control ⑧ sets the gain of the input amplifier for the MIC and LINE inputs. Proper setting of the TRIM control is essential for good noise and headroom performance. Trim pot settings may vary widely depending upon the input level. The output of different keyboards, drum machines, guitar effects boxes, etc., vary from extremely weak to so hot that they can practically be connected directly to speakers. See pages 1, 20, or 25 for advice.

A BIT MORE ON MIX-B/FLIP

In Section 2, we described FLIP's use during tracking and mixdown. Before you actually get involved with recording, we'd like to spring a couple of block diagrams on you that may clarify things further.

The switch labeled FLIP selects the input that is actually fed into the channel fader (and the MIX-B control).

As the label indicates, the MIC/LINE input (after Mic/Line preamp) is fed to the channel fader when the FLIP switch is in the *up* position (Figure 5). When FLIP is *up*, the channel is fed MIC/LINE and MIX-B gets TAPE. That way, you can use MIX-B to monitor the signal as it comes back from the recorder. This is the normal mode for tracking and overdubbing.

In the *down* position, the TAPE return (the output signal from the corresponding track of your recorder) is fed to the channel fader (Figure 6). When FLIP is *down*, the channel input is TAPE and MIX-B receives MIC/LINE. That enables you to use each channel's MIC/LINE input for another input during mix down. Thus *down* is the normal position for *mixing*.

For PA, leave the FLIP switch *up*.

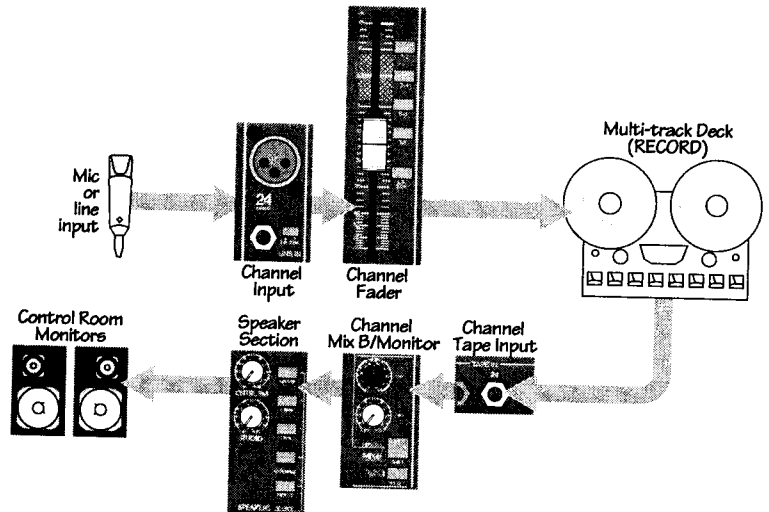


Figure 5: FLIP/MIX-B signal path during tracking ("flip" switch up)

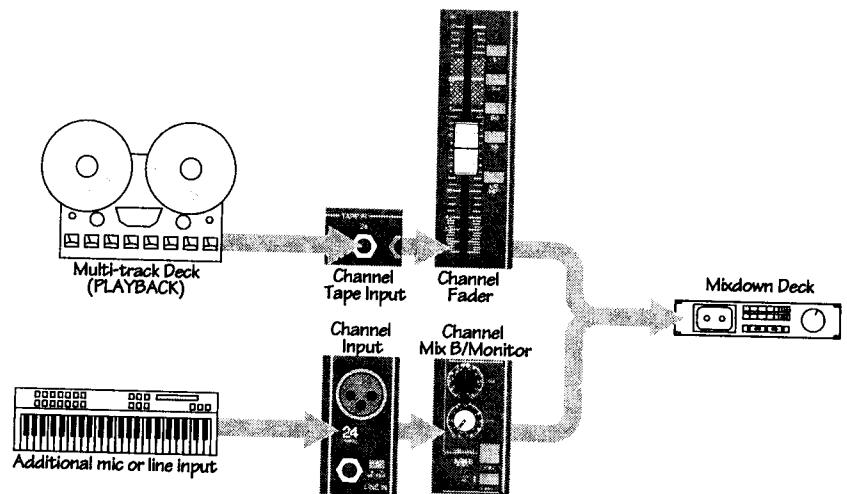


Figure 6: FLIP/MIX-B signal path during mixdown showing additional input ("flip" switch down)

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15 16 17 18

Each 8•Bus channel strip equalizer section has four bands (HI MID, LO MID, LO, and HI) plus a low-cut filter. The equalizer can be split between main channel signal and MIX-B signal.

15 16 17 18

The HI MID EQ section (15) is a true 3-control parametric design, offering: **A)** bandwidth variable from 3 octaves to 1/12 octave (16); **B) ±** 15dB of equalization (17); and; **C)** variable frequency center from 500Hz to 18kHz (18)

We spent a lot of time on this part of the 8•Bus Series' EQ. One of the things we've always noticed about lower-priced consoles was how "drastic" their Hi Mid EQ sounded. Even a little boost could induce honkiness and nasality. Frankly, Hi Mid EQ on many boards is not that useful in studio applications for just this reason.

When you put these consoles on the test bench, you see that the bandwidth of their EQ is simply too narrow — often around one octave. This is OK for extreme adjustments in live situations, but that's about it. It's just not wide enough to gently alter the multiple octaves that voices and instruments span.



This is just the opposite of "classic" big-studio consoles.

Their EQ (including Hi Mid) can be adjusted to be extremely broad — three or more octaves wide in many cases. When you apply this wideband EQ, it sounds far more "sweet" and "natural." Changes sound so gradual and smooth that you can add considerably more EQ than would ever be possible with narrower-band consoles, yet without compromising the overall sound.

On the other hand, ultra-narrow-band EQ also has its place, both as a corrective tool and for special effects. So why not give you even tighter control than would be possible with a 1/3-octave graphic equalizer? Thus was born the 8•Bus' 1/12-octave to 3-octave width range.

At this point, as we played around with the Bandwidth and Boost/Cut controls using actual music, we realized that to limit this circuit to *just* Hi Mid (typically above 2.5kHz) would actually limit its usefulness. The flexibility that true parametric control provides is nice to have in the octaves below the Hi Mid region as well. So we came up with what is probably the first console Hi Mid control that can be swept all the way down to 500Hz. If Hi Mid doesn't do it for you as a name, think of these three controls as the Roving-Parametric-Problem-Solver-O-Matic-O-Rama.

Now you know some of the design philosophy behind our Hi Mid EQ. It's time to start experimenting with it yourself. Don't forget to try the control at its bandwidth extremes — especially at the 3-octave end. You'll be pleasantly surprised.

Conversely, if you hate technology and yearn for the sound of that old board you sold to buy an 8•Bus, you can achieve a semi-parametric sound by leaving the BANDWIDTH knob in the middle at 2 octaves.

19

The LO MID EQ control (19) is a semi-parametric (sweepable) equalizer with a broad, fixed bandwidth of 2 octaves, ±15dB of equalization and frequency variable from 45Hz to 3kHz.

Boosting in this range can put warmth and body into vocals and instruments. Cutting can really help the clarity of some sounds by reducing boxy and boomy tones.

20

The HI EQ control (20) is a fixed 12kHz shelving equalizer with ±15dB of equalization available. A great treble control, it is switched with LO EQ into the MIX-B circuit if the SPLIT button is depressed.

Shelving equalizers work on a very broad range of frequencies, and consequently, are very musical. In a 12kHz shelf like this section, that means that all the upper harmonics of a sound are raised evenly, basically keeping their original musical relationship to each other. A high-frequency shelving EQ is great for putting shimmer into acoustic guitar and piano tracks and sizzle into vocals.

20

The LO EQ control (20) is a fixed 80Hz shelving equalizer with ±15dB of equalization available. It's a fine bass control, and is switched with HI EQ into the MIX-B circuit if the SPLIT button is depressed.

A low-frequency shelving equalizer will add or remove bass in a smooth, musical fashion. Good for working on bass drum and bass guitar, fattening up (or thinning out) a piano or contouring an entire mix.

NO
21

22

The IN switch (22) bypasses the EQ (though not the LO CUT filter) when *up*. Depress it to enable equalization.

23

The LO CUT switch (23) inserts an 18dB/octave low-cut (high-pass) filter with a -3dB point of 75Hz into the main channel signal. The LO CUT filter is unaffected by the EQ IN switch.

A low-cut filter is handy to get rid of room rumble, traffic noise, wind noise, popping, and other unwanted very-low-frequency sounds. It can also be combined with LO EQ boosts to produce some interesting bass curves as shown in the last EQ graph on this page. *It is highly recommended that this switch be engaged for vocal microphones, especially in a live P.A. situation.*

AUX SENDS 24 25 26

Auxiliary sends are generally used to provide mixes for headphone cueing and for effects sends, and in Sound Reinforcement use, can be assigned monitor submix duties. The 8•Bus Series consoles provide 6 mono auxiliary sends with several routing options.

Note: All of the 8•Bus Series AUX sends have a very wide range of gain (just like the MIX-B Level control discussed earlier). The first half of the control's rotation reaches from the *off* position to unity gain (U). This half of the control's range corresponds to the full range of a conventional mixer. The second half of the control's rotation provides you with even more gain, from unity to +15dB.

For example, when you want a super "wet" sound (mostly reverb), the extra gain allows you to bring the channel fader part way *down* (and the AUX send way up) so the sound is mostly reverb with just a touch of "dry" signal.

Don't worry about having mono effects sends: almost all effects units, mono or stereo, have mono inputs. Even if there are two jacks labeled Left and Right, they are almost always combined into mono internally. In the rare cases when an effect actually has true stereo inputs, it is often more convenient to ignore that and treat it as mono anyway. If you really need to send in stereo, just use two AUX sends from the console. Route AUX 1, for example, to the left input and AUX 2 to the right. Then adjust the stereo perspective by favoring one or the other send. For instance, if you have a stereo synth into channels 23 and 24, use AUX 1 on channel 23 and AUX 2 on channel 24.

Note: In PRE mode, all AUX SENDS are connected pre-fader but post-EQ. This is usually the most useful setup, but if you prefer, all AUX SENDS can be modified to be pre-EQ, pre-fader. See Appendix C: Modifications on page 52.

24

Auxiliary sends 1 & 2 (24) have two independent level controls and share a common PRE/POST switch.

When in the normal post mode, sends 1 & 2 get their signal after the channel fader and the MUTE switch. This is the usual configuration for an effects send, since the level of the effect should follow the volume adjustments made by the channel fader.

25

With the PRE switch (25) depressed, AUX SENDS 1 & 2 receive their input from a point before the channel fader and MUTE switch, and are not affected by changes in these controls. This is the normal switch setting for creating stage monitor and/or studio cue headphone sends.

AUX SENDS 3 4 5 6 26

Auxiliary sends 3, 4, 5, and 6 (26) are very similar to sends 1 & 2. The differences are in the SHIFT and SOURCE switches. Read on.

SHIFT 27

The SHIFT button (27) connects the two level



controls, as a pair, to either sends 3 & 4 (in the *up* position) or to sends 5 & 6 (in the *down* position). Twice the sends in half the space.

28

The SOURCE switch (28) breaks AUX Sends 3-6 away from the main channel strip world that sends 1 & 2 are stuck in, and allows the sends to tap from another source: the MIX-B signal for that channel. This is a very handy thing, whatever you happen to be using MIX-B for.

If MIX-B is an additional input for your mix-down, then (when the SOURCE switch is depressed) AUX Sends 3-6 are effects sends connected to that source. If MIX-B is your studio monitor submix from the recorder, then AUX Sends 3-6 are a great source of headphone cue signals. They will follow playback, record and punch-ins automatically as your recorder does its stuff.

29

The PRE switch (29) for Sends 3-4-5-6 functions just like the PRE switch sends 1 & 2, when the AUX SOURCE switch is in the CHANNEL (*up*) position.

Additionally, the PRE switch will tap the pre-level (and pre-mute) signal of whichever SOURCE you have selected-channel or MIX B.

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