

MACKIE® Church Sound NOTEBOOK™

Summer
1997

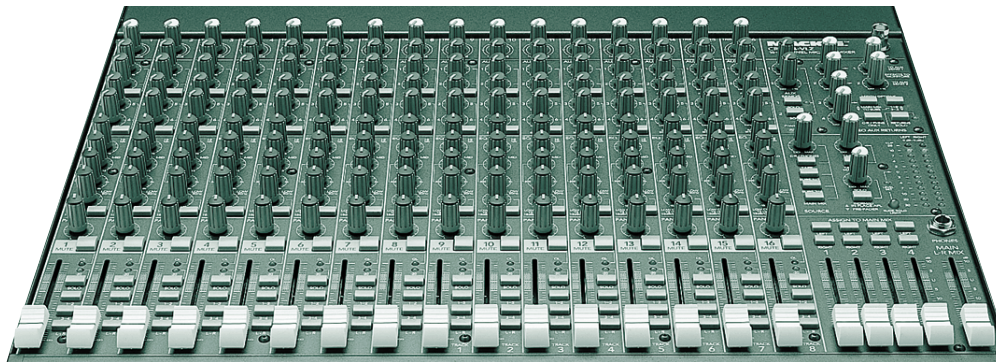


Cure yourself of “knob-phobia”

To those who have not spent their lives attached at the finger tips to a mixing console, the sight of all those knobs and buttons can be daunting. Don't be intimidated — nuclear physics or brain surgery can seem a little complicated too, at first glance. But armed with an understanding of a few basic principles, and with a little practice... well, getting around on the mix-

prevent the addition of noticeable noise and avoid overload distortion.

Aux Sends (Monitor or Effects). Think of the aux (auxiliary) sends as level controls for additional mixes of your audio sources. If you mix on a Mackie SR24•4, for example, you can perform six auxiliary mixes in addition to your main mix. These aux sends may be routed to an outboard effect (such as re-



Gobs of knobs on the CR1604-VLZ®. Don't let 'em scare you!

ing console need not be a problem.

Getting over being intimidated by all those knobs and buttons is fairly easy.

Divide controls into functional sections.

The first step is to realize that the console can usually be divided into just a few basic sections. Most mixer controls can be identified as belonging to either an *input* section — known at Mackie as “channel strips” — or a *master* section.

Input Section (Channel Strip) Controls

A channel strip is generally comprised of the following features:

Trim. Here you adjust the level of the incoming signal to that which is optimum for mixing. By proper trim adjustment you can

verb, delay, etc.), and returned to be added to the main mix, or they may be used to perform a monitor, recording, assistive listening, or broadcast mix.

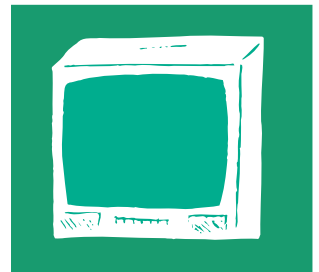
EQ. Ahhh, the great equalizer. EQ comes in many forms, but basically is a way of increasing or decreasing the amplitude of a certain frequency range within a signal. The bass and treble knobs on your home stereo are essentially EQs. Some mixers, including most Mackie models, have Low Cut Filters that cut all signal below a certain frequency. These tend to be used on channels with mics because they can help eliminate mic thumps, wind noise, etc.

Signal Routing. These controls include: *Pan*, used for placing a source to the left, center, or

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MACKIE DESIGNS' QUARTERLY GUIDE TO “BLESSED MIXINGS”

The “ins” and “outs” of Inserts

One of the features often found on the rear panel of a mixing console is the Channel Insert. The insert serves simultaneously as both an input and an output for either a single channel or for some other signal path, such as a submix or main output bus. It is a point in the signal path at which the signal can be detoured —

- **Use it as a direct out** (post mic preamp, but pre low cut filter, mute, EQ, fader, etc.). Just because you’re sending something out doesn’t mean you have to bring it back. You can use each insert to send a “direct out” signal to a line-level input of a tape recorder, or to another mixer for a broadcast or recording feed. At the

- **Send a signal through a “Y” — using the insert as both a direct out and an effects loop.** As an alternate approach, create your effects loop as described earlier, then insert a “Y” adapter *after* the processor to affect (compress, for example) both the direct out and the individual channel in the mix. A good application for this

might be to compress a pastor’s lav mic or a pulpit mic, in both the house mix and a recording or broadcast.

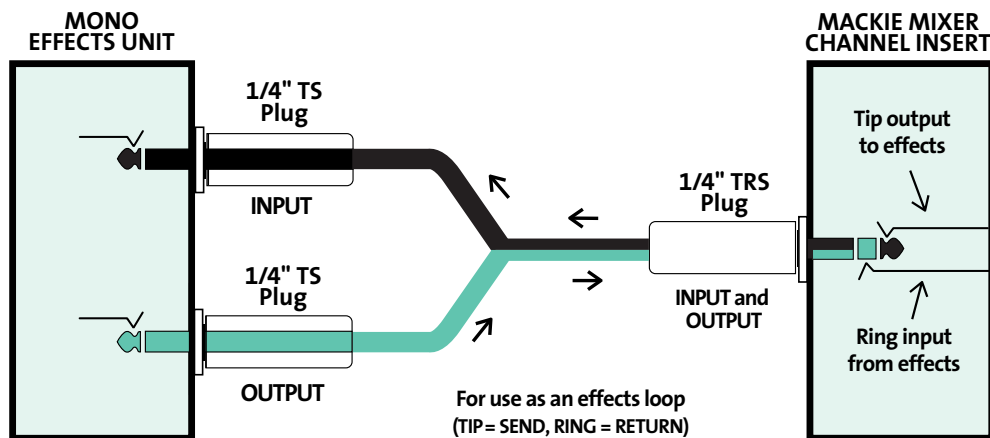
Whether you use them as a part of your normal setup every week, or just to solve an occasional routing problem, inserts add tremendously to the versatility of your mixing console.

¹ As an alternative to inserting a two-conductor plug half-way, you may choose to make a custom cable which can be inserted all the way, but will still maintain signal flow in the mixer channel.

To do this, use a three-conductor 1/4” TRS (or tip/ring/sleeve) phone plug instead of a two-conductor TS plug. Wire the TRS plug so that the tip and ring are both wired to the center (positive) conductor of your shielded audio cable.

NOTE: Be sure to mark this cable in some obvious way to show it is for this special purpose only. It cannot be used with a balanced TRS jack without shorting the signal.

Above: Standard “Y” cable with TRS to 2 TS. **Below:** Two “Y” cables hooked together to allow effects to be fed to two separate places. Note that one “Y” is a TRS to 2 TS, while the other has 3 TS connectors.



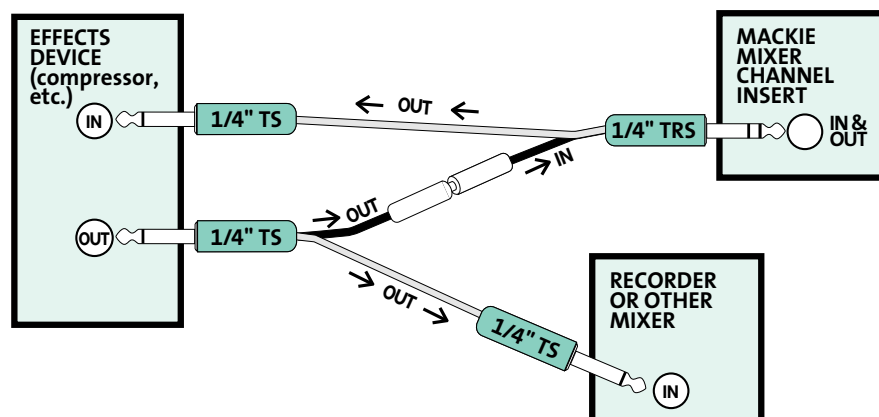
sent out of the mixer and then returned to its normally scheduled programming, creating what is called an effects loop. In other words, it allows you to “insert” an outboard device into the signal path. (Like a 10 HP Evinrude. Or, more likely, a reverb.)

On the Mackie SR40•8 large-format mixing console, each insert point includes a separate output and input jack. On most mixers, however, a single 1/4” three-conductor jack provides connections for both an input and an output.

What would you do with such a strange jack?

- **Apply effects to a channel or submix.** Because an insert is both an input and an output, you can route the signal from the channel out to a reverb, compressor, limiter, etc., and then back into the channel. You might send the signal to a noise gate unit in order to automatically “turn off” a mic when it’s not in use. Reducing the number of mics that are on, or “open,” reduces the risk of feedback and improves your signal-to-noise ratio.

mixer end of your direct out cable, you’ll want a standard 1/4” mono (or TS, “tip/sleeve”) phone plug. Push the phone plug *part way* into the insert jack, just to the *first click*. This will route the direct out signal via the cable, without interrupting the signal flow in the mixer. If you insert the plug all the way to the second click, you will still get a direct out signal, but the signal in the channel will be interrupted at that point — removed from the mix.¹



The Tale of the Aux Return

No, this is not the story of the return of the prodigal ox, this is the story of the Aux Return. An aux (auxiliary) return is a line-level input that differs from the line inputs on each of the channel strips in a couple of ways. First, the aux return normally lacks a lot of the control features — things like trim and EQ, for example. Second, aux returns are commonly routed directly to the main output bus, where they are mixed with individual channel signals. Thus the extent of control for an aux return is most often a rotary level control. Let's look briefly at a couple of applications for the aux returns.

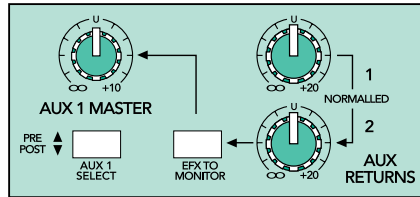
The principle use of the aux return is to bring a mix from an effects send (aka *aux send*) back into the console after it has been enhanced (or abused) by an outboard signal processor, such as a reverb system. The aux return's level control provides the means of adjusting how much of that processed ("wet") signal will be mixed together with the "dry" (or unprocessed) original signal.

OK, let's say you're doing a sound reinforcement mix of a couple of vocalists on a Mackie MS1202-VLZ® or MS1402-VLZ®. You're using Aux Send 1 to provide a monitor mix for the singers while creating an effects mix on Aux Send 2. Fairly simple so far. But now let's say the singers discover how nice the reverb sounds in the main mix and decide they could go for some of that in their monitors as well. What's a sound technician to do? Fortunately, there's an easy solution. Remember, however, you should never let on just how simple such a feat might be; you'll only spoil the musicians and they'll keep on wanting more.

Adding reverb (or other effects)

Here's how you can add reverb to both the house and monitor mixes, with level control over the reverb content of each.

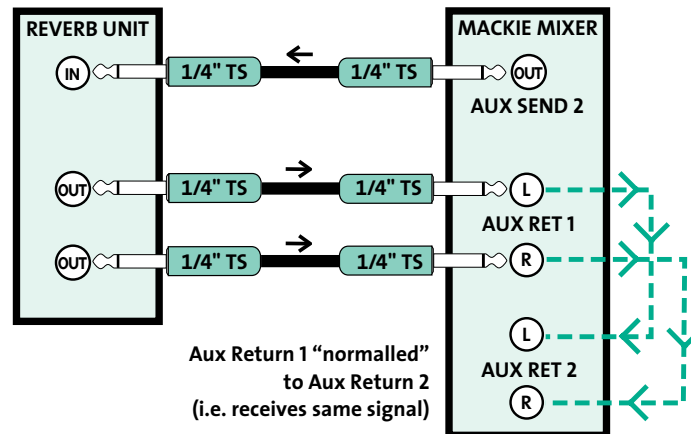
1. Patch the Aux Send 2 output (effects send) into the input of your reverb.
2. Bring the reverb output (typically two-channel) back into Aux Return 1, Left and Right. Do not plug anything into Aux Return 2.



The "aux" controls on the MS1402-VLZ® provide great versatility.

3. Depress the button marked "EFX To Monitor," adjacent to the Aux Return 2 level control. This routes the signal at Aux Return 2 into the monitor mix, and into the main mix.

Because Aux Return 1 is *normalled*¹ to Aux Return 2, the reverb signal coming into Aux Return 1 will also be



present at Aux Return 2. You can now easily control the amount of reverb content in your main mix with the Aux Return 1 level control, and the amount of reverb in the monitor mix with the Aux Return 2 level control.

Aux return as extra set of inputs

Aux returns can serve another valuable function. One of the more obvious considerations in selecting a mixer is the required number of inputs. People seldom buy a mixing console with the belief that it will not

handle all the audio sources in their system. Still, it is hard to find a church sound technician that hasn't occasionally wished for more inputs.

The aux returns on your mixer might be just what you need at some point to easily add another stereo or mono line-level source to your main mix. Here are some examples of sources you might like to route through the aux returns:

- **Tape deck or CD player**
- **An ancillary mixer** (submix of keyboards, other instruments, praise team, choir, etc.)
- **Audio tracks from a VCR**
- **Line-level output of a wireless mic receiver**

If you're bringing a *mono* line-level source into an aux return of your Mackie mixer, note that the Left Aux Return 1 input is also marked MONO. This means that the left channel is normalled to the right channel. So

plugging your mono source into the Left input will give you the same results as plugging it into both the Left and Right. The ability to use the left channel of a stereo input for routing a mono signal to both channels is common to all Mackie mixers.

So now you know

Aux returns are more than just 1/4" holes and finely sculptured knobs. In fact, they may be just

the topic you need to enliven the conversation at your next potluck dinner. It's almost guaranteed to provide you with job security as the church sound operator.

¹ "Normalled" means that, under "normal" conditions, spring-loaded contacts interconnect these jacks so that the signal coming into Aux Return 1 is shared with Aux Return 2. When you insert phone plugs into the Aux Return 2 jacks, these connections are broken, and Aux Returns 1 and 2 become independent.

Physics: Simple, phun, usephul phacts

Throughout history, man has written “laws” in an attempt to explain those forces within creation that we are sort of, well, stuck with. Of particular interest to those of us who mix audio should be certain laws of physics having to do with the behavior of sound. We shouldn’t have to point out that the laws of physics are a little different than some of those other laws we encounter every day. You can choose to honor or violate a dog-leash law, for example. Violation may result in no more than a ticket and a lawn to clean. The laws of physics, on the other hand, can only be broken in Saturday morning cartoons and in the sales literature of some lumatic-fringe hi-fi equipment manufacturers.

Inverse Square Law

One of the greatest tools in your sound system operation arsenal is a rule which describes a behavioral characteristic of sound. Known as “Inverse Square Law,” this tool costs nothing, takes up zero rack space, and requires no regular maintenance. But understanding Inverse Square Law can help you make intelligent microphone placement decisions which can lead to reduced risk of feedback and dramatically improved recordings.

Inverse Square Law refers to the way sound levels decrease as you move away from their point of origin.¹ It should come as no major revelation that the sound of a person speaking to you from two feet away will be louder than what you hear from a distance of six feet. But how much louder? How great a change takes place?

Fortunately, there is a term to describe the amount of change between two such sound levels, or to compare their strength. While their actual amplitude is measured in dynes/cm, a more practical way of discussing sound is to refer to *sound pressure level* or SPL. The unit used to compare these levels is the *decibel* (abbreviated dB). A decibel is one tenth of a Bel.² When we hear a

sound that we perceive to be twice the level of another, that difference is about 10dB.

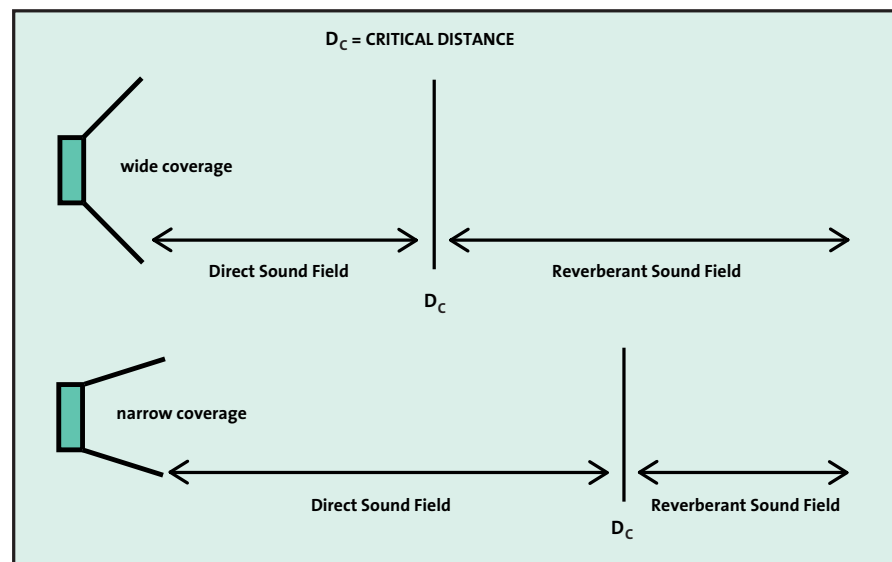
When we say that the SPL from a speaker system is 80dB at the third row back in the auditorium, we are also comparing two levels. 80dB SPL (or any SPL) is referenced or compared to 0dB SPL, which is the theoretical “threshold of hearing” – the softest sound level that can be heard by sensitive, young ears not yet subjected to rock and roll. Note that 0dB does not indicate the absence of sound, just that there is zero difference from the reference level.

Back to Inverse Square Law. According to the law, the intensity of sound varies inversely according to the

more directional, critical distance is moved farther out.

Beyond critical distance, the clarity of speech becomes adversely affected – with intelligibility severely limited for the listener seated too far back in the reverberant field. One way of dealing with this situation is to locate the system speaker or speakers closer to the listeners. Care should be taken in using speakers in multiple locations within the room, though; adding speakers can actually *decrease* intelligibility by presenting the listener with multiple sound arrival times.

No doubt you have been in a highly reverberant space, such as a gymnasium, in which it was difficult to converse over a distance of only a few



square of the distance. Simply stated, measured sound level will drop 6dB for each doubling of the distance from the source.

Indoors, this drop in level from the sound source will occur only to a point referred to as *Critical Distance* (D_c), at which the level becomes equal to that of the reflected sound from walls, ceiling and floor. This distance varies with the acoustics of the room and the directional characteristics of the sound source. The more reverberant or “live” the room acoustics, the shorter will be the path to critical distance. As the sound source is made

yards. You probably noticed that you could be understood better if you would cup your hands to your mouth, making your voice more directional by creating a small horn. Critical distance can be increased by using highly directional loudspeakers, adding absorptive material to room surfaces, or by removing the walls and ceiling. (The latter being a last resort.)

Using the law to annihilate feedback

Let’s look at how a knowledge of the inverse square law can help you

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eliminate feedback. To do that, we need to know what conditions bring about feedback.

Feedback occurs when sound from a system's loudspeaker re-enters the microphone at a level that is equal to or greater than the level of the sound arriving from the original sound source.

So, if we can speak or sing louder into a microphone than the level of our amplified voice arriving back at the mic, we win — right?

Let's say we have a soft-spoken missionary standing two feet from the microphone. Her voice produces an average SPL of 59dB at the mic, and our sound technician knows he must amplify that to 75dB SPL at his mixing position in order for everyone in the auditorium to hear.

We'll further assume that once this level has been achieved, the missionary's amplified voice arrives back at the mic at 60dB SPL. Our sound technician will try in vain to reinforce her voice without feedback. The technician inches the fader gently upward, the system begins to howl, and the missionary (who last saw a sound system while on furlow in 1951) reacts by backing away from the mic, making matters worse. Now there are only two choices for the technician:³ throw the system into terminal oscillation and run out the door, hands over ears, screaming "I can't take it anymore!" or back off on the fader and admit defeat. (If you're grading the technician, choice B is the correct one.)

What are the possible solutions? First we can move the mic or the missionary to bring them closer together. If we divide the distance in half, to one foot, the level of the unamplified voice at the microphone will be 65dB SPL — a 6dB increase. Of course this will increase the system output level by 6dB as well. But when the operator decreases the channel gain to the desired level, the SPL from the loudspeaker at the mic will once again be 60dB, or 5dB below the level of the voice. If the missionary can be urged to step even closer to the mic, say to a distance of six inches, the mixer gain may again

be lowered by 6dB. Now the voice will be 11dB above the level of the amplified sound at the microphone. It is generally considered good practice to *always operate your system at least 6dB below the feedback point.*

Feedback is a signal-to-noise ratio problem. The greater the ratio between the levels of the desired signal (in this case, the "amplified version" of the sound) and the unwanted signal, the better. By simply decreasing the distance between the sound source and microphone, you can dramatically improve your signal-to-noise ratio. This same principle, of course, applies to controlling the pick-up of *any* ambient noise. If you are recording or broadcasting using a mic that is not connected to a speaker system in the same room, feedback is not a problem. But you'll diminish the sounds of coughs, sneezes, babies, reverberation, and snoring in your recording if you move the mic closer to the sound source.

Other tools have you, which are not of this article.

Keep in mind that there are other tools which can help you eliminate feedback. Some of these include: a linear microphone and loudspeaker frequency response, reduced system bandwidth, directional microphones and loudspeakers, and maintaining a minimal number of open microphones. We'll be addressing each of these subjects in MCSN, as space permits.

The intelligent use of Inverse Square Law is often the tool you need to eliminate feedback or improve a recording. And, hey... you can't beat the price.

FOOTNOTES

¹ We'll try to keep this simple for the sake of discussion, so we must warn the acousticians, physicists, and child prodigies that we're taking the grenade approach to this topic — we'll get close enough to be very useful, while not losing most of our readers with confusing details. Inverse Square Law actually assumes that the sound source radiates in all directions equally and that there are no reflective surfaces for the sound to encounter.

² The Bel is named for Alexander Graham Bell. Not as well known is that Bell discovered sound as he sat with his friend Isaac beneath an apple tree. A passing breeze loosened an apple which dropped onto Sir Isaac Newton's head. Klunk! Sir Isaac was struck with the concept of gravity, which he failed to patent, and Alex, sensing the gravity of the situation, realized the percussive potential of the Klunk. He soon invented the bell, an early percussion instrument in the church, and went on to start the audio industry. (Early church bells were called "klunks," which was German for "bonk." Refinements to the clarity of their sound led someone to note that the klunk now sounded "clear as a bell." After that, it just seemed natural to refer to them as bells. Besides, the name had a nice ring to it. There is no connection between this old saying and Alexander's old spinster cousin Clarissa Bell.

Of course the lyrics of numerous songs have had to change over the years. Some of these included: "Thud the Klunks, Thud the Klunks, Let the Whole World Know," "I Heard the Klunks on Christmas Day," and "Thud the Klunks of Heaven, There is Joy Today."

³ The suggestion has been made that the enterprising sound tech might find a third option. Noting the distress of the congregation, as evidenced by the glowering faces rotated in the direction of the control booth, the imaginative tech might use this opportunity to take up an offering for new sound equipment on the pretense that it alone can eliminate the feedback. 🗑️

TRANSIENT RESPONSE *

Wow! Looks like the cockpit of a 747!

* The unsolicited comments of congregation members as they pass by the mixer.



Cure yourself of “knob-phobia”

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right in the stereo horizon; *Channel Assignment switches*, which allow you to assign a channel to specific submixes; *Mute*, which “turns off” the channel; and *Solo* (or Cue or Monitor), which allows you to monitor the channel on headphones or control room monitors, without affecting the mix.

Fader (level control). The fader is the slide control that allows you to adjust the amount of signal that appears in the main mix. While the trim control is used to adjust incoming signal levels to be nearly the same — to establish a “starting point” for the mix — the fader allows you to adjust the relative levels of all channels being mixed. (If your mixer has a rotary, rather than slide control, it may be referred to as a “gain” or simply “level” control.)

Other fun things: Some channel strips include a *Polarity switch*, used to reverse the polarity of the incoming signal. Each channel may also have its own phantom power switch, which allows phantom power to be used only with those condenser mics that need it.

Learn one channel strip and ya got 'em all

A mixer may have 12, 14, 24 or even 56 channel strips. That's a lotta knobs. But keep this in mind: most channel strips are exactly the same. Aside from stereo channel/aux return strips, which tend to have a few less knobs, and aside from some very exotic consoles, once you know one channel strip you know them all. You're halfway there already!

Master Section

Submix Sends. You can usually assign channels to specific submixes. All of your backing vocals may be sent to one submix, and the rhythm section may be sent to another. This is sometimes called “group” mixing, as you can control the relative levels of groups of microphones with just one fader per group.

EQ. Some mixers have a broadband graphic EQ section in the master area — these are essentially system tone controls. This is most commonly seen in

combination mixer/amplifiers, where the advantages of being able to select a particular external equalizer may be offset by the demand for portability.

Aux Returns. Used to control the amount of signal being returned to the mixer from an outboard signal processor or other source. See The Tale of the Aux Return, page 3.

Monitoring. Look in the master section for selector switches with which you determine what you hear in your headphones or studio monitor speakers, and level controls that allow you to adjust their volume.

Master Fader(s). Just as the name implies, this is your master level control, normally used to adjust the “main mix” Left and Right signals that go to the amps and on to your PA speakers, or to a 2-track recorder such as a DAT or cassette deck.

Practice and listen.

One of the best ways to become comfortable with the console is to spend a little time with it in private. Just the two of you. It need not be an expensive night out; it's quality time that counts.

The purpose of this exercise is to learn cause and effect at some more appropri-

A channel strip of Mackie's CR1604-VLZ®.



ate time than during a worship service. Here's a good way to begin. First, you'll need a sound source, or a selection of sound sources. There are two types of sources you could use: one is an acoustic source picked up by a microphone; the other is a direct source, such as tape or CD. We suggest you begin by using a tape or CD player, connected to a line input of your mixer.

You'll also need decent headphones or some good quality studio monitors connected to the control room or main mixer outputs. By monitoring the console output as directly as possible, you will be able to hear qualitative changes as you make them — including subtle differences that would otherwise be obscured by room acoustics or ambient noise.

The idea is to send your program source through an input channel, play with the various channel controls (EQ, pan, etc.), and monitor the results. You may want to connect your source to a stereo channel to do some of your tests and then connect to a mono input to discover the differences. The stereo and mono inputs may differ, for example, in the EQ control available. Also, listen to how the pan (or balance) control differs on a stereo channel from the pan on a mono channel.

The area you'll want to spend the most time with is the EQ. Listen to a variety of program material to determine how your mixer's channel EQ affects the sound of the signal. Some of the best recordings to use include: male speaking voice and solo recordings of piano, violin and guitar. The higher the recording quality the better, but almost any recording will give you some idea of what your EQ will do.

Using recorded sound as your source, rather than a miked voice or instrument, provides exact repeatability for comparison of different control settings. It also allows you to hear only what the mixer settings are doing, without the confusion of hearing di-

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rect, unamplified sound, as you would expect when mixing in an auditorium.

Focus on the controls you're actually using, or using the most.

One factor that should make the sea of knobs in front of you a little less imposing is the fact that during a typical church service, relatively few controls need to be touched. Many of the controls on your console are used to pre-adjust levels or determine the destinations to which signals will be routed. It is not unusual for an operator to use only the mute buttons and faders during a service. Trim controls should be set in advance and need to be adjusted only if conditions change dramatically. Likewise, channel assignments to sub sends or main outputs are normally set and left alone. Once channel EQ adjustments have been made for particular microphone applications, it is quite common for there to be no need to adjust EQ further during the service.

The tip of the iceberg.

Of course, there's a lot more to mixing sound than we can get to in this article. That's what *MCSN* is for. Throughout this issue and future editions, we'll further explore what it takes to provide the best sound possible for your congregation. 

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Feedback

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(relative to an omni) if the mic were placed in a diffuse sound field. This is referred to as the Distance Factor, or DF. For example, let's say we have an omni positioned 10" from our pastor's mouth, and that the sound we do *not* want to pick up (sound from the

POLAR PATTERN	REE	DISTANCE FACTOR (DF)	NULL ANGLE (DEGREES)
Omni	1.0	1.0	N/A
Cardioid	.333	1.7	180
Supercardioid	.268	1.9	126
Hypercardioid	.25	2.0	110
Bi-directional	.333	1.7	90

loudspeaker) is arriving from all directions equally. Assuming that the pastor's voice is louder at the mic than is the sound from the speaker system, the difference between the two is the *signal-to-noise ratio*. Of course if the loudspeaker wins, we have a noise-to-signal ratio — we also have feedback.

Omnidirectional microphones are given an REE of 1.0 as a reference point for comparing other polar patterns. The cardioid's REE, 0.333, indicates that it picks up just 1/3 the energy that the omni picks up within a diffuse field. Because this reduced sensitivity is at the sides and the rear of the cardioid, it is ambient noise, not signal, that is lowered in level. The result is that the cardioid can achieve the same signal-to-ambient noise ratio as the omni at a distance 1.7 times that of the omni, or 17" in our example.


While the REE of a mic is good to know, it should be understood that the figures on the chart are for *mathematically perfect* microphones (*micus nonexistentus*) in a *totally diffuse field* (*environmentus acousticus nonexistentus*). While they are a good basis for understanding, your mileage may vary.

Proximity Effect

Directional microphones also differ from omnis in another respect. As a directional mic is brought close to a

small sound source (a human mouth, for example) the mic's response becomes non-linear, with lower frequencies emphasized. The frequencies affected, how much they are boosted, and the distance within which proximity effect will occur, will vary with the design of individual models. Because there is no boost of

signals arriving from a distance (such as those returning from the loudspeaker), proximity effect can greatly reduce feedback potential when a mic is used close to a sound source.

Much of the boosted output resulting from proximity effect occurs in the voice range, making male vocalists sound more grown up, or female voices more full. The boost in mic output may be 20dB or more in the lower portion of the spectrum, meaning care must be taken to insure that the gain control at the top of your mixer's channel strip is properly set to compensate for the higher microphone level. Because the increased output is predominantly low frequencies, you may still suffer some intelligibility loss, or may end up with a "muddy" sound, even after you have adjusted the gain control. If so, turn down the low-frequency control on the channel EQ to restore a natural quality. Microphones designed specifically for close vocal use are often designed to have a rolled-off low frequency response, measured at a distance. When the mic is used up close, as intended, proximity effect restores the low end. Such mics may sound a bit anemic, of course, when used at a distance. 

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- Lavalier mic tips
- What's in a NOM?
- RFI: What it is and what to do about it
- Answers to readers' questions

Feedback and the common microphone

You're probably quite aware that certain microphones tend to contribute less to feedback problems than do others. And you may have wondered why one microphone works well two feet from a sound source, while it is nearly impossible to use another at more than a few inches without the system going into feedback. What's the difference between the mics? Let's look at the role microphone selection can play in our attempts to reduce the potential for feedback. To do this we need to learn a little bit about the performance characteristics of different microphone types.

Linearity

A microphone converts acoustic energy (sound) to electrical energy. If it were able to do so with absolutely equal sensitivity to all frequencies, we would say it had a "flat" or "linear" frequency response. A peak in a microphone's response curve indicates that the microphone is more

sensitive to some frequencies than to others. This peak may cause feedback to occur at those frequencies before the required system gain can be achieved. A linear response is particularly desirable as the distance is increased between the microphone and the sound source. Typical distant applications include miking choirs, dramas, and children's programs.

Bandwidth

This has no connection to the girth statistics of your praise band. It refers to how extended the response of the microphone is. While we usually equate *more* with *better*, a frequency response of DC to purple is not necessarily a great thing. Low frequency feedback problems can often be eliminated by using a mic with a less extended low-frequency response; some mics offer a bass roll-off switch to limit their response. The Low Cut filter on each channel of many Mackie mixers will give you this control where you need it, when you need it.

Polar patterns

The microphone polar pattern chart (see page 7) reveals several things about the ability of each microphone type to reduce feedback. First, it tells us the Random Energy Efficiency (REE) of the mic. REE is a measurement of how much energy a microphone picks up (relative to an omnidirectional mic) when subjected to all frequencies of sound arriving with equal intensity from all directions. This is not an indication of the mic's *sensitivity* rating.

The REE dictates what the relative working distance of the mic would be

continued inside on page 7

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