# AUDIO ENGINEERING BASICS

## Basic Analog Course

### UES Course No. AE-101


Prepared By:

UNIQUE ENGINEERED SOLUTIONS

Serving Sophisticated Clients

**AUDIO ENGINEERING BASICS**

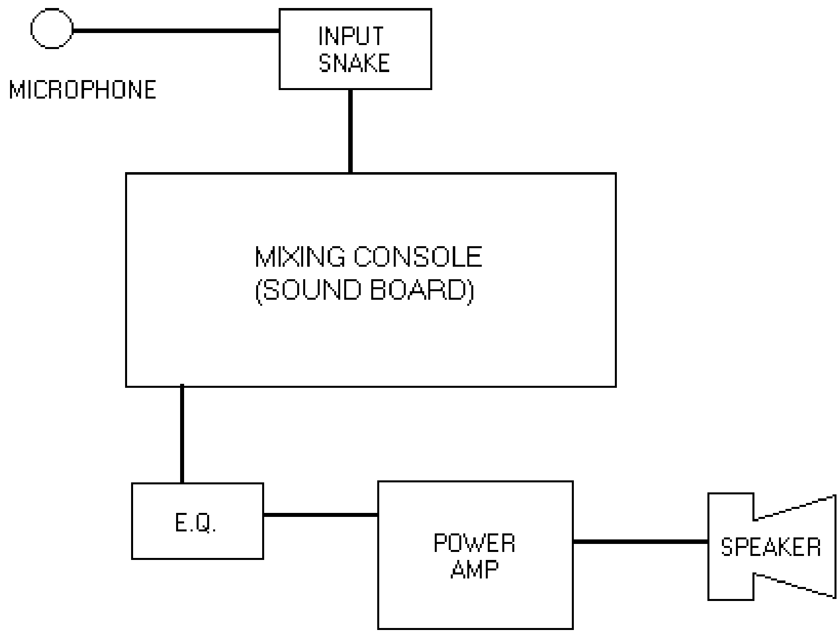
# TABLE OF CONTENTS

An audio engineer (also known as a sound engineer) is the person responsible for the technical aspects of a sound recording, sound system, or sound broadcast. They operate the equipment, (ie. mixing console, microphone) and ensure that the material is of the highest quality available (high fidelity).

In a recording studio the audio engineer generally has very little input into the creative process, this is left up to the record producer. The audio engineer must interpret the direction of the producer and use appropriate techniques to achieve that goal.

Audio engineering covers a wide range of disciplines including, acoustics and psychoacoustics, electronics, and music. Good audio engineers also have good personal skills and work well with others.

In this part of training, we will take an overall look at the various components involved in a professional audio sound reinforcement and/or recording system.




A BASIC SYSTEM

In a basic public address system, you need a microphone (INPUT TRANSDUCER), an amplifier (SIGNAL PROCESSOR) and a speaker (OUTPUT TRANSDUCER). Also needed are a power source and interconnecting cables. When properly configured, these basic components will allow a speaker, vocalist or instrument to project (or amplify) his or her voice/sound/etc..

What is sound reinforcement & why is it necessary?

Sound reinforcement is the method by which a sound system is used to amplify & deliver a boosted output back to an audience via loudspeakers in the form of sound. A sound system can be defined as a group of electronic systems/networks that amplify sound. There are many reasons why this could be done. The most obvious would be to allow a voice on-stage to be heard at the back of an auditorium. In this case, the objective would NOT be to make the sound any louder than it is heard up close. Other reasons could include amplifying vocals in a musical so that they are "larger than life" to add excitement. Sound may also need to be routed into another room for a secondary crowd. (In the case that a lector hall may be filled, etc.)

What components are necessary in a sound reinforcement system?
A sound system contains three main groups into which the components are grouped. The input transducers, signal processors, & output transducers. An input transducer is a device that converts sound waves or other patterns into a signal. (A microphone or contact pickup) A signal processor is any device that takes the signal & changes one or more characteristics of it. (An amplifier, mixer, etc.) An output transducer is a device that converts the processed signal back into sound. (Loudspeaker, headphone, etc.)

- INPUT TRANSDUCERS

There are several different microphones & contact pickups. Just to name a few, there are CD laser pickups, phonograph pickups, microphones, magnetic pickups, etc. Contact pickups convert a pattern (sound wave, fluctuating magnetic field, etc.) into an electrical signal. A signal is an electrical representation of a sound wave in the form of a fluctuating voltage.

- SIGNAL PROCESSORS

These devices take the signal from the input transducers (microphone) & change one or more properties. Amplifiers strengthen the signal while an equalizer varies the frequency response. There are also other complex units. For example: A BBE processor will take a signal & bring it as close as possible to the way it sounded in the recording studio.

- OUTPUT TRANSDUCERS

There are many different types of transducers. A woofer reproduces the low frequencies, a mid-range driver reproduces mid frequencies, while a tweeter reproduces the highs. There are also full range speakers that cover all of these frequencies, although most of them don't cover the extreme low or high frequencies as well.

WHAT IS THE ROLE OF A MIXING CONSOLE (OR MIXING BOARD)?

A mixer (Which falls into the signal processing category) has 4 basic functions. (Plus many other luxuries, but these four are present in most.)

- Mix down several signals (inputs) down into one output signal
- Bring turntable or microphone signals to line level (pre-amplification)
- Provide a cuing system to monitor channels
- Equalize or fine-tune each channel individually

A mixer is a device similar to an electronic funnel, allowing for volume adjustments of several channels as they are simultaneously mixed down into one output. It also contains a preamplifier that amplifies microphone & turntable inputs thus bringing them up to line level. This is done to compensate for the differences in volume between line sources (CD players, cassette decks etc.) & sources which produce a low amplitude (level) gain such as a microphone or turntable. The pre-amplifier also equalizes turntable inputs to match the R.I.A.A (Recording Industry Association of America) standard. This is the standard set by the recording industry to control how a record sounds when being played on various turntables.

Usually mixers contain a type of cueing device to monitor individual channels or the overall mix. Some mixers have an amplifier specially built in with a volume control. PFL (Pre-fade listen) or SOLO will allow you to monitor a source before it is faded into the mix. (Fader down) I personally love this feature because it allows you to monitor several channels simultaneously.

Advanced mixers contain extra gadgets to adjust, but usually at least a three-band equalizer for each channel is included. Other individual adjustments may also be present, such as a sub-channel selector. This allows you to assign many channels to different "sub masters". This is included on advanced boards & is handy if you want to have one sub master with all of your percussion channels under 1 master & all your vocals sub-divided under one master, etc. The higher priced sophisticated models usually allow for auxiliary output feeds. This allows you to direct your output to monitors, delay processors, etc. In the case of delay processors, you would direct one of the auxiliary outs to the input of the processor & then the output of the processor back into another channel on your mixer. This way you can control exactly how much reverb you would like to add to your mix. If your mixer has auxiliary returns, you could also choose to route the "wet" (reverberated) signal back into the board via an auxiliary return.

BASICS OF SIGNAL FLOW

When preparing this text, I thought long about when to get into some of the more technical aspects of audio engineering. It boiled down to an internal debate over could I teach “how” before teaching “why”. As it turns out, the debate is still ongoing. However, it was brought to my attention that many new comers to professional audio don’t have a good enough grasp on basic analog audio signal flow.

In most consumer electronic audio devices, you can almost stumble into the fact that an output on device “A” goes into the input on device “B” and so on. Such as the outputs from a CD player plug into the CD inputs on the receiver. Unfortunately, professional audio equipment is not so simple. In fact, nowadays, most consumer audio equipment does not have mixed impedance issues (High impedance versus low impedance).

It is common to refer to various connections in a system as "low impedance" or "high impedance". This often causes confusion. When trying to determine if you can connect source component A to load component B, forget the terms "high Z" and "low Z".

(Also, to avoid confusion when discussing connections, keep signal flow in mind. I often see folks say "...I connect my amp's inputs into my mixer's L/R outputs...". The signal does not go from the amp to the mixer, it goes from the mixer to the amp. So you need to say "... I feed my mixer's L/R outputs into my amp's inputs...".)

When you want to connect two pieces of gear (a "source" and a "load"), you need to ask these questions:

1) What signal level does the receiving (load) device's input want to see? (Usually mic level, line level, or speaker level.)
2) Is the source device's output able to produce the required voltage?
3) What is the minimum load impedance into which the source component can produce the required signal level?
4) 4) What is the input impedance of the load device, and is it within the range that can be driven by the source component?

Let's take a look at a few common loads that we might want to drive such as the "low impedance" microphone input on a mixer. Most mics are specified as being something on the order of 150 ohms. So the mic input's impedance must be around that, right? Nope! Commonly it is about 2000 ohms (2K ohms). (150 ohms is likely the output impedance of the mic, and is usually of no concern to us.)

Another example would be that rack of five power amplifiers behind the speaker stack. All the channels are daisy chained. The input impedance is: 2K ohms - The same as the mic input. (The output impedance of the mixer or crossover that's driving these amps is likely between 50 and 100 ohms - pretty close to that of the mic, and again not of particular concern). The difference between these inputs is the signal level that the load device requires. The typical signal level needed to drive a mic input that's set up for a vocal mic might be around -30dBu (about 0.025 volts). With a rock band playing, that stack of power amps could require around +12dBu (about 3.0 volts). Plug a mic @ -30dBu into the amps and you'll get nothing out. Plug a mixer output @ +12dBu into a mic input and you'll get a grossly distorted signal (if it does not blow the mic input). (We can of course turn down the mixer's output level, but the result would have a lot of hiss.)

We need to see if the mixer's output is designed to deliver the required voltage into the load impedance we have. A typical mixer will deliver + 12dBu, +20dBu, or even more into a 2K ohm load, so this is usually no problem. However, some entry-level mixers, EQs, or crossovers may be limited to driving a 5K or 10K load at this signal level. Once the load impedance decreases below a certain point, the signal level that the source can safely deliver into the load decreases. Our cheapo mixer can likely drive a 2K ohm load fine at a reduced signal level, so if we use amps that only need say 0dBu, we might be ok. (Of course if we can afford that many amps, we're not likely to be using a cheapo source component.)

To help us further understand signal flow and gain, I would like to turn your attention to the following documents found in our appendix:

- Appendix C - Signal Flow
- Appendix E - Unity Gain and Impedance Matching
- Appendix F - Setting Sound System Level Controls **(**IMPORTANT **)**

A BIT ABOUT EQUALIZERS

An equalizer separates a signal into separate frequency bands to allow for individual volume adjustments of each band. Through my experience, I have found MUCH controversy when it comes to setting an EQ! I've talked to a few guys that swear by the smiley face method or a slight variation of it! (The EQ settings gradually dipping down & then back up from low to high bands) They seem to think that it sounds a lot better this way then keeping the EQ flat. HOWEVER, I have found more people in favor of just leaving the EQ flat, finding the problem frequencies that are culprits of feedback & cutting or reducing those & then using your ear for any small adjustments. I for one believe in this theory, because the more you mess around with an EQ, the more the sound deviates from it's original state. Unless you have to compensate for a bad sounding speaker system, then you might have to play with the EQ a bit.

The feedback problem frequencies that you should keep an eye out for are usually around 1-4 kHz. Find out which ones they are by ringing out the system and either cut them or reduce them as necessary.

Ringing out a system consists of bringing up a microphone until you start to get feedback, and then "EQing away" the problem frequencies by cutting/reducing them. This method works in most cases and can also be used for monitors as well.

Basically it all comes down to trusting your ear. If it sounds good to you, it will sound good to your audience.

ELIMINATING NOISE

Noise in the system can be the result of many sources. Here are some that are most likely to give you problems:

- Poor wiring in the area of the system
- Fluorescent lights
- Dimmers
- Poor design (particularly shielding & location of mixer's transformer)
- Magnetic fields (induced by close components-- mainly power amplifiers)
- Radio stations or other transmitters in area
- Large motors near by
- Grounding problems (mainly when using equipment manufactured by various companies)
- Noise in the input signal (mainly guitar pickups)
- Poor cables (mainly on the inputs)

The majority of the above could be dealt with by using high quality cables & balanced lines.

To locate the problem, unplug all the inputs going into the mixer. (Make sure to turn off the amplifiers first, because this could induce a loud pop or transient that could damage the speakers) If the noise has disappeared, then you know the problem is in one of the inputs. Plug each input back in, one by one & as soon as a noise is heard, then you know which source is causing the problem. From there, check the cable going from the source to the mixer & look for grounding problems, pickup noise, etc.

Here's a helpful suggestion list to help with input noise problems:

- Change the cable to higher quality with a high degree of shielding
- Where possible, use balanced sources. (With some sources you have a choice of a balanced/unbalanced output) Even using balanced cable will even improve noise in unbalanced sources if the mixed inputs are balanced
- Use direct boxes whenever possible for instruments that plug into the mixer. This will isolate the signal, & convert any high impedance instruments to balance low impedance values and allow you to "lift" the grounds on these devices
- Reduce all lengths of cables
- Check to make sure input cables are not situated too close to a transformer, motor, amplifier or any source of magnetic radiation
- Plugging system components into different circuits can in some cases cause problems. Try to connect all PA components into the same circuit, even if you have to run extension cords in order to do this.
- Turn all the lights on the dimmers off, or if possible turn them fully on

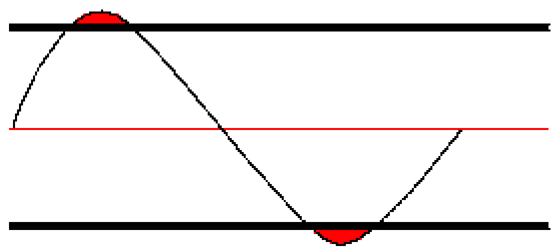
If the noise doesn't disappear after unplugging all the inputs, then the problem is located somewhere else in the system. Go through the process of elimination in order to locate the problem. Plug the sources directly into the equalizer or amplifier & if the noise disappears, then the problem is in the mixer. If this is the case, start by disconnecting all the effects devices such as reverb, delays, etc. & if the noise disappears, then it is a particular effects device that is causing the problem.

Note: Some hiss occurs when an input is at a high level. (All the way up) This is perfectly normal.

## CLIPPING

What is clipping? Clipping is basically a form of distortion. It is usually used to describe distortion in general, but is in fact a type of distortion that clips off the peak of a signal. This involves closer analysis of a signal to better understand the concept of clipping.

A signal is an alternating strength of voltage usually changing from positive to negative repeatedly. The highest extent of the signal is called the amplitude. When a signal's voltage is plotted against time, the result is that of the image below and is properly referred to as a "waveform."



The voltage value is vertical in this diagram & it is plotted against time on the horizontal axis. The horizontal red line in the middle represents "0" voltage.

Please note that the above diagram is a clipped signal. The two solid black lines at the top and bottom represent the amplifier's maximum voltage. Note that the signal's amplitude extends above & beyond the amplifier's maximum voltage & results in getting "clipped off." The filled red region represents the clipping region. The end result is severe distortion through the PA, which is very unpleasant & could damage your speakers.

Most amplifiers have an LED that warns when clipping occurs. If this light is excessively lit up or on constantly, turn down the volume immediately.

- Input Clipping (front end clipping)

Each input on a mixer has a preamplifier as stated earlier in this article. If this amplifier is being clipped, (much the same way we discussed above) the output of the device is up too loud & the peak of the signal is being "clipped off" by the preamplifier. Several mixers have input clip lights for each channel and if this light is on frequently, set the gain/trim down (usually located at the top of the channel) on the mixer for that channel. Several other signal processing components (equalizers, reverb, crossovers, etc.) use the same principle, and if the clip light is flashing on constantly or on all the time, turn down the output of the device causing the problem until the light disappears.

WHY DO SPEAKERS BLOW?

There are many reasons why speakers blow, but most fall into the following categories: over powering, under powering, transients, feedback, dropping & bad cables.

- Over Powering

This is the method that most people associate the action of "blowing a speaker", but it is often not the one that causes the problems. If the RMS rating of the amplifier is close to the RMS rating of the speakers & the amplifier is not clipping, then everything should be fine.

- Under Powering

Didn't think you could blow a speaker this way, did you? This is one of the frequent causes for speaker failure. The key here is that if an amplifier is used that puts out a significantly lower power rating than the speakers & clips the amplifier in the process, the speakers receive a greater voltage than they would when the signal was clean. This is converted into mechanical energy by the speaker (moving it back & forth) and literally causes the coil to burn. As long as the amplifier is not clipping, nothing will damage.

- Transients

This is a technical term for "sudden loud bursts" in a speaker system. When a speaker receives an extremely high amplitude signal suddenly, the cone wants to keep travelling forward due to the force of inertia. If this extends too far, the cone could tear &/or hit the magnet assembly & break. This usually happens when a microphone is dropped while on, or when a cable is plugged in or unplugged from the mixer, etc.

- Feedback

Feedback is the loud squeal through the PA when a microphone picks up audio, then feeds it back through the PA & continues this cycle over & over again. If a squeal lasts very long, the intense vibration could burn the coils of the tweeters and/or horns out.

- Dropping

Most cabinets can take a beating, but if a cabinet takes a hard impact, internal parts can shift & throw the component out of alignment. If this occurs, parts could rub & the component that is rubbing could rub through & cause failure.

- Bad Cables

A missing ground could cause severe oscillations. These vibrations are so high-pitched that the human ear cannot detect them, but it wears out the tweeters. This could eventually burn the tweeters out. A high quality cable could solve this problem, because it will have a smaller chance of a bad ground connection & a blown tweeter can pay for a significant number of good cables.

An **audio engineer** (also known as a **sound engineer**) is the person responsible for the technical aspects of a sound recording, sound system, or sound broadcast. They operate the equipment, (ie. mixing console, microphone) and ensure that the material is of the highest quality available (high fidelity).

In a recording studio the audio engineer generally has very little input into the creative process, this is left up to the record producer. The audio engineer must interpret the direction of the producer and use appropriate techniques to achieve that goal.

Audio engineering covers a wide range of disciplines including, acoustics and psychoacoustics, electronics, and music. Good audio engineers also have good personal skills and work well with others.

In this part of training, we will explore many aspects of MIXING CONSOLES.

In professional audio, a **mixing console**, **mixing desk**, or **audio mixer** is an electronic device for combining—also called "mixing"—(Brain, n.d.), routing, and changing the level, tone, and/or dynamics of audio signals. A mixer can mix analog or digital signals, depending on the type of mixer.

Mixing consoles are used in many applications, including recording studios, public address systems, sound reinforcement systems, broadcasting, television, and film post-production. An example of a simple application would be to enable the signals that originated from two separate microphones (each being used by vocalists singing a duet, perhaps) to be heard through one set of speakers simultaneously). When used for live performances, the signal produced by the mixer will usually be sent directly to an amplifier, unless that particular mixer is “powered”.

Each signal that is input to the mixer has its own “channel.” Depending on the specific mixer, each channel is stereo or monaural. On all mixers, each channel has an XLR input, and many have RCA or quarter-inch line inputs. Each channel on a mixer always has a linear pot, or potentiometer, controlled by a sliding volume control, that allows adjustment of the level, or amplitude, of that channel in the final "mix". A typical mixing console has many rows of these sliding volume controls. Each control adjusts only its respective channel; therefore, it only affects the level of the signal from one microphone or other audio device.

Above each control, there may be several rotary controls (knobs) that attenuate the signal to that channel and that equalize the signal by separately attenuating a range of frequencies (e.g., bass, midrange, and treble frequencies). On the right hand of the console, there are typically one or two master controls that enable adjustment of the console's output level. Prior to the master controls, there may be an intermediate set of outputs and inputs for the addition of an external equalizer or audio effect, such as delay or reverb. Finally, there are usually one or more VU meters to indicate the levels for each channel, or for the master outputs, and to indicate whether the console levels are over-modulating or clipping the signal. All mixers have at least one *additional* output, beside the one main output. The operator can vary the mix (or levels of each channel) for each output. All mixers also have equalizers of varying sophistication.

For an in-depth look at mixers, we would look at Appendix "A" excerpted from "The Soundcraft Guide to Mixing". Soundcraft is the premier manufacturer of high-end Mixing Consoles. Also provided, is a FAQ sheet on mixers which I found on Soundcraft's website.

An **audio engineer** (also known as a **sound engineer**) is the person responsible for the technical aspects of a sound recording, sound system, or sound broadcast. They operate the equipment, (ie. mixing console, microphone) and ensure that the material is of the highest quality available (high fidelity).

In a recording studio the audio engineer generally has very little input into the creative process, this is left up to the record producer. The audio engineer must interpret the direction of the producer and use appropriate techniques to achieve that goal.

Audio engineering covers a wide range of disciplines including, acoustics and psychoacoustics, electronics, and music. Good audio engineers also have good personal skills and work well with others.

In this part of training, we will explore many aspects of AMPLIFIERS.

## Amplifier Basics

The primary thing to know about operation of an amplifier is, when properly connected it either works or it doesn't. To further emphasize this point, most professional power amplifiers in a sound reinforcement system have very few, if any, adjustments knobs.

It is a good idea to have at least one spare amplifier of sufficient size to use as a back-up during live performances. If you are bi-amping or tri-amping in stereo, a simple reconfiguration of your system will do in a pinch, if your amps are properly sized.

If you run across a unit that is not working, be sure to check for proper connection (power wiring, input wiring and output wiring) as well as fuses. Before pulling the unit from service, it's a good idea to get a second opinion. Often times a minor incorrect connection can be overlooked by veteran audio engineers. Most modern amplifiers have many hook-up options available and thereby offering plenty of opportunities to get it wrong. Another source of your problem may be the wire or cables. A cable or continuity tester as well as spare cables will help this sort of problem from defeating your successful hook-up.

**Cooling**

When you are using amps that are fan-cooled and want to allow spaces between pieces of equipment in your rack, make sure you block the front with blank, solid (not perforated) panels. This will allow the rack to act as a chimney with hot air exhausting at the top, not re-circulating between adjacent amplifiers. When you are using convection-cooled amps in very high ambient temperatures, you may find that your temperature indicators are starting to illuminate. Typically, adding modest amounts of air movement will enable your amp to dissipate any excessive heat and regain its normal composure. However, if your temperature indicators continue to illuminate, consider the following possible causes:

- Insufficient air movement.
- Overdriving of the input stage (severely into clip).
- Very low-impedance loads.
- High ambient temperatures.

If you can't, or don't want to change the preceding conditions, two possible alternatives are available to add the necessary air movement. First, you can add fans to direct air onto any surface of the amplifier. Second, you can space the amplifiers in the rack using perforated panels or leaving the empty slots open. This will allow the top and bottom covers to act as radiators. In extreme conditions, a combination of these two methods may be required, as would be expected for proper thermal functioning of any amplifier.

**Hum & Buzz**

- It is imperative that all of your electrical equipment share the same power ground reference.

- Unless you are interfacing to a microphone, the shield of the cable should only be connected at one end.

- Do not pass signal ground between electrical components in a grounded source system.

- If you wish to avoid ground loops, it doesn't matter if you lift the input or output signal ground or your system topology, just be consistent. Personally I prefer to lift the input signal ground and it has always been successful.

- NEVER use a ground lift adapter to lift the power ground on a 3-wire AC cord; this is not its intended purpose. It is better to have it SAFE than SILENT!! Look for the true source of the noise.

- Even when interfacing to an unbalanced load, it is preferable to use two-conductor shielded cable.

- Get rid of the lighting company!

**Input Wiring**

- For all input connectivity, use shielded wire only. Cables with a foil wrap shield or a high-density braid are superior. Cables with a stranded spiral shield, although very flexible, will break down over time and cause noise problems.

- Try to avoid using unbalanced lines with professional equipment. If you have no choice, keep the cables as short as possible.

- To minimize hum and cross-talk, avoid running low-level input, high-level output and AC power feeds in the same path. Try to run differing signal paths at 90 degrees to one another. If you must use a common path for all cables, use a star-quad cable for the low-level signals.

- When changing input connectors or wiring, turn the amplifier level controls all the way down (counter-clockwise) before connecting or disconnecting input plugs.

- When changing output connections, a professional dude will turn the amplifier level down and the AC power off to minimize the chance of short-circuiting the output.

**Output Wiring**

- Choose carefully when selecting speaker enclosure connectors.

- To prevent possible short circuits, wrap or otherwise insulate exposed loudspeaker cable connectors.

- Do not use connectors that might accidentally tie conductors together when making or breaking the connection (for example, a standard, 1/4-inch stereo phone plug).

- Never use connectors that could be plugged into AC power sockets. Accidental AC input will be an electrifying experience for your equipment. But you will find out real quick if your speakers are any good at 60 Hz.

- Avoid using connectors with low current-carrying capacity, such as XLRs.

- Do not use connectors that have any tendency to short.

- To maintain good bass response, use the lowest DC resistance cable you can afford which will terminate safely in your connectors.

**Speakon Connectors**

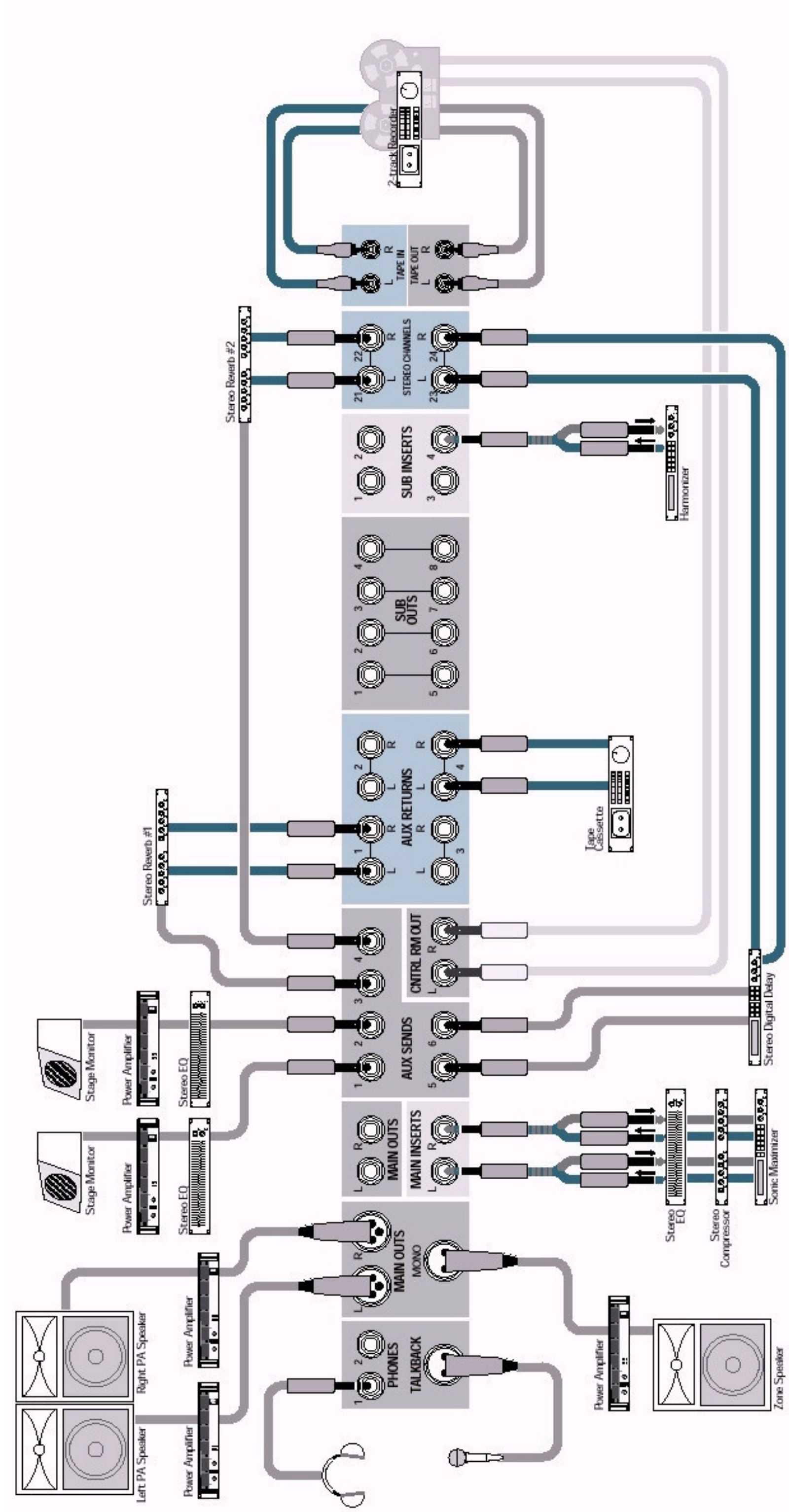
For amplifiers, the most popular termination device on professional products has been the dual banana. However, recent regulatory requirements in Europe have outlawed the use of the dual banana plug and forced users to terminate speaker cables with spade lugs or bare ends an approach that is clearly not advantageous to the customer who wants to reconfigure his system or quickly change out a defective product. It is possible that similar regulatory controls will appear worldwide over the next few years.

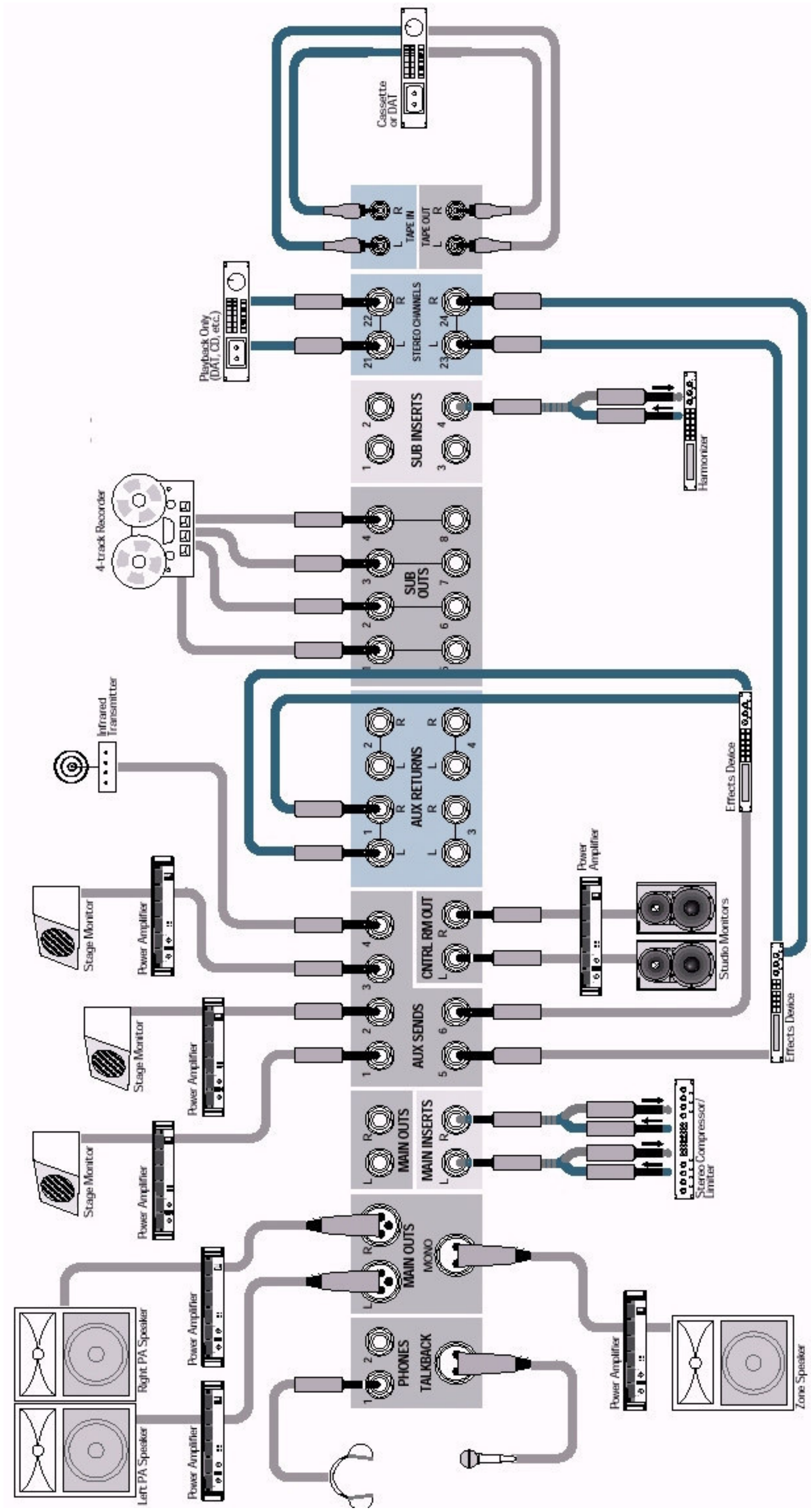
One solution to this problem is to use the Neutrik Speakon connector. Most amplifier manufactures wanted to develop a system for you that eliminated the need for specialized, time-consuming, interface cables. The major loudspeaker manufacturers have been using Speakon connectors for the input termination on their products for several years now, so you can be assured of the connector's reliability in the workplace. With Speakon connectors, you can plug straight from the amp to the speaker, and start making those great sounds right away.

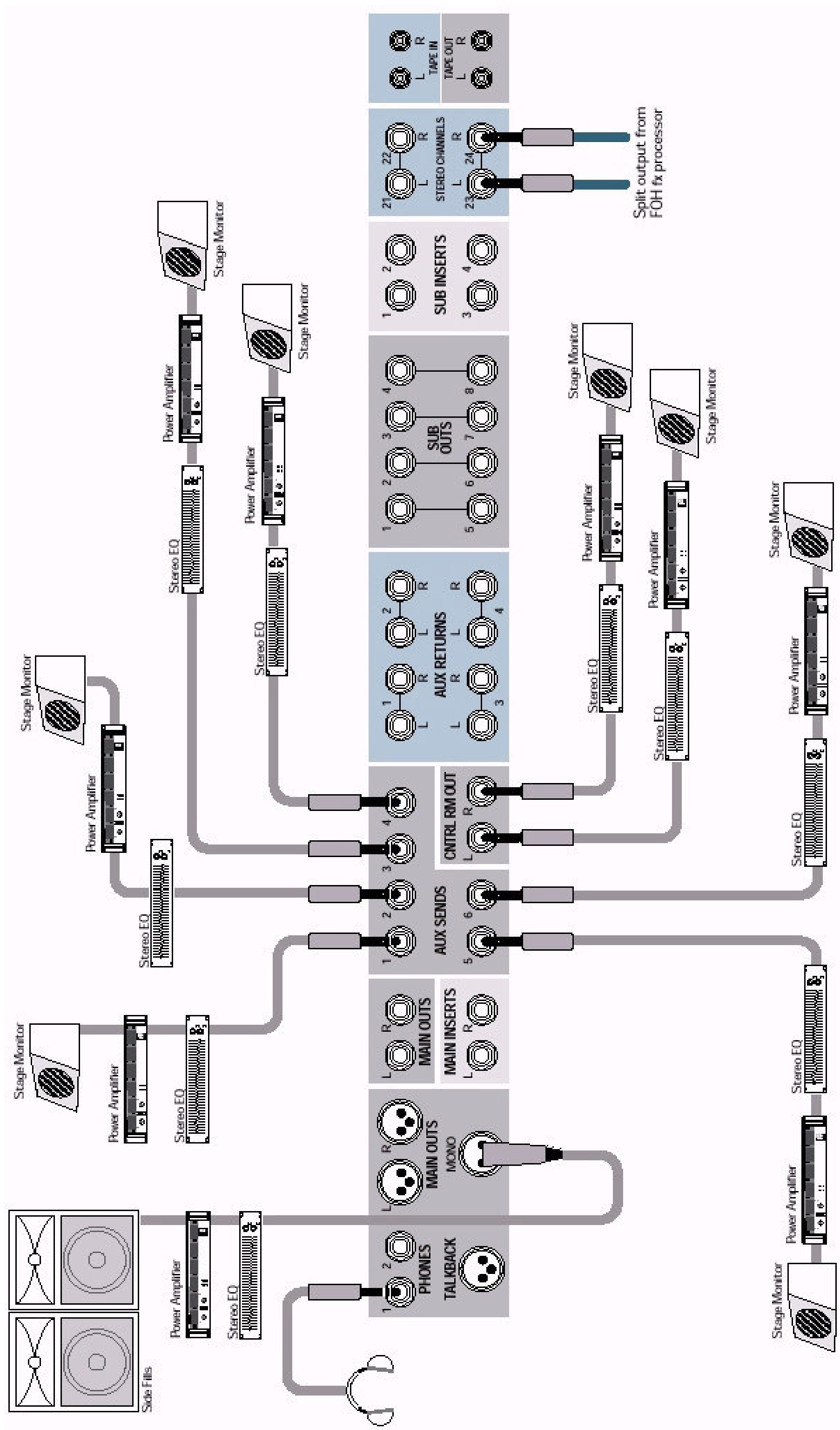
The Speakon connectors used on most amplifiers meet all known safety regulations. Once wired correctly, the connector cannot be plugged in backwards, causing the type of inverted polarity situations that are common with banana hookups. It will provide a safe, secure and reliable method of interfacing your amplifier to the load.

## Hook-ups

The following diagrams indicate various schemes for wiring your system amplifiers for both FOH as well as monitoring.


2-track Recorder
TAPE IN
TAPE OUT
Stereo Reverb #2
STEREO CHANNELS
SUB INSERTS
Harmonizer
SUB OUTS
AUX RETURNS
Stereo Reverb #1
Tape Cassette
CNTRL RM OUT
AUX SENDS
Stereo Digital Delay
Stage Monitor
Power Amplifier
Stereo EQ
Stage Monitor
Power Amplifier
Stereo EQ
MAIN OUTS
MAIN INSERTS
Stereo EQ
Stereo Compressor
Sonic Maximizer
MAIN OUTS
MONO
Right PA Speaker
Power Amplifier
Left PA Speaker
Power Amplifier
PHONES
TALKBACK
Power Amplifier
Zone Speaker


Cassette or DAT
TAPE IN
TAPE OUT
Playback Only (DAT, CD, etc.)
STEREO CHANNELS
SUB INSERTS
Harmonizer
4-track Recorder
SUB OUTS
Infrared Transmitter
AUX RETURNS
Effects Device
Stage Monitor
Power Amplifier
CNTRL RM OUT
Studio Monitors
AUX SENDS
MAIN OUTS
MAIN INSERTS
Stereo Compressor/ Limiter
MONO
PHONES
TALKBACK
Right PA Speaker
Left PA Speaker
Zone Speaker


TAPE IN
TAPE OUT
STEREO CHANNELS
Split output from FOH fx processor
SUB INSERTS
SUB OUTS
AUX RETURNS
CNTRL RM OUT
AUX SENDS
MAIN OUTS
MAIN INSERTS
MAIN OUTS
MONO
PHONES
TALKBACK
Stage Monitor
Power Amplifier
Stereo EQ
Side Fills

An **audio engineer** (also known as a **sound engineer**) is the person responsible for the technical aspects of a sound recording, sound system, or sound broadcast. They operate the equipment, (ie. mixing console, microphone) and ensure that the material is of the highest quality available (high fidelity).

In a recording studio the audio engineer generally has very little input into the creative process, this is left up to the record producer. The audio engineer must interpret the direction of the producer and use appropriate techniques to achieve that goal.

Audio engineering covers a wide range of disciplines including, acoustics and psychoacoustics, electronics, and music. Good audio engineers also have good personal skills and work well with others.

In this part of training, we will explore many aspects of GROUNDING.

OK, You have a buzz in your sound system.....NOW WHAT?

Don't be discouraged by all the reading and information you are about to encounter. Buzzes, hums and ground loops are a VERY complex problem. Just take your time and pay close attention to the following images and you will be rewarded with a very quiet and hum free system.

### Ground Loops

Almost all cases of noise can be traced directly to ground loops, grounding or lack thereof. It is important to understand the mechanism that causes grounding noise in order to effectively eliminate it. Each component of a sound system produces its own ground internally. This ground is usually called the audio signal ground. Connecting devices together with the interconnecting cables can tie the signal grounds of the two units together in one place through the conductors in the cable.

Ground loops occur when the grounds of the two units are also tied together in another place: via the third wire in the line cord, by tying the metal chassis together through the rack rails, etc. These situations create a circuit through which current may flow in a closed "loop" from one unit's ground out to a second unit and back to the first. It is not simply the presence of this current that creates the hum—it is when this current flows through a unit's audio signal ground that creates the hum. In fact, even without a ground loop, a little noise current always flows through every interconnecting cable (i.e., it is impossible to eliminate these currents entirely).

The mere presence of this ground loop current is no cause for alarm if your system uses properly implemented and completely balanced interconnects, which are excellent at rejecting ground loop and other noise currents. Balanced interconnecting was developed to be immune to these noise currents, which can never be entirely eliminated. What makes a ground loop current annoying is when the audio signal is affected. Unfortunately, many manufacturers of balanced audio equipment design the internal grounding system improperly, thus creating balanced equipment that is not immune to the cabling's noise currents. This is one reason for the bad reputation sometimes given to balanced interconnects.

A second reason for the balanced interconnects' bad reputation comes from those who think connecting unbalanced equipment into "superior" balanced equipment should improve things. Sorry. Balanced interconnect is not compatible with unbalanced. The small physical nature and short cable runs of completely unbalanced systems (home audio) also contain these ground loop noise currents. However, the currents in unbalanced systems never get large enough to affect the audio to the point where it is a nuisance. Mixing balanced and unbalanced equipment, however, is an entirely different story, since balanced and unbalanced interconnections are truly not compatible. The rest of this note shows several recommended implementations for all of these interconnection schemes.

The potential or voltage, which pushes these noise currents through the circuit, is developed between the independent grounds of the two or more units in the system. The impedance of this circuit is low, and even though the voltage is low, the current is high, thanks to

Mr. Ohm, without whose help we wouldn't have these problems. It would take a very high-resolution “ohm meter” to measure the impedance of the steel chassis or the rack rails. We're talking thousandths of an ohm. So trying to measure this stuff won't necessarily help you. We just thought we'd warn you.

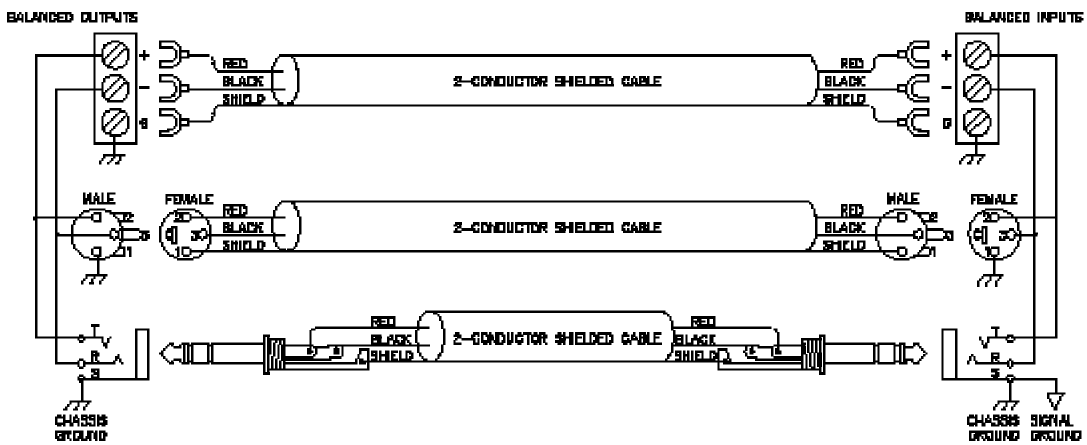



Figure 1a. The right way to do it

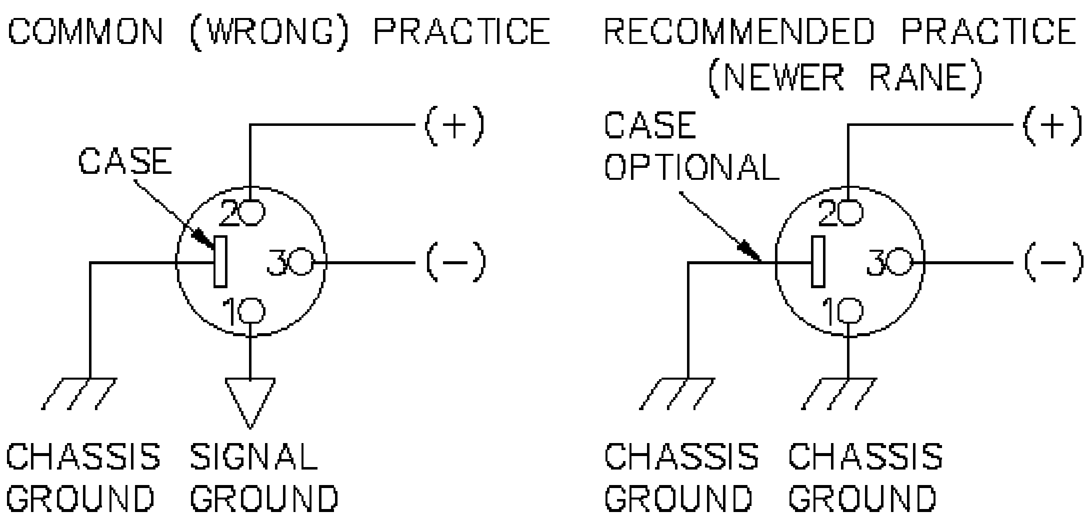



Figure 1b. Recommended practice.

**The Absolute Best Right Way To Do It**

Use balanced lines and tie the cable shield to the metal chassis (right where it enters the chassis) at both ends of the cable.

A balanced line requires three separate conductors, two of which are signal (+ and -) and one shield (see Fig. 1a). The shield serves to guard the sensitive audio lines from interference. Only by using balanced line interconnects can you guarantee (yes, guarantee) hum-free results. Always use twisted pair cable. Chassis tying the shield at each end also guarantees the best possible protection from RFI [radio frequency interference] and other noises [neon signs, lighting dimmers].

It is absurd to think that you can go out and buy pro audio equipment from several different manufacturers, buy standard off-the-shelf cable assemblies, come home, hook it all up and have it work hum and noise free. Plug and play. Sadly, almost never is this the case, despite the science and rules of noise-free interconnect known and documented for over 60 years.

It all boils down to using balanced lines, only balanced lines, and nothing but balanced lines. This is why they were developed. Further, that you tie the shield to the chassis, at the point it enters the chassis, and at both ends of the cable (more on `both ends' later).

Since standard XLR cables come with their shields tied to pin 1 at each end (the shells are not tied, nor need be), this means equipment using 3-pin, XLR-type connectors must tie pin 1 to the chassis (usually called chassis ground) — not the audio signal ground as is most common.
Not using signal ground is the most radical departure from common pro-audio practice. Not that there is any argument about its validity. There isn't. This is the right way to do it. So why doesn't audio equipment come wired this way? Well, some does, and since 1993, more of it does. So why doesn't everyone do it this way? Because life is messy, some things are hard to change, and there

will always be equipment in use that was made before proper grounding practices were in effect.

Unbalanced equipment is another problem: it is everywhere, easily available and inexpensive. All those RCA and ¼" TS connectors found on consumer equipment; effect-loops and insert-points on consoles; signal processing boxes; semi-pro digital and analog tape recorders; computer cards; mixing consoles; et cetera. The next several pages give tips on how to successfully address hooking up unbalanced equipment. Unbalanced equipment when "blindly" connected with fully balanced units starts a pattern of hum and undesirable operation, requiring extra measures to correct the situation.

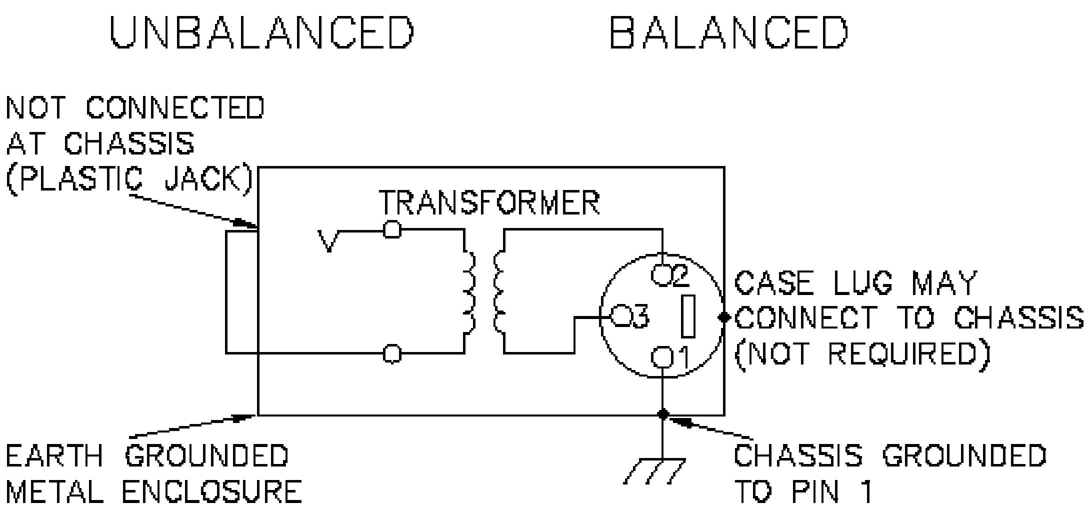



Figure 2. Transformer Isolation

**The Next Best Right Way To Do It**

The quickest, quietest and most foolproof method to connect balanced and unbalanced is to transformer isolate all unbalanced connections. See Figure 2.

The goal of these adapters is to allow the use of standard cables. With these transformer isolation boxes, modification of cable assemblies is unnecessary. Virtually any two pieces of audio equipment can be successfully interfaced without risk of unwanted hum and noise.

Another way to create the necessary isolation is to use a direct box. Originally named for its use to convert the high impedance, high-level output of an electric guitar to the low impedance, low level input of a recording console, it allowed the player to plug "directly" into the console. Now this term is commonly used to describe any box used to convert unbalanced lines to balanced lines.

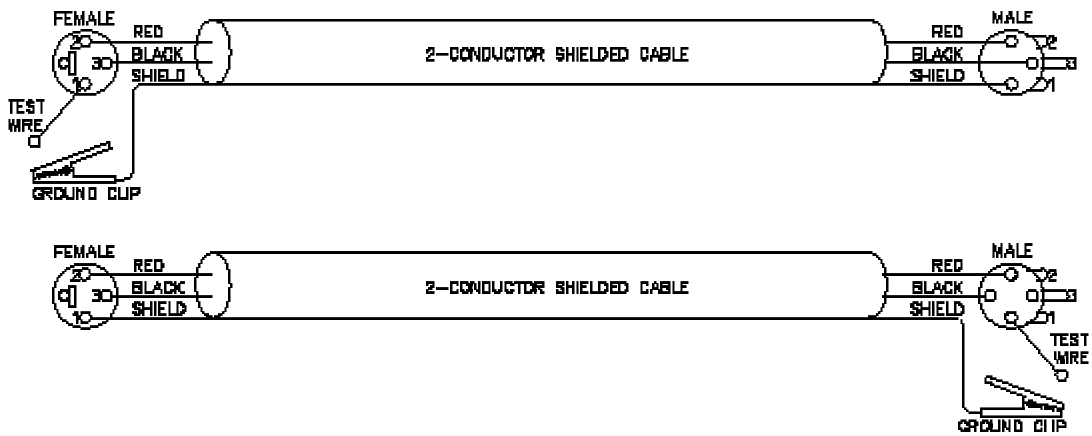



Figure 3. Test cables

**The Last Best Right Way To Do It**

If transformer isolation is not an option, special cable assemblies are a last resort.  The key here is to prevent the shield currents from flowing into a unit whose grounding scheme creates ground loops (hum) in the audio path (i.e., most audio equipment).

It is true that connecting both ends of the shield is theoretically the best way to interconnect equipment though this assumes the interconnected equipment is internally grounded properly.  Since most equipment is not internally grounded properly, connecting both ends of the shield is not often practiced, since doing so usually creates noisy interconnections.

A common solution to these noisy hum and buzz problems involves disconnecting one end of the shield, even though one can not buy off-the-shelf cables with the shield disconnected at one end. The best end to disconnect is a matter of personal preference and should be religiously obeyed; choose inputs or outputs and always lift the side you choose (our drawings happen to disconnect the input end of the cable—the output of the driving unit). If one end of the shield is disconnected, the noisy hum current stops flowing and away goes the hum — but only at low frequencies. A one-end-only shield connection increases the possibility of high frequency (radio) interference since the shield may act as an antenna. Many reduce this potential RF interference by providing an RF path through a small capacitor (0.1 or 0.01 microfarad ceramic disc) connected from the lifted end of the shield to the chassis. The fact that many modern day installers still follow this one-end-only rule with consistent success indicates this and other acceptable solutions to RF issues exist, though the increasing use of digital and wireless technology greatly increases the possibility of future RF problems.

If you've truly isolated your hum problem to a specific unit, chances are, even though the documentation indicates proper chassis grounded shields, the suspect unit is not internally grounded properly. Here is where special test cable assemblies, shown in Figure 3, really come in handy. These assemblies allow you to connect the shield to chassis ground at the point of entry, or to pin 1, or to lift one end of the shield. The task becomes more difficult when the unit you've isolated has multiple inputs and outputs. On a suspect unit with multiple cables, try various configurations on each connection to find out if special cable assemblies are needed at more than one point.

**Ground Lifts**

Many units come equipped with ground lift switches. In only a few cases can it be shown that a ground lift switch improves ground related noise. (Has a ground lift switch ever really worked for you?) In reality, the presence of a ground lift switch greatly reduces a unit's ability to be "properly" grounded and therefore immune to ground loop hums and buzzes. Ground lifts are simply another Band-

Aid® to try in case of grounding problems. It is, however, true that an entire system of properly grounded equipment, without ground lift switches, is guaranteed (yes guaranteed) to be hum free. The problem is most equipment is not (both internally and externally, AC system wise) grounded properly.
Most units with ground lifts are shipped so the unit is "grounded" — meaning the chassis is connected to audio signal ground. (This should be the best and is the "safest" position for a ground lift switch.) If after hooking up your system it exhibits excessive hum or buzzing, there is an incompatibility somewhere in the systems' grounding configuration. In addition to these special cable assemblies that may help, here are some more things to try:

1. Try combinations of lifting grounds on units supplied with lift switches (or links). It is wise to do this with the power off!
2. If you have an entirely balanced system, verify all chassis are tied to a good earth ground, for safety's sake and hum protection. Completely unbalanced systems never earth ground anything (except cable TV, often a ground loop source). If you have a mixed balanced and unbalanced system, do yourself a favor and use isolation transformers or, if you're cheap, try the special cable assemblies described here and expect it to take many hours to get things quiet.
3. Balanced units with outboard power supplies (wall warts or "bumps" in the line cord) do not ground the chassis through the line cord. Make sure any such units are solidly grounded, by tying the chassis to an earth ground using a star washer for a reliable contact. Any device with a 3-prong AC plug, such as an amplifier, may serve as an earth ground point. Rack rails may or may not serve this purpose depending on screw locations and paint jobs.

**AUDIO ENGINEERING BASICS**

# APPENDICES



UES COURSE NO. AE-101

# THE SOUNDCRAFT GUIDE TO MIXING – CONTENTS

# STARTING OUT

## A. What does a Mixer do?

No matter how sophisticated or expensive, all mixers carry out the same basic function - to blend and control the volume of a number of input signals, add effects and processing where required and route the resulting mix to the appropriate destination, which could be power amplifiers, the tracks of a recording device - or both. A mixer is the nerve centre of these sources, and therefore the most vital part of your audio system.

## B. Guidelines in Choosing a Mixer

Audio mixers come in many different sizes and at all price levels, so it's little wonder that people are confused as to what type is actually needed for the job in hand. However there are several questions you can to ask yourself that will help you narrow your search to the most appropriate models.

- What am I going to be using the mixer for - i.e. multitrack recording, live PA work or both?
- What is my budget?
- How many sound sources do I have? As a guideline your mixer needs to have at least as many inputs as sound sources. If you are likely to be buying more equipment in the future you should budget for extra inputs.
- What particular mixer facilities **must I have** for my application? i.e. plenty of EQ, auxiliaries, or Direct Outs for recording.
- How portable does the mixer need to be?
- Will I be doing any location work where there won't be any mains power available?
- Have I read the Soundcraft Guide to Mixing from cover to cover?

Once you can answer these questions satisfactorily you should have a fairly accurate specification for the mixer you need.

## C. The Controls - A Description

This is where we get into the nitty-gritty of what controls and inputs/outputs you'll find on a typical mixer. For this example, we've used a Spirit SX . **If you are already familiar with what the controls on a standard mixer do, then it's OK to skip to section 2.** If you find a term particularly difficult, further explanation can be found in the Glossary (Section 8).

### MONO INPUTS

**A Mic In**
Use this "XLR" input to connect your microphones or DI boxes.

*For Mic Input Wiring Explanations see section 7.*

**B Line In**
Use this connector for plugging in "Line Level" instruments such as keyboards, samplers or drum machines. It can also be used to accept the returns from multitrack tape machines and other recording media. The Line Input is not designed for microphones and although it may be used, will not provide optimum performance with them.

*For Line Input wiring explanations see section 7.*

**C Insert Point**
This is used to connect external signal processors such as compressors or limiters within the input module. The Insert Point allows external devices to be placed within the Input Path - see Fig. 1.1.

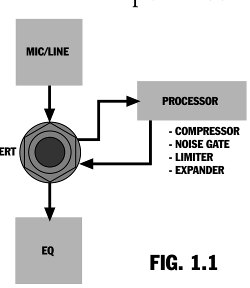
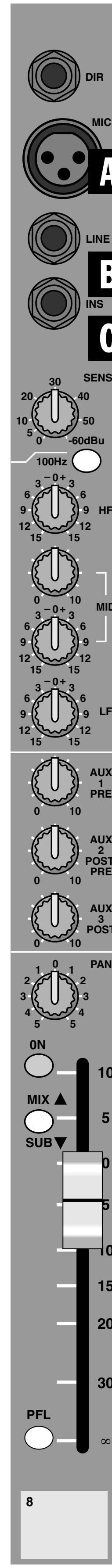



FIG. 1.1

*See Section 2 and 3 for more detail on how to use processors, and Section 7 for information on wiring.*



**D Direct Out**

This allows you to send audio direct from your channel out to a multitrack tape recorder, or to an effects unit when the channel requires its own special effect.

*See sections 2 and 6 for more details on connections and studio techniques.*

**E Gain Control (Input Sensitivity)**

Sets how much of the signal from the mic or line inputs is fed to the channel.

**F HPF (High Pass Filter)**

As the name suggests this switch cuts out the very lowest frequencies of a sound whilst allowing the higher frequencies to "Pass Through". It's particularly useful in live situations to reduce stage rumble or microphone 'popping', which can produce a muddy mix, or to 'clean-up' male vocals and filter out low frequency hum. Some manufacturers may also use the term "low - cut" filter to describe the HPF. See Fig. 1.2.

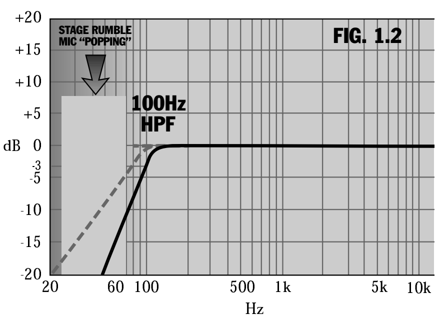


FIG. 1.2

**G EQ Section**

Usually the most closely scrutinised part of any mixer, the equaliser section allows you to change the tone of the sound on each input. An EQ is normally split into "bands", which control a range of frequencies, in a similar fashion to the treble and bass tone controls on your Hi-Fi. Indeed a simple "2 band" EQ is little more than an input treble and bass control. The more bands an EQ has the more sophisticated it is. SX has a 3 band EQ, with a separate control for the middle audio frequencies. This control is also "swept" which provides even more sophistication. Simply described, a sweep EQ allows you to choose the exact frequency to cut and boost, rather than having it chosen for you, as on normal "fixed" controls.

*We will talk in more detail about EQ in section 3.*

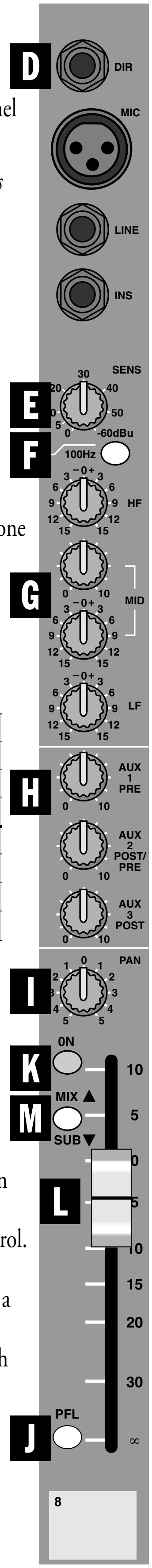


FIG. 1.3

**H Auxiliary Section**

Typically, these controls have two functions: First, to control the levels of *effects* such as reverb from external effects units that have been added to the input signal, and second to create separate musician's *"foldback"* mixes in the studio or on stage.

*How to use auxiliaries, connecting them to external equipment and other applications are described in section 3.*

**I Pan (Panoramic Control)**

This determines the position of the signal within the stereo mix image or may be used to route (send) the signal to particular GROUP outputs as selected by the ROUTING SWITCHES (see below).

**J Solo (PFL and Solo in Place)**

The PFL solo switch allows you to monitor an input signal independently of any other instruments that have been connected, which is useful for troubleshooting, or setting an instrument's Input Preamp Gain and EQ setting.

Pre-Fade Listen (PFL) is a type of solo that allows you to monitor your sound BEFORE THE FADER. In other words when you move the input fader in PFL mode the level will not change, nor will you hear any effects. Because effects and volume are not a distraction, PFL solo is very useful for setting proper input preamp levels.

Some Soundcraft mixers use SOLO IN PLACE, which allows you to monitor signals after the fader in their true stereo image, and with any effects that have been added. This type of Solo is less good for level setting, but more useful in mixdown situations for auditioning sounds.

*See section 3 - Setting Gain, for more information on using PFL.*

**K Mute/Channel On-Off Switch**

This turns the channel on or off and is useful for isolating the channel when not in use or pre-setting channel levels which may not be needed until later, ie: theatre scene-setting or support acts/performers.

**L Fader**

This determines the level of the input signal within the mix and provides a visible indication of channel level.

**M Routing**

By selecting the routing switches the input signal is sent to a choice of the mixer's outputs - typically the main mix outs or the group outputs. The switches are used in conjunction with the PAN control to route the signal proportionately to the left or the right side of the mix or to odd/even groups/subs if PAN is turned fully left or right.

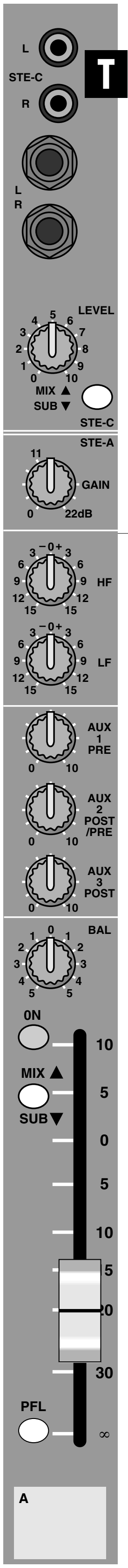



FIG. 1.4

## STEREO INPUTS

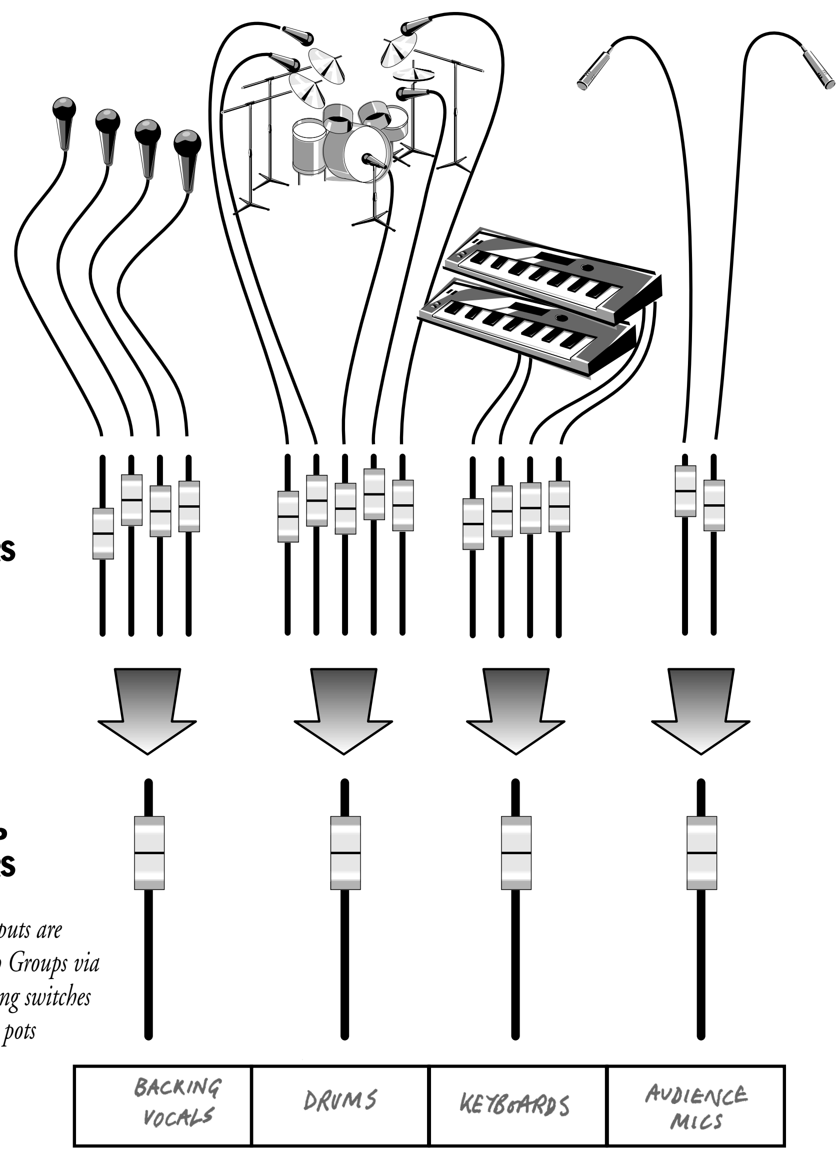
Guitar amps and mic'd sound sources only provide you with mono signals. However keyboards, samplers, drum machines and other electronic media often provide true stereo outputs with separate left and right signals. Stereo Inputs on mixers simply allow you to connect both of these signals individually and control them from a single fader. Stereo inputs tend to incorporate fewer facilities than mono inputs as most keyboards are already equipped with plenty of internal effects and tone control options.

*NB: Soundcraft jack stereo inputs default to Mono when the left input is used. RCA phono connectors do NOT have this option.*

## SUBGROUPS

These allow the logical assignment of groups of instruments or vocalists so that they may be controlled by just one pair of faders, or even a single fader, once individual instruments' relative levels have been balanced. They also act as additional outputs with separate volume/level controls – ideal for speaker fills or recording a number of instruments to one tape track.

FIG. 1.5




*Note: Inputs are routed to Groups via the routing switches and Pan pots*

## THE MASTER SECTION

**N Mix Outputs**
Mix outputs provide left and right level control of the final stereo mix. Many consoles feature mix insert points too, allowing the connection of signal processors across the whole mix.

**O Monitor "Engineer's" / Control Room Outputs**
These let you listen to any solo, mix, submix from a group, or the 2 Track tape return via an external amplifier and speakers, or the headphone socket.

**P 2 Track Tape Returns**
Allow you to connect the outputs of your cassette or DAT player and listen back to your completed masterwork. They may also be used for playing pre-show music at a gig using 2-Track to Mix switch (not shown in illustration).

**Q Aux Masters**
These govern the overall output levels from the auxiliary outputs and therefore the amount of signal going to an effects unit or a musician's foldback mix.

**R AFL**
Allows monitoring of the actual signal leaving the Aux Masters.

**S Meters**
Normally they show mix output levels. When any Solo button is pressed, the meters automatically switch to show the solo level. They provide visual indication of what's going on in your mixer.

**T Stereo Returns** (see *Stereo Inputs* earlier in this section)
These allow signals from external equipment, such as effects units, to be returned to the mixer and routed to the stereo Mix or Groups, without using up valuable input channels.

**U +48v or Phantom Power**
Some microphones, known as condenser mics, require battery power to operate. Alternatively the power may be provided by the console. This is known as 'phantom power' and runs at 48vDC. Simply press "Phantom Power" and any condenser mics connected will operate without the need for batteries.

> *Caution: DO NOT ACTIVATE A GLOBAL PHANTOM POWER SWITCH IF AN UNBALANCED SIGNAL SOURCES IS CONNECTED TO ANY MIC INPUT. Because of the voltage present on pins 2 and 3 of the XLR connector, you will damage your microphone/signal source.*
>
> *Always refer to your Mixer's User Guide.*

*More Information on Condenser Mics can be found in Section 3 - Mixing Techniques.*

*Further detail on mic wiring may be found in Section 7 - wiring.*

**V Headphones**
Allow you to listen to your mix without annoying your neighbours or being distracted by ambient sounds.

*That's it, the basic features of your average mixing console. If you found it a little heavy going, don't despair: it does get easier!*

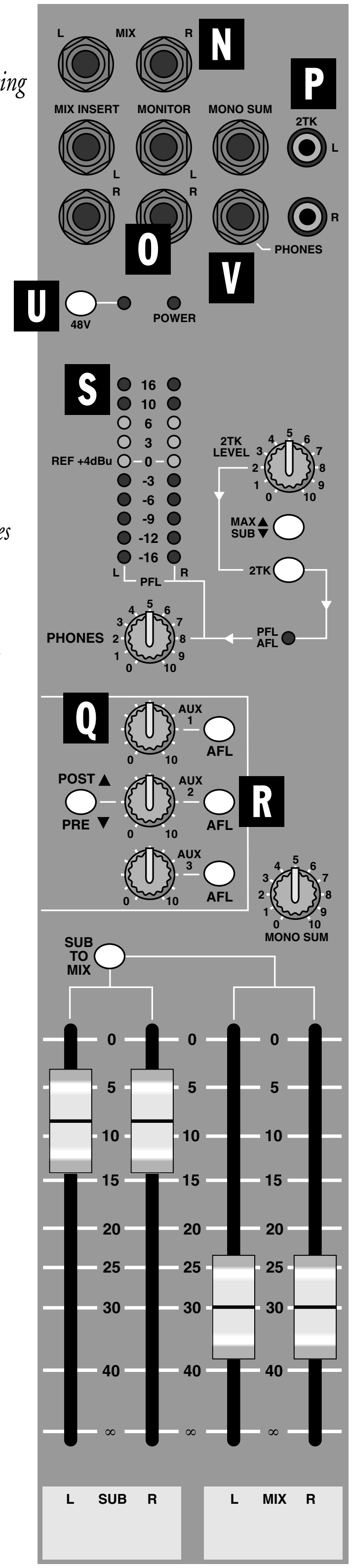



FIG. 1.5

## D. Signal Flow

Now the typical mixer features have been explained in detail it is important to understand how they form together. The route which a signal source takes through a mixer is normally shown using one of two devices: a **block diagram** or a **signal flow diagram**.

Both diagrams provide a 'visual' description of the key elements of the mixing console. They allow you to identify which components are used in the audio path and help the engineer to "troubleshoot" when signal sources don't appear to be doing what they should! In simple terms, they are electronic maps.

An example of a signal flow diagram is shown here. This is the most basic representation of console layout, showing a how a single sound source may pass through an input strip to the various other parts of the mixer.

Block diagrams are slightly more complex, showing more detail, electronic information, including the location of resistors and capacitors, and the structure of the entire console including bussing: **an example is shown on page 37**. Block diagrams also use a number of symbols to represent electronic elements. A few minutes spent understanding them some time during your journey through this booklet will most definitely pay-off in future mixing projects.

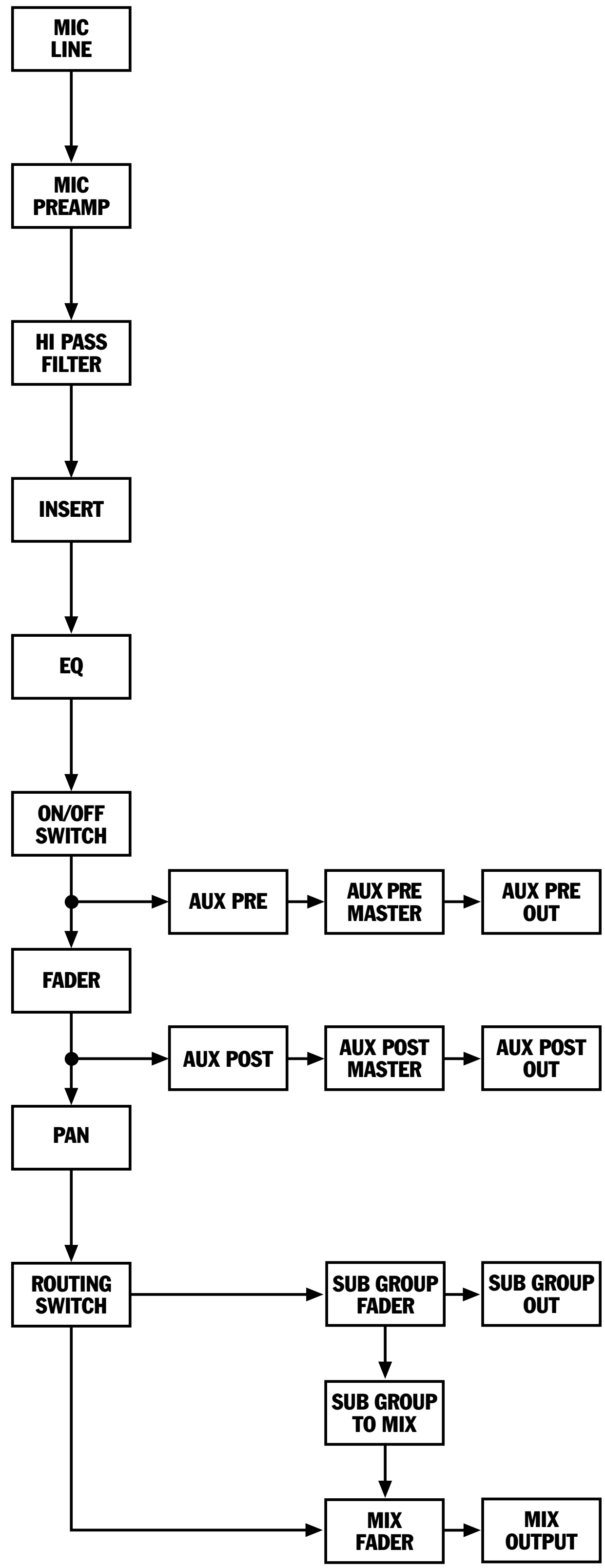



**FIG. 1.6**

**A Typical Signal Flow Path**

# CONNECTING EQUIPMENT TO YOUR MIXER

As we explained in the last section, it is the job of the mixer to accept the various signal sources, set the levels and route those signals to the correct destination.

We'll now take a quick look at where to connect the 'peripheral' equipment that you will be using with your mixer. If you have already created your own set-ups successfully in the past, you should only need to skim this part.

## A. Input Devices

### Microphones

All microphones should be connected via each input's XLR connectors. Do *not* use line inputs.

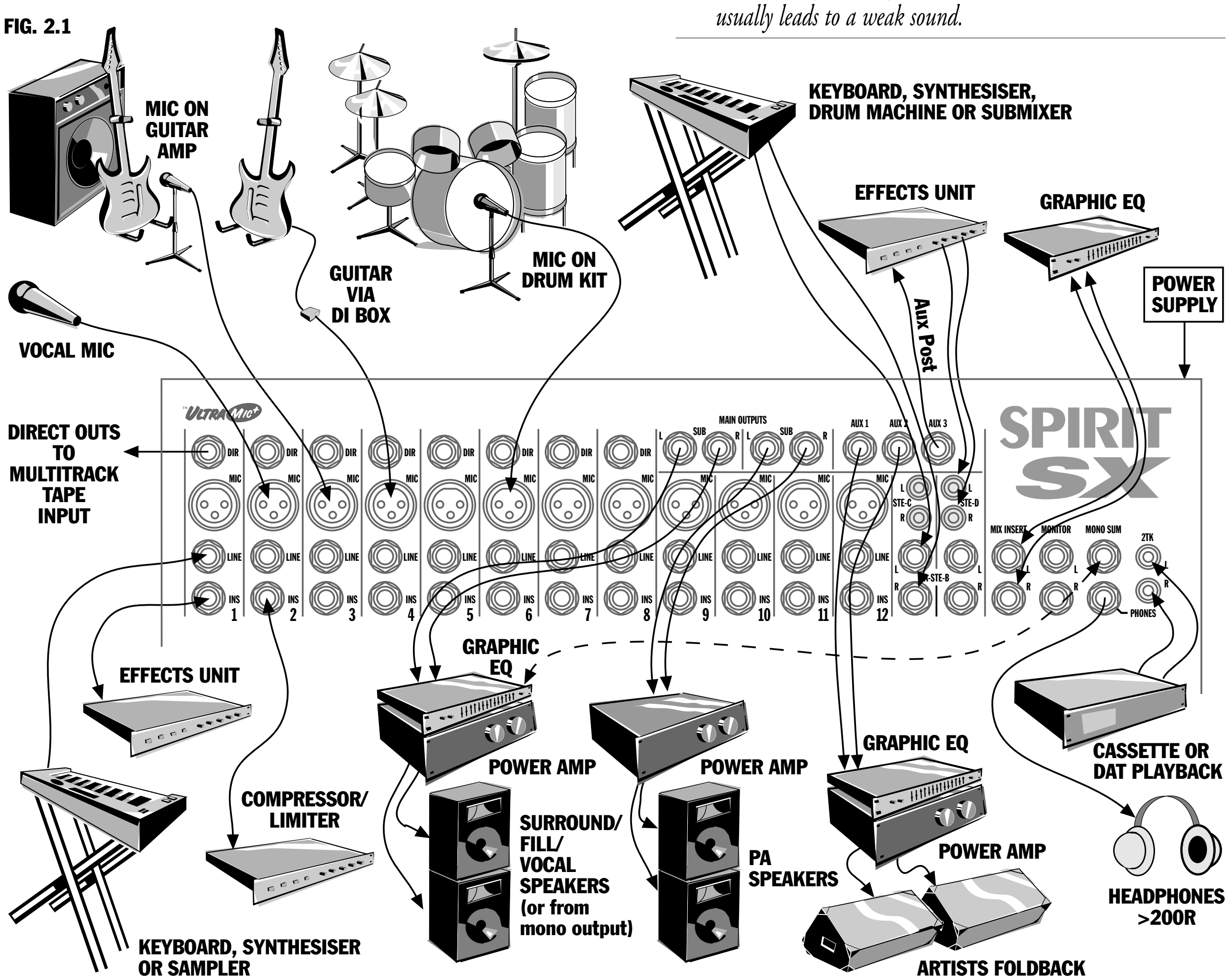
*For more information on miking up individual instruments, refer to sections 4 and 6 - PA Mixing and In the Studio.*

### Direct Injection Box (DI Box)

- A DI Box allows you to connect a guitar or bass directly to the mixer's input, rather than miking up the instrument's amp/speaker. This technique is often preferred by musicians who require a "clean" sound. The best DI boxes are ACTIVE and require Phantom Power like condenser microphones. They should be connected to XLR mic inputs.

*NB: Although electric guitars and basses may be connected to a mixer's line inputs without danger, the results will be far from ideal, because the IMPEDANCE of these instruments will not match up with typical line levels. Direct connection usually leads to a weak sound.*

FIG. 2.1



**Electronic Line Output Devices**

- Keyboards, Drum Machines, CD Players, DAT Machines, Wireless Mic Receivers, all provide line level outputs, and should all be connected straight into the Mixer's Line Inputs. If some of your instruments are STEREO connect their left and right outputs to a spare stereo input. Alternatively connect to an adjacent pair of mono inputs and Pan the inputs hard left and right to create a stereo image.

## B. Equipment Requiring Both Inputs and Outputs

**External effects units**

Connect the input of your effects unit marked "mono" to A POST FADER AUXILIARY OUTPUT. If you are uncertain, Post fader auxiliaries are coloured blue on Soundcraft mixers with the relevant channel aux pots usually marked "post". The left and right outputs from the effects unit should be connected to a pair of stereo returns, or stereo inputs if stereo returns are not available. If intensive EQ is required, use a pair of Mono Inputs. Remember, the effects signal is no different from any other audio signal – it still requires an input to the mixer.

*See Section 3 Mixing Techniques or a detailed explanation of post fader auxiliaries.*

*NB: YOU DO NOT HAVE TO CONNECT UP BOTH THE LEFT AND RIGHT INPUTS OF YOUR EFFECTS UNIT TO SEPARATE AUXs. Most units only require "pseudo-stereo" operation and will mimic a stereo reverb or effect inside before providing a stereo output to the mixer's returns.*

**Signal Processors**

Connect signal processors, such as compressors to the insert jack using a special insert 'Y' cable. This allows the signal to be sent and returned to the mixer using only one connector.

*Refer to section 7 for wiring information.*

It is also possible to connect the processor to the console without using the insert jacks by connecting an instrument direct to the processor first. However, the advantage of using processors in the mix/group or channel inserts is that any level changes made by the processor can be monitored by the mixers meters.

*NB: A signal processor can be used in a channel to control one audio source, across a group to control a number of audio sources or across the entire mix.*

**Tape machines**

Multitrack machines are used for initial track-laying in either studio or live recording situations.
For more sophisticated work, a stand-alone machine offers better sound quality and greater versatility than a cassette multitracker. The new generation of digital multitracks are also very attractive, but analogue, open-reel multitracks are also capable of professional sounding results. Aim for a minimum of eight tracks if your budget will allow.

**Mastering Machines**

Your final mix should be recorded on the best quality machine that you can afford. A recording is only as good as the weakest link in the chain, and a good cassette machine is fine for demos, but for more serious work, consider a DAT machine or perhaps a second hand, open-reel 2-track.

## C. Output Devices

### Amps and Speakers (Monitor and FOH)

**Studio Monitoring**

A high-powered hi-fi amp of around 50 watts per channel is fine for home recording, but to ensure adequate head-room you should consider a well-specified rack mount amp. Similarly, a pair of accurate hi-fi speakers will do the job, but for more serious work we would recommend purpose-designed nearfield monitors. Always remember that no matter how good the recording or performance, a poor monitoring set-up will not allow you to make qualitative judgements about the mix.

**Headphones**

When choosing headphones for monitoring, you'll obviously want a pair that give the best sound reproduction for the price. But, bear in mind that in order for you to fully concentrate on the mix, the headphones should exclude outside noise - therefore open-back designs will be of little use.

Furthermore, you could be wearing the headphones for several hours at a stretch so comfort is essential.

*NB: Make Sure that the IMPEDANCE of your headphones matches the specification of your mixer.*

**PA Work**

PA work requires high-powered, rugged, and honestly specified amps and FOH (Front of House) speakers. The power rating of the system will depend on the size of venues you will be playing.
See PA Mixing, Section 4, for more information.

# MIXING TECHNIQUES

## A. Choosing the Right Microphone

### Microphone Types

The choice of microphone depends on the application that the microphone will be used for and individual preference. However, broadly speaking microphones fall into two main types:

**Dynamic Microphone -**

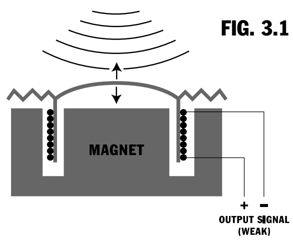



FIG. 3.1

• A robust design which uses a thin diaphragm attached to a coil of wire arranged about a permanent magnet. Any variation in air pressure on the diaphragm will cause the coil to generate a minute electric current which then requires amplification.

Dynamic mics are relatively inexpensive, rugged and require no electrical power to operate. They are ideal for all-round high sound pressure levels (SPL) and tend to be used for live applications. However, they are not as sensitive to high frequencies as condenser types.

**Condenser Microphone -**

• A type of microphone which picks up sound via a thin, flexible diaphragm placed in proximity to a metal plate - as opposed to the rigid diaphragm-and-coil system used by dynamic microphones. They need power to operate - the most common source being +48v DC PHANTOM POWER.

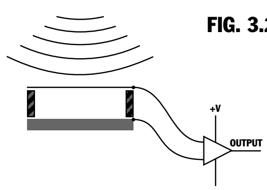



FIG. 3.2

Condenser mics are very sensitive to distant sounds and high frequencies. Because of this sensitivity they are often used in studio recording situations.

*N.B. +48v Phantom power is used to charge the diaphragm and plate. It also supplies a small amplifier which boosts the small voltages generated by diaphragm movements.*

### Microphone Pick-up Patterns

A pick-up (Polar) pattern refers to the area(s) from which a microphone "picks up" its sound. It is important to choose the right pattern for your application, or you may pick up sounds from areas you don't want or lose sound information you need.

**Omni Pattern**

The most basic type of microphone pattern.

FIG. 3.3

• A 360° polar response which picks up sound equally in all directions.
  This pattern is ideal for picking up groups of vocals, audiences, ambient sounds but is most susceptible to feedback.

**Cardioid Pattern**

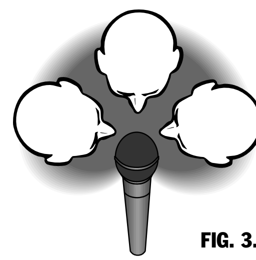


FIG. 3.4

• The 'heart-shaped' polar response of a microphone meaning that most of the sound is picked up from the front.
  Used for most basic recording or in any situation where sound has to be picked up from mainly one direction. Dynamic cardioid mics are mostly used for live applications because they help reduce unwanted spill from other instruments, thus reducing the risk of feedback.

**Hyper-cardioid**

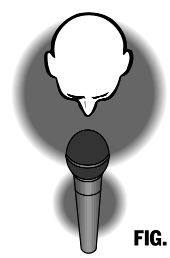


FIG. 3.5

• Similar to a cardioid pattern but with greater directionality.
  Used for live vocal microphones because it provides the greatest protection from unwanted spill and feedback.

**Figure of Eight**

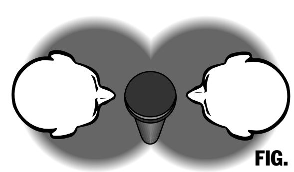


FIG. 3.6

• Sound is picked up from the front and back but not from the sides.
  This pattern is used mainly in studios for picking up two 'harmony' vocalists, or solo vocalists who require some room ambience.

## B. Setting Up a Basic Mix

### Setting the Gain

Input gain is designed to take an audio signal, and adjust it to the level which the mixer understands.

All audio circuits, mixers included, produce a low level of electronic noise or hiss, and while this can be made very low by careful design, it can never be completely eliminated. It is also true that any audio circuit can be driven into distortion if the input is too high in level; hence care has to be taken when setting the input level so as to preserve the best possible sound quality. Ideally the input signal should be as high in level as possible while still leaving a margin of safety to prevent distortion on loud sections. This will ensure that the signal is large enough to render the background noise insignificant, whilst keeping the signal clean. The remaining safety margin is known as *Headroom*.

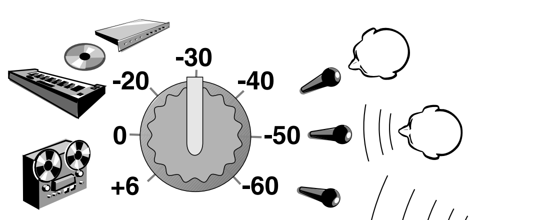



**FIG. 3.7**
**TYPICAL GAIN SETTINGS FOR DIFFERENT INPUTS**

To set the gain on the mixer;

- Press the PFL/Solo switch on the relevant input.
- Adjust gain/input sensitivity until meters read within the yellow (‘3’ to ‘6’ on meter scale). This allows for the extra 10dB of gain that is available on Soundcraft input faders.
- Release PFL/Solo.
- Repeat for all other inputs.

*NB: EQ affects gains settings. If you adjust the EQ you will need to re-check your gain level using the above method.*

Once you have optimised the gain your mixer will give the best possible signal quality with the minimum of noise and distortion.

### Balancing Fader Levels

Faders allow you to make fine adjustments to your sounds and act as a visual indication of the overall mix levels.

It is important to keep your input faders around the ‘0’ mark for greater control. This is because fader scales are typically logarithmic and not linear, so if your fader position is near the bottom of its travel then even a small movement will lead to huge leaps in level. Similarly try not to have your fader at the top of its travel because this will leave you no room to further boost the signal.

*See diagram below.*

### Balancing Output Levels

**Master Outputs**

Set your master outputs to ‘0’ on the scale. There are two reasons for this:

1 You have the maximum fader travel for fading out your mix.
2 If your faders are set below ‘0’ you will not be getting the full benefit from the meters because you will only be using the first few LEDs on the meter scale.

*NB: Your mixer is not an amplifier. So the master output faders should be set to maximum (‘0’ on scale). If extra output is required, then turn up your amplifier.*

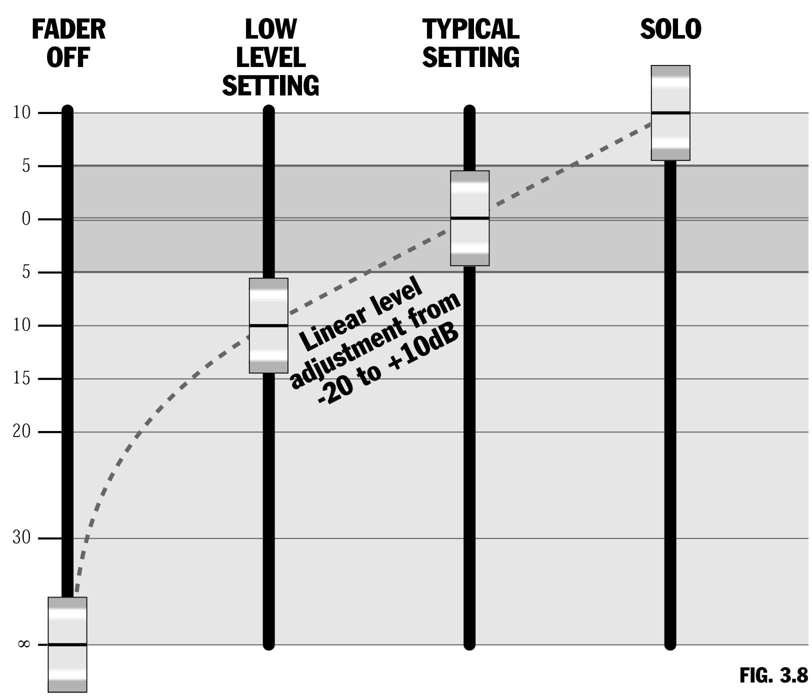



**FIG. 3.8**

## C Using the Mixer's EQ

Equalisation is useful for making both corrective and creative changes to a sound, but it needs to be used with care. Corrective applications include making tonal changes to compensate for imperfect room acoustics, budget microphones or inaccurate loudspeaker systems. While every effort should be made to get the sound right at source, this is less easily achieved live than in the more controlled conditions of the recording studio. Indeed, the use of equalisation is often the only way to reach a workable compromise in live situations.

Creative applications, on the other hand, are equally as valid in the recording studio as they are live, and an equaliser with a swept midrange control is infinitely more versatile than one that has simple high and low controls. The only rule of creative equalisation is - 'If it sounds good, it is good!'

### Fixed EQ

Most people will be familiar with the operation of high and low frequency controls; they work in a similar manner to the tone controls on a domestic stereo system.

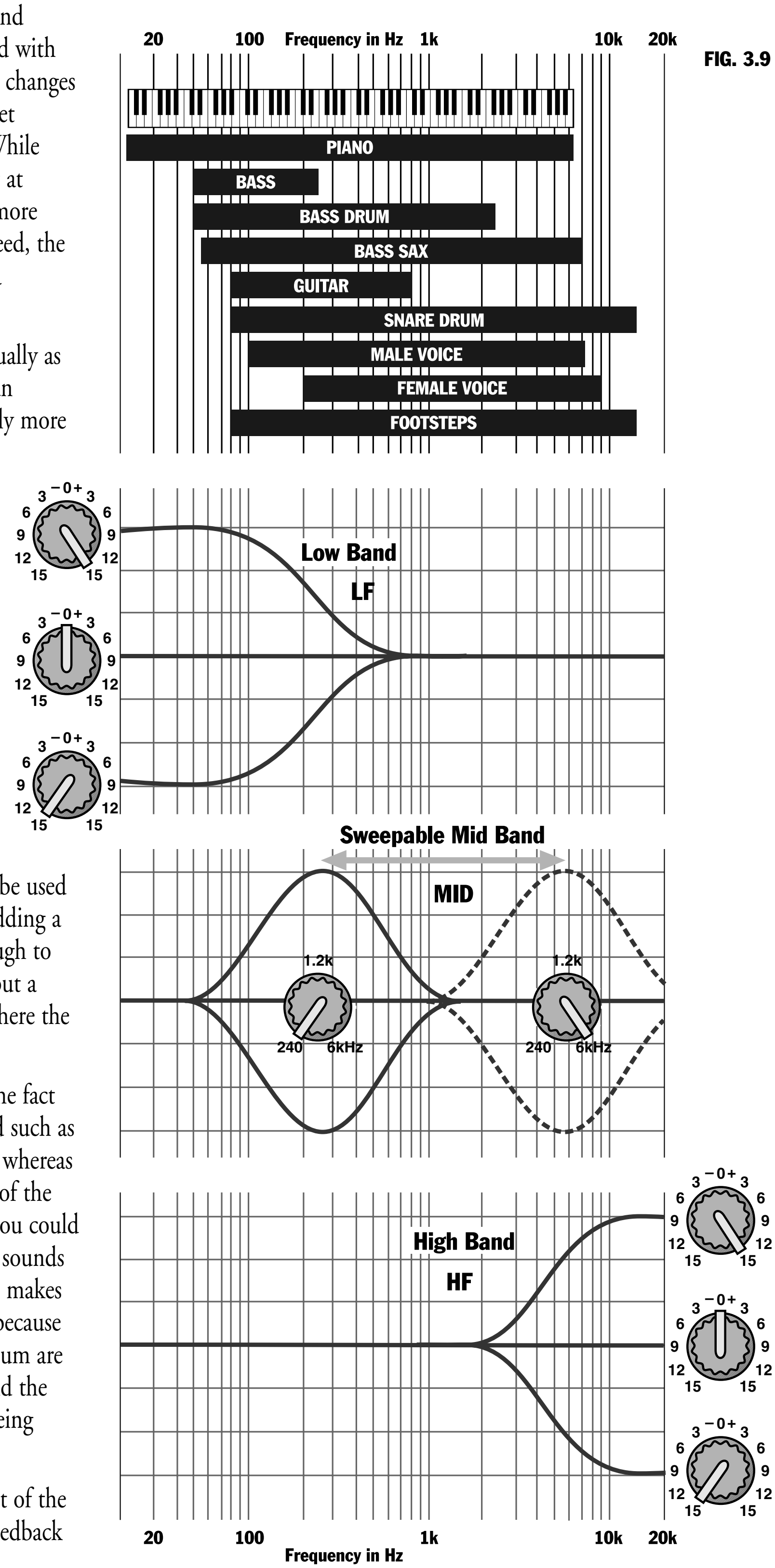
In the centre position the controls have no effect, but rotate them clockwise and they will provide boost, or rotate them anticlockwise and they provide cut. Despite their apparent simplicity, however, high and low controls should be used with caution as overuse can make things worse. Adding a small amount of high or low boost should be enough to add a touch of brightness or warmth to a sound, but a quarter of a turn should be sufficient, especially where the low control is concerned.

The drawback with fixed controls often lies in the fact that you may want to boost just a particular sound such as the punch of a bass drum or the ring of a cymbal, whereas a fixed control influences a relatively large section of the audio spectrum. Apply too much bass boost and you could find the bass guitar, bass drum and any other bass sounds take on a flabby, uncontrolled characteristic which makes the mix sound muddy and badly defined. This is because sounds occupying the lower mid part of the spectrum are also affected. Similarly, use too much top boost and the sound becomes edgy with any noise or tape hiss being emphasised quite considerably.

In a PA situation, excessive EQ boost in any part of the audio spectrum will increase the risk of acoustic feedback via the vocal microphones.

**THE FREQUENCY RANGE OF DIFFERENT INSTRUMENTS AND WHICH EQ BANDS AFFECT THEM**

FIG. 3.9



Bearing the above points in mind, the best approach is to use small amounts of boost, especially when working live. EQ cut, on the other hand, causes far fewer problems, and *rather than boost a particular sound it is frequently more rewarding to apply cut in whichever part of the audio spectrum that appears to be overpowering*. In this application, the sweep mid control is also very effective.

**Using a sweep-mid equaliser**

Like the high and low controls, the sweep mid can provide either cut or boost, but its strength comes from the fact that it can be 'tuned' into the specific part of the audio spectrum that needs treatment. Like the high and low controls, it is more forgiving if used to cut rather than to boost. However, when first tuning in the mid control, it helps to set it to full boost, so that when the frequency control is adjusted, the effect is most apparent. This is true even if the final EQ setting requires cut rather than boost.

**Procedure**

Below is a simple way of eliminating unwanted sounds:

> *Caution: when adjusting EQ, there is a danger of feedback which can cause damage to your speakers. You may need to reduce your levels to compensate.*

- Increase sweep-EQ gain.
- Sweep the frequency pot until the aspect of the sound you wish to modify becomes as pronounced as possible. This should only take a few seconds.
- The cut/boost control is now changed from its full boost position to cut. The exact amount of cut required can be decided by listening to the sound while making adjustments.
- Even a small amount of cut at the right frequency will clean up the sound to a surprising degree.

Other sounds may benefit from a little boost, one example being the electric guitar which often needs a little extra bite to help it cut through the mix. Again, turn to full boost and use the frequency control to pick out the area where the sound needs help. Then it's a simple matter of turning the boost down to a more modest level and assessing the results by ear.

## D. Using Effects Units

### The Different Types

The problem with mixing 'dry' (using no effects) within a live or recording environment is that the results can often sound boring and lacking in colour. This is especially the case as most of us are used to listening to highly polished CDs at home. These productions are actually achieved by using effects which electronically produce certain atmospheres. The different types of effects that can be used are explained below;

**Reverb**

Reverberation is the most commonly used studio effect, and also the most necessary. Western music is invariably performed indoors where a degree of room reverberation is part of the sound. Conversely, most pop music is recorded in a relatively small, dry-sounding studio, so artificial reverberation has to be added to create a sense of space and reality. Reverberation is created naturally when a sound is reflected and re-reflected from the surfaces within a room, hall or other large structure. See fig. 10.

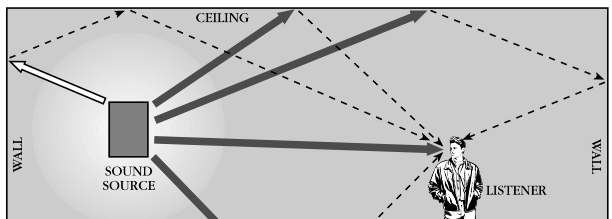



**FIG. 3.10 REVERB**

**Delay**

Often used to make a sound 'thicker' by taking the original sound, delaying it, then mixing it back with the original sound. This short delay added to the original sound has the effect of doubling the signal.

**Echo**

A popular effect that was used extensively on guitars and vocals in the 60s and 70s. It is not used on vocals so much nowadays, but quite effective on guitars and keyboards. A neat trick is to set the echo delay time so that the repeats coincide with the tempo of the song.

**Chorus & Flanging**

Both chorus and flangers are based on a short delay, combined with pitch modulation to create the effect of two or more instruments playing the same part. Flanging also employs feedback and is a much stronger effect. Both these treatments work well on synth pad sounds such as strings and are best used in stereo where they create a sense of movement as well as width.

### Pitch Shifters

These change the pitch of the original signal, usually by up to one octave in either direction and sometimes by two. Small pitch shifts are useful for creating de-tuning or doubling effects. Which can make a single voice or instrument sound like two or three, while larger shifts can be used to create octaves or parallel harmonies.

*NB: For useful effects settings with different instruments refer to Section 6 'In the Studio'.*

## Setting up an effects loop

- Set effect unit to full 'wet' signal
- Connect your effect units as per Section 2, Input Devices.
- On the relevant input channel, set the post fade aux to maximum
- Select AFL on your aux master
- Set aux master level so that the meters read '0'
- Adjust input level on effects unit until 'effects meters reads '0' (nominal)

*NB: You can now use the mixer AFL meters to monitor effects unit levels as both meters have been calibrated.*

- Release aux master AFL and select effects return PFL

*NB: If you are using a simple stereo input with no PFL, adjust input gain for required effect.*

- Adjust effects return input gain until meters read around '0'.
- De-select PFL and adjust effects return fader level for required effect level.

*NB: The original 'dry' signal is mixed with the effects 'wet' signal.*

## Pre- and Post-fade Auxiliaries

### Pre-Fade

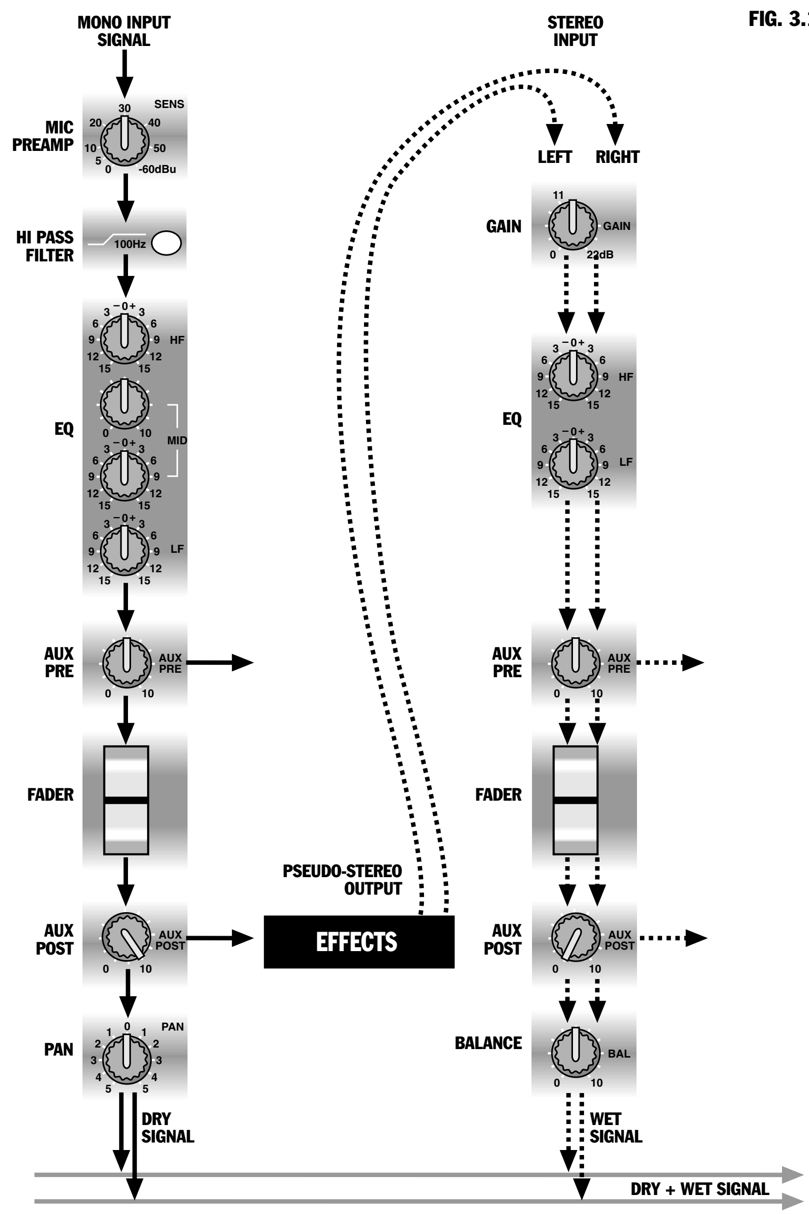
Pre-fade auxiliaries are independent of the fader so that the amount of effect will not change with new fader levels. This means you will still hear the effect even when the fader is at the bottom of its travel.

### Post-Fade

It is important to use post fade auxiliary sends for effects units. This is because post fade auxiliaries 'follow' the input fader so that when input level changes the amount of effect remains proportional to the new input level.

*NB: Effects Return Aux Post Control must be set to minimum or feedback will occur*

FIG. 3.11



## E. Using Signal Processors

### The Difference between Signal Processors and Effects

Unlike effects, which are creative in nature, signal processors are used to control and manipulate sounds to achieve the best audio quality performances and recordings.

Effects and signal processors should never be confused. Whereas effects are “mixed” with an input to provide a combined sound, signal processors alter an input, group or mix signal completely. The signal is actually taken out of the mixer entirely, “processed” and returned in its altered state, in series with the original audio signal.

For this reason signal processors should be connected using Insert Points and not the Auxiliary Send and Return Loop (effects loop).

*NB: Effects can be connected to inserts if necessary, but then the proportion of the effect in the signal is governed solely by the effects unit mix control.*

### The Different Types of Signal Processors

Broadly speaking, there are 5 different types of signal processor in common use:

#### Graphic Equalisers

Graphic Equalisers work by splitting the sound spectrum into narrow, adjacent frequency bands and giving each band its own cut/boost slider. The term Graphic comes about because the position or ‘curve’ of the sliders gives a graphic representation of the way in which the settings affect the audio frequency range.

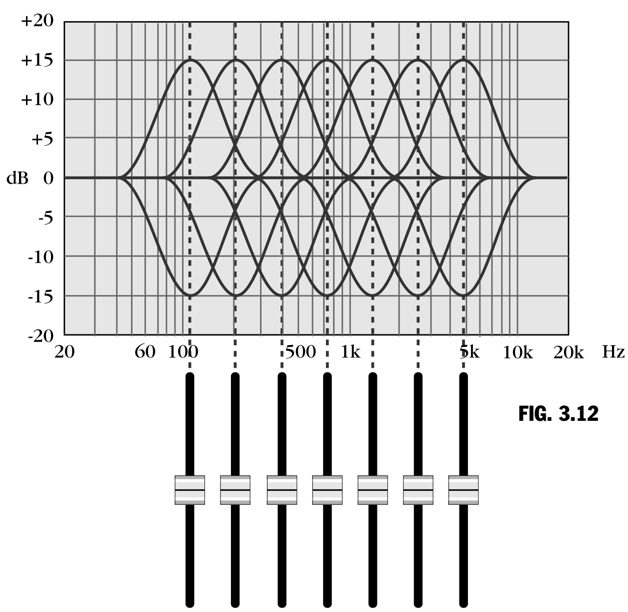



**FIG. 3.12**

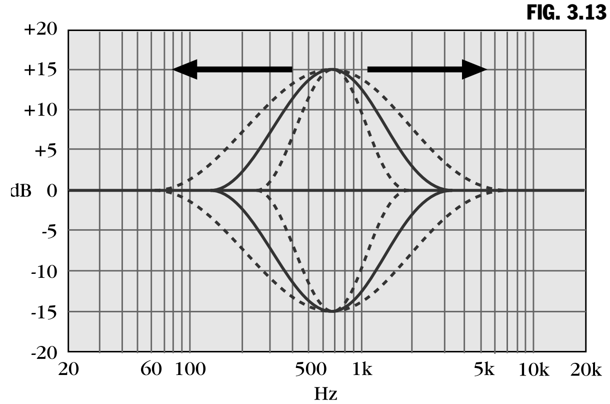
Graphic Equalisers are most often used to process the mix at live venues by notching out troublesome frequencies that may be causing feedback. They may also be used to enhance a mix at a poor sounding venue. In recording they are used to create "flat" listening environments.

*For more detail on venue acoustics go to section 4 - PA Mixing.*

#### Parametric Equalisers

These are similar to the EQ found on an input channel but may include more bands and additional bandwidth (Q) controls which define how many frequencies in the band are affected.

**FIG. 3.13**




They are most often used to provide additional creative control over an input signal when a mixer’s EQ is not sufficient.

#### Gates

A gate is designed to shut down the audio signal path when the input signal falls below a threshold set by the user. It may be used to clean-up any signal that has pauses in it. For example gates are widely used to prevent ‘spill’ between adjacent mics on a multi-mic’d drum kit where, say, a tom-tom mic may pick up the snare drum.

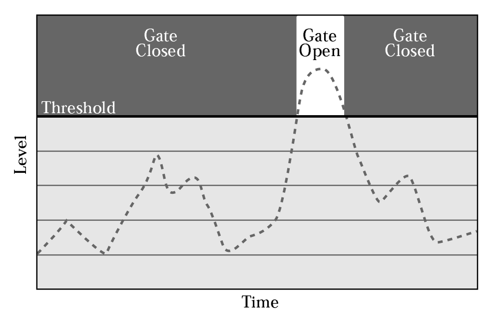



**FIG. 3.14**

#### Expanders

Expanders accomplish much the same task as gates, though they are more like compressors in reverse. Compressors affect the gain of signals exceeding the threshold, while expanders act on signals falling below the threshold. A gate will close completely when the signal falls below its threshold, but an expander works like an automatic mixing engineer who pulls down the signal when the signal falls below the threshold; the more it falls below the threshold - the more he pulls down the fader.

Expanders are most often use in Studio recording to provide the best mix signal to noise ratio when producing final masters.

**Compressor/Limiters**

A compressor reduces the difference between the loudest and quietest parts of a performance. It works on a threshold system where signals exceeding the threshold are processed and those falling below it pass through unchanged. When a signal exceeds the threshold the compressor automatically reduces the gain. How much gain reduction is applied depends on the 'compression ratio' which on most compressors is variable: the higher the ratio, the stronger the compression. Very high ratios cause the compressor to act as a limiter where the input signal is prevented from ever exceeding the threshold.

Compressors are the most commonly used processor and are particularly popular for maintaining constant vocal and bass guitar levels live and in the studio. This is because, out of all instruments, singers tend to vary their levels the most. Compressors help to achieve the much sought-after tight, "punchy" sound.

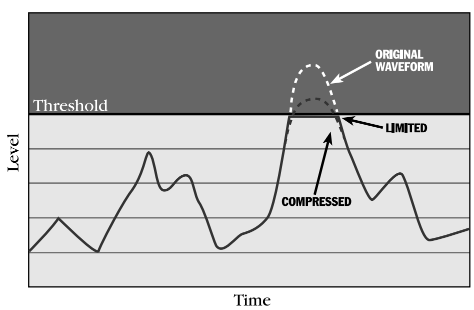



**FIG. 3.15**

## Setting up a Signal Processor

- Connect your processor to the relevant mixer insert jack (mono, group or mix insert), using a insert 'Y' lead.

*Refer to section 7 for wiring information*

- Set your processor to unity gain (x1), i.e. no additional gain.
- Make your adjustments on your signal processor
- Beware that your processor settings may alter your mixer input output levels. Re-adjust levels to '0' on meters, if necessary.

*NB: Remember a signal processor can be used in a channel to control one audio source, in a group to control a number of audio sources, or to control the entire mix.*

## F. Creating a Foldback/ Monitor Mix

Performers usually require their own mix independent from the main/engineer's mix. This is because to achieve the optimum performance they need to hear themselves above other voices or instruments. This performer's mix is known as a foldback/monitor mix.

The procedure is as follows;

- Set the **pre-fade** aux to maximum on the relevant performers input channel.
- Select AFL on your aux master.
- Set aux master level so meters read '0'.
- Create a foldback mix for the performer by setting the pre-fade aux levels on the other performer's input channels.
- Release aux master AFL.

*NB: It is typical that the performers' own vocals/instruments will be two thirds louder than any other sources in their own monitor mix.*

Each performer may require a separate monitor mix/auxiliary output.

*NOTE: Pre-fade rather than post-fade auxiliaries must be used. This is because they are independent of the input faders. If post-fade auxiliaries are used, then foldback mix levels will alter with every input fader change made by the FOH engineer. This will annoy the band and may lead to feedback which can damage speakers and headphones.*

**Now that you know how to connect and set up different elements of your system let's look at some real-world examples of systems in use.**

# PA MIXING

## A. A Typical Live Performance

### Introduction

There are so many different types of 'live' scenarios that it would be almost impossible for us to describe each one in a book of these modest proportions. Instead, our 'typical live gig' is represented by a small band, whose set-up is shown in the "Mixing Live" diagram.

### Microphones

Most of the microphones used in live applications are dynamic cardioids because they are tough, produce an intelligible sound and their directional response helps prevent spill or feedback. Dynamic microphones can handle anything from drums to vocals. However, condenser types, with their greater sensitivity to high frequencies are invariably used for jobs such as overhead pick-up on a drum kit or mic'ing acoustic instruments.

### Cables and Connections

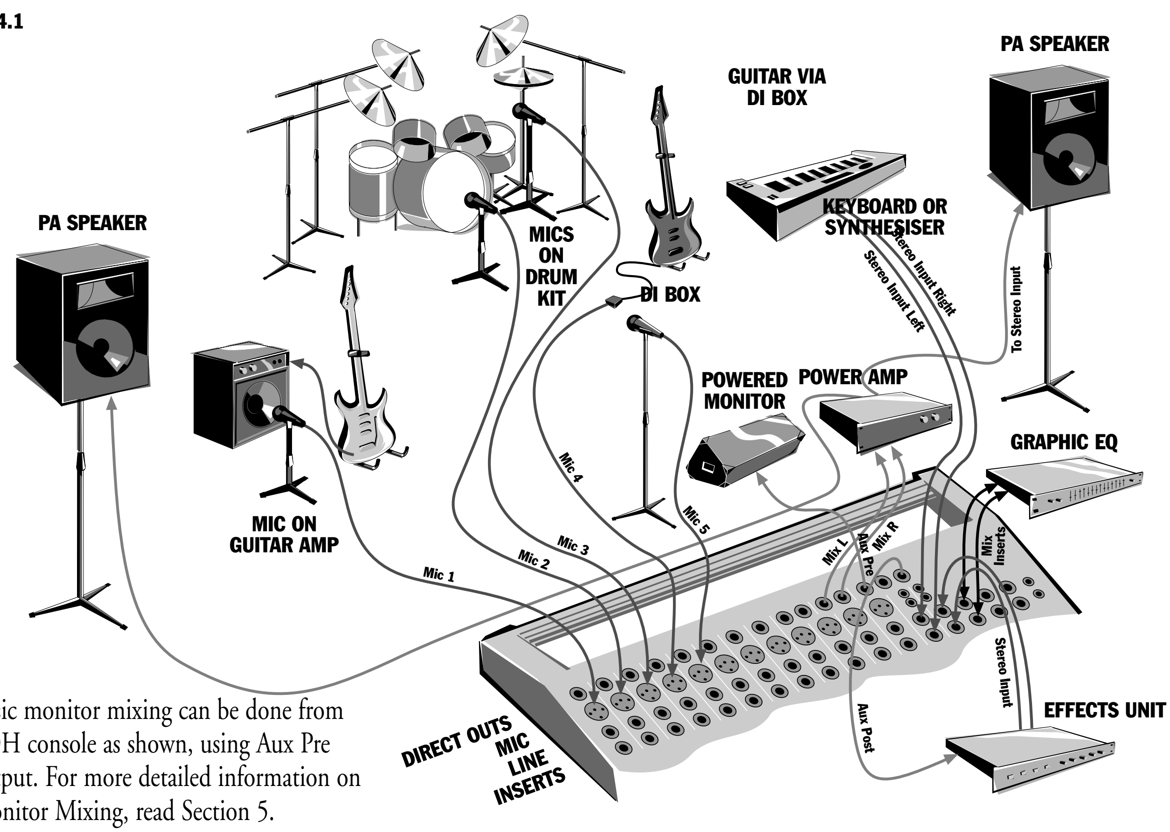
Interference and hum can be avoided! A few minutes spent checking cable runs and connectors pays dividends.

- A balanced audio connection provides low noise operation by cancelling out any interference in a signal. It does this by using a 2-conductor mic cable surrounded by a shield. Any interference picked up will be of the same polarity on the two conductors and is therefore rejected by the mic input's Differential Amplifier.
- Don't skimp on interconnecting cables - always buy the best that you can afford. Make sure that all connections are sound and keep cable runs as short as is practicable.
- A multicore cable and stage box will keep trailing cables to a minimum and presents a tidy and practical approach.
- If your mixer has a separate power supply unit, keep it well away from the console.
- Where signal and mains cables must cross, make sure they're at 90° to each other. This will help reduce the risk of hum and noise.
- If the venue has a three-phase supply, don't share the same phase as lighting controllers.



## MIXING LIVE

FIG. 4.1




- Basic monitor mixing can be done from FOH console as shown, using Aux Pre output. For more detailed information on Monitor Mixing, read Section 5.

- It is dangerous to lift the mains earth when trying to eliminate hum. You can isolate hum by lifting the appropriate audio signal shield.
- When using wireless mics, set the receiver on stage and run back to the console at balanced mic level. This will help avoid interference from digital sources and lighting controllers.
- Keep unbalanced 'insert' leads away from mains and keep them short - no longer than about 2 metres.

## Connecting External Effects and Processors

We talked about Effects and Processors in Sections 2 and 3, so you're now aware of their functions and applications. Effects units are best connected via the console's Auxiliary Send and Return Loop (sometimes known as the Effects Send and Return Loop) or the Insert Point. When used in the Aux Send system, the dry signal level should be turned off on the effects unit, but when used via Insert Points (for guidance on how to wire a jack for use with Insert Points, see Section 6), the dry/effects balance must be set on the effects unit itself. Processors treat the whole of the incoming signal and therefore may only be used via console Insert Points or directly 'in-line' with a signal: they cannot be used in the Aux Send/Return loop system.

## Setting Up

- Position the mixing console so that you can hear the on-stage performance as the audience will hear it. Ensure that you have a clear view of the performers.
- After setting up, switch the power amps on last to prevent any thumps occurring when effects or instruments are powered up. Ensure the console's master gain is down before you switch on the amplifiers.
- Don't set up the vocal mic directly in front of the drum kit or a guitar stack.
- Make sure the speakers aren't obstructed by the audience and that the majority of the sound is being directed towards the audience, not towards the rear or side walls.
- Set up the vocal levels first - it's no use getting a great drum sound if the vocals feed back before they can even be heard.
- Keep the vocals panned towards the centre of the mix. Not only will this sound more natural, but it will allow the greatest vocal level before feedback or distortion occurs.
- Be sparing on the use of artificial reverb. Most venues are too reverberant anyway, and excessive reverb will ruin the intelligibility of the vocal performance.
- Do not use reverb on low frequency sound sources such as bass, kick drums and tom toms.
- Keep backline amp levels down: let the mic and mixer do the work!
- Always leave a little gain in hand so you can wind up the level slightly as the show progresses.
- Putting high levels of bass guitar or kick drum through a small PA can overload the system and distort vocal quality. Try rolling off some of the low bass, you'll get a higher subjective sound level without overload.

## Ringing Out: Nulling Room Acoustics

> **!** *Caution: Ringing out can cause howl around which can damage speakers, so use care when adjusting levels.*

As experienced engineers will tell you, there's no such thing as a perfect venue. To help tailor the sound to the room acoustics, insert a Graphic Equalizer into the console's mix insert jacks which are effectively between the mixer and the power amp.

'Ringing Out' the system prior to the sound check will help reduce troublesome feedback. To Ring Out, follow this procedure:

1 Set all graphic EQ controls to centre (0).

2 Turn up amp volume until feedback is just beginning to 'ring'.

3 Turn back the amp volume slightly to prevent accidental feedback.

4 Starting from the left, adjust the first graphic EQ frequency gain control to 'max': if the system doesn't feedback, then this is not a problem frequency. Return this gain control to centre position.
If the system feeds back, reduce the EQ gain by the same amount you boosted to get feedback.

5 Repeat this procedure for all graphic EQ frequencies.

## Setting the Mix

- Turn down the amplifier gain before the system is first switched on. This will avoid unwelcome howls of feedback and can prevent loudspeaker damage due to switch-on transients.
- Set all the channel EQs to their flat or neutral position and optimize the input gain control setting for each channel in turn using PFLs.
- If low frequency background noise is a problem, switch in the High Pass Filter on each of the microphone channels being used, except on low frequency sound sources such as basses and kick drums.
- Ring out the system as described above, with the vocal mics open, and notch out any obvious trouble spots.
- Establish the maximum working level for the lead vocal mic so as not to incur feedback and then work a little below this level to allow a margin of safety. Again, see the notes on ringing out the system.
- Set up the backing vocal mics and check that there is no feedback problem when both the backing vocal and lead vocal mics are on. If there is, reduce the master gain setting until the feedback disappears.
- Now the instrument and direct line inputs can be balanced relative to the vocals. Start with drums and work through to the bass and rythm instruments.
- Test out any effects units connected to the system and establish the correct balance of dry and effected sound.

## Avoiding Feedback

- Turn down or mute any mics not in use. This reduces the risk of feedback and avoids the back line being picked up.
- If feedback is a real problem, consider moving the main PA speakers away from the mics a little. Also check the back of the stage, because if the wall is acoustically reflective, some sound from the room will be reflected back into the mics increasing the risk of feedback.
- Avoid excessive use of boosted EQ as this can encourage feedback and may also spoil the basic character of the sound. Consider it an aid to fine tuning rather than as a means of making radical changes.
- The use of stage monitors will also worsen the feedback situation so run these at the lowest volume that the performers can comfortably work with. Position the cabinets so as to allow as little direct sound as possible to enter the vocal microphones. If possible, use a graphic EQ on each monitor.

*NB: Remember, people soak up sound! The perfect mix achieved in an empty venue will have to be tweaked when the crowds arrive. Sound waves are also affected by heat and humidity.*

## B. Larger Performances

Although the example shown in the 'Mixing Live' diagram shown at the beginning of this section is of a small band, the principles are the same no matter the size of the live performance or venue. However, for larger PAs additional speakers, monitors, effects and processors may be required as well as slightly different positioning for each of these pieces of equipment. These additional requirements are outlined below:

### Medium Sized Venues

The console used will require more input channels. For example it is likely you will want to mic up all of the drums, and there are also likely to be more instruments, backing singers and sound sources in general.

More monitor sends will also be required - a single monitor will not be enough for larger bands. The bass and drums will require a monitor between them. The vocalists will want a monitor each so they can hear themselves above the band.

More speaker outputs may be needed in larger venues so that all the audience can be reached, without there being "holes" in the amplified audio signal. It may be necessary to record the event. This will require additional level controlled stereo outputs or direct outs if a multitrack is being used.

*NB: For simplicity, these diagrams do NOT show any outboard equipment.*

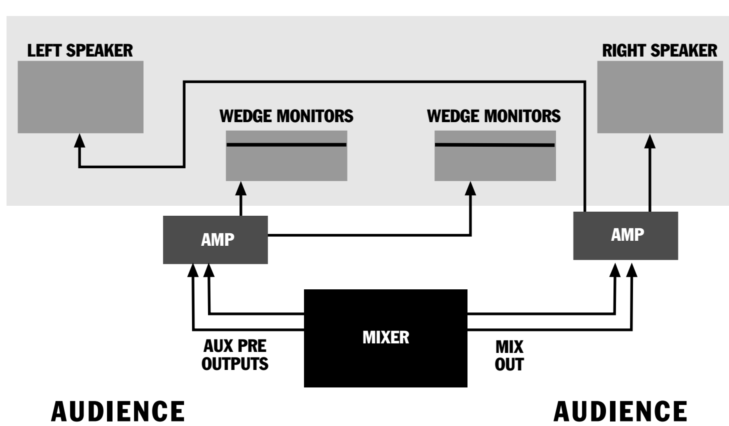
### Large Sized Venues

Large venues will require a separate "Front of House" (FOH) console for the audience mix and a Monitor console for the band, as with a larger stage area each band member will require at least one monitor wedge. The auxiliary send system of the FOH console will not be able to cope with these demands alone as it will have to deal with several effects units.

The FOH console will have a large number of mic/line inputs, plus a large number of matrix outputs so that a complex range of speaker clusters can be placed around the auditorium.

## SMALL VENUES

FIG. 4.2



## MEDIUM SIZED VENUES

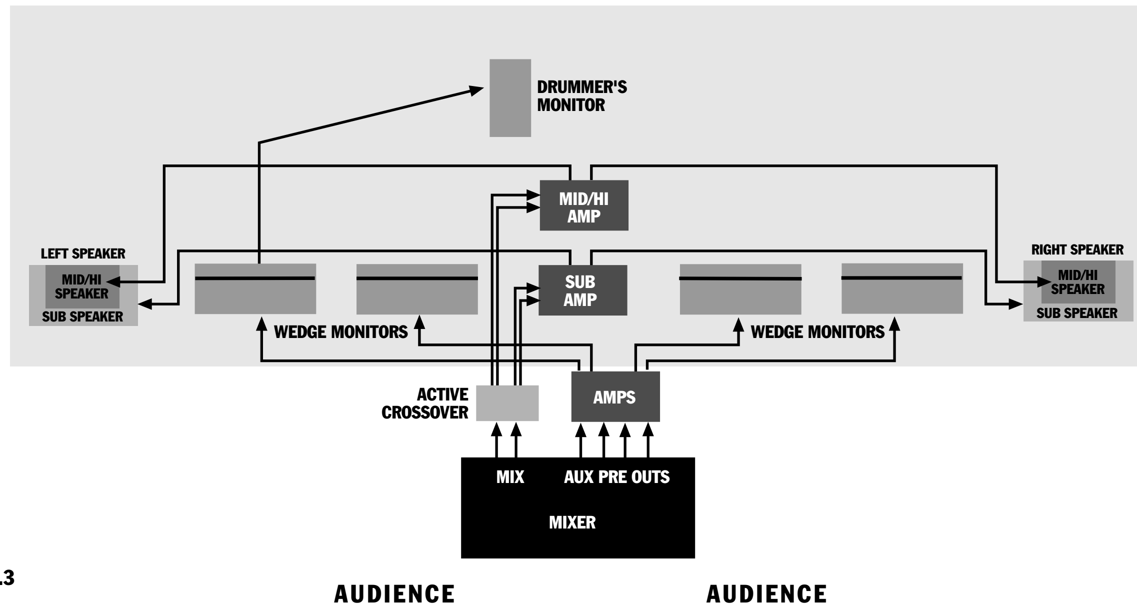



FIG. 4.3

## LARGER VENUES

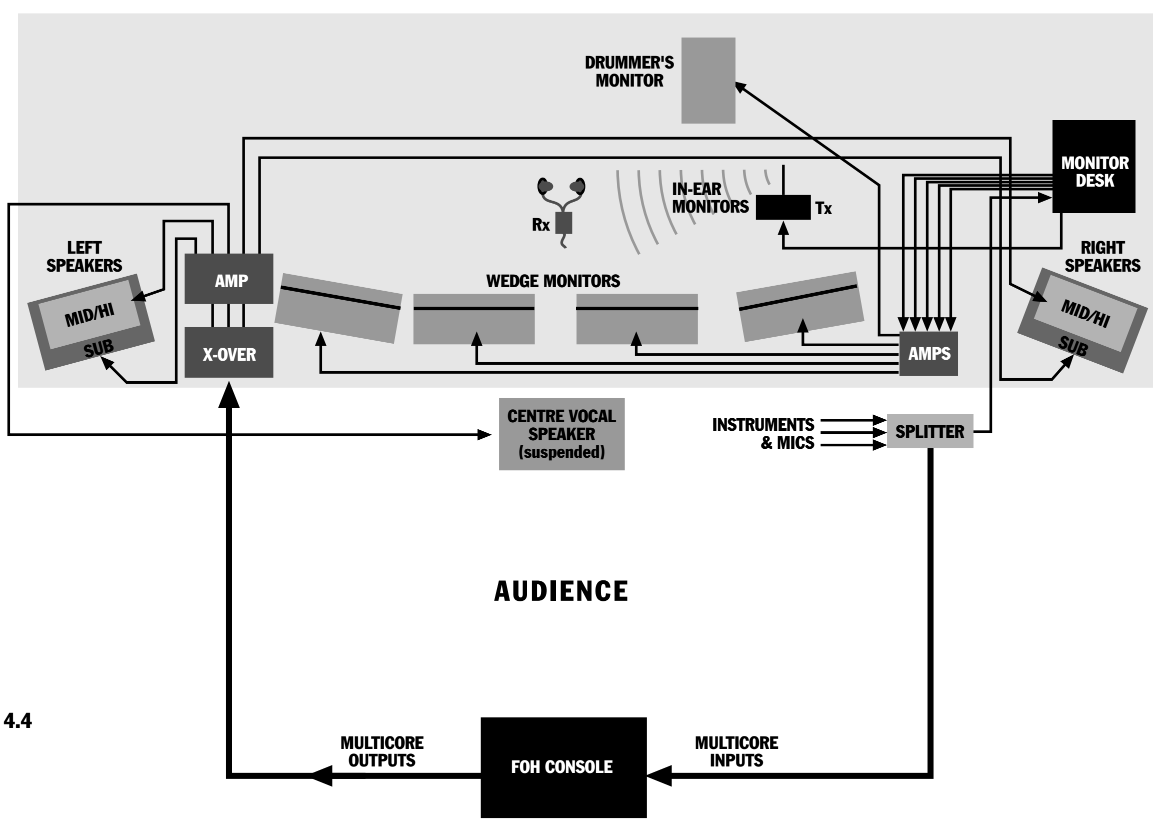



FIG. 4.4

## C. Recording Live

In some situations, you may want to record a performance. Depending on the situation, the feed for recording may come from the FOH mixer, microphone splitter boxes, or your own microphones which have been set up alongside those of the band.

The diagram below shows a typical example of the sound sources being split between FOH and Recording. The recording console operates independently from the FOH mixer.

*NB: When using Folio SX it will be necessary to re-patch for multitrack playback.*

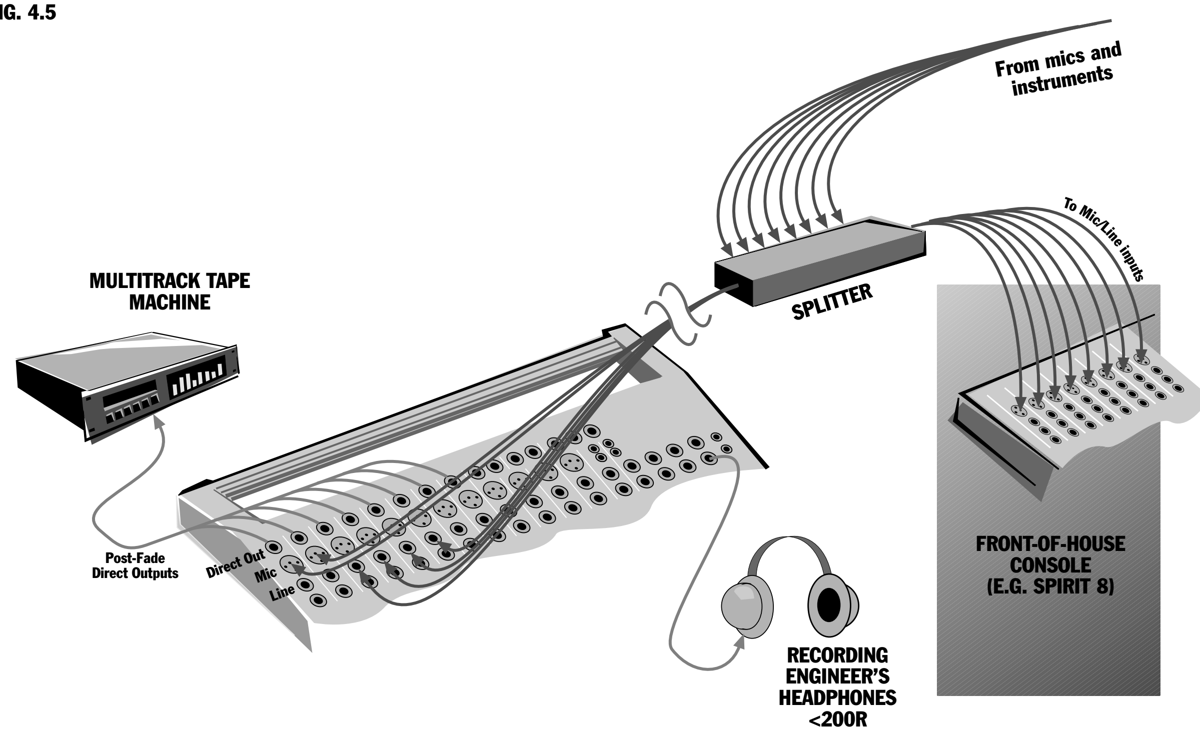
*NB: Subgroups can be used for submixing many inputs (e.g. drums) to a multitrack input. This is useful when tape track availability is limited.*

### Hints & Tips

- Try to locate the mixer in a different room to the performance to avoid distraction from the live sound. If this is not possible, use a good pair of noise-excluding headphones for monitoring.
- Wherever possible, take feeds from mic splitters - this will provide clean, low-noise signals suitable for recording.
- Often, Tape Sends are unbalanced, so keep signal paths as short as possible between output and recorder to avoid interference.
- If there aren't enough microphones, use a stereo pair to pick up the overall sound and the rest to emphasize individual performers.
- Use a compressor/limiter to avoid overloading the digital input of the recorder.

**RECORDING LIVE**

FIG. 4.5



# OTHER APPLICATIONS

## A. Monitor Mixing

Monitors are used to allow band members to hear themselves.

When dealing with the monitoring requirements of, say, a large live band, it is common practice to keep the monitor mix function totally separate from the Front of House console.

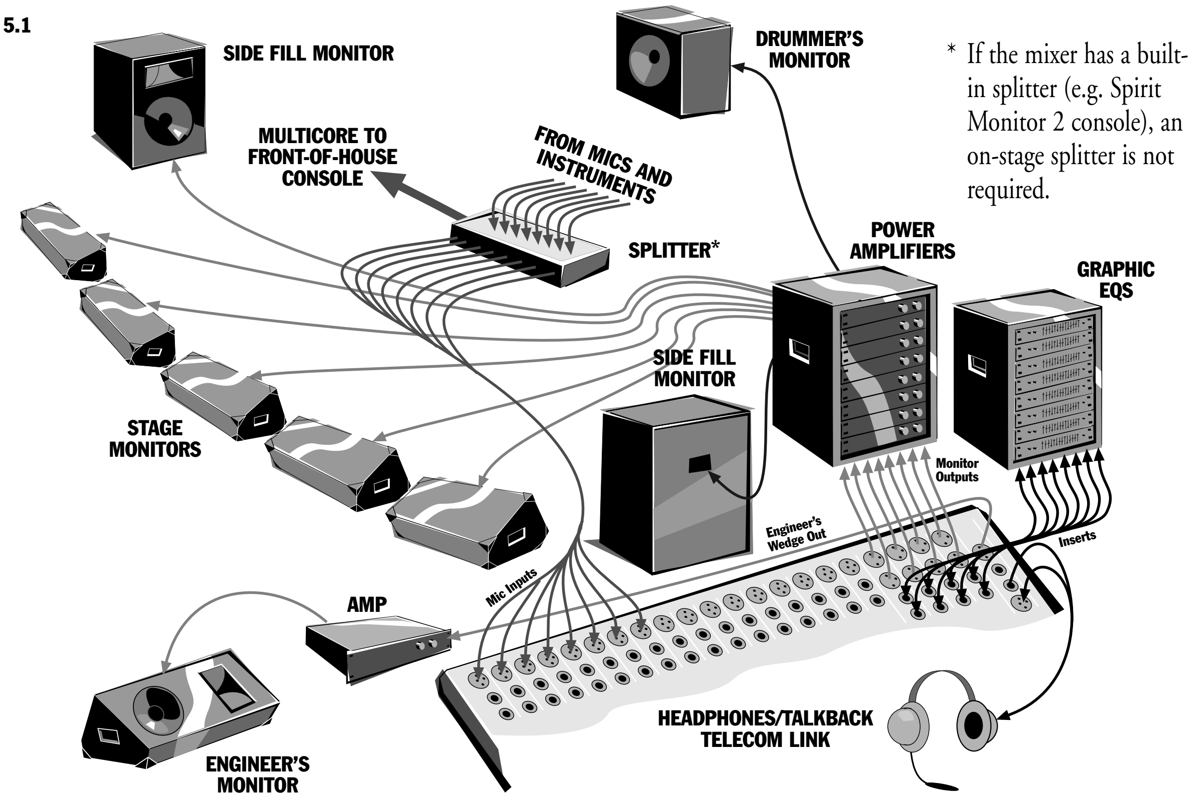
Some form of graphic equaliser in line with each monitor speaker is desirable as it allows troublesome frequencies to be notched out. The monitor system is rung out in exactly the same way as the main PA (see Ringing Out Section 4), and the final ringing out must be done with both the monitor and main PA systems set at their normal operating level.The monitoring console is situated off-stage and derives its feed direct from mic splitters. Note: the Spirit Monitor 2 console has its own built-in mic splitters.

- It is normal for a telecommunication link to be used between the FOH and monitor engineer so that they can talk to each other during the performance.
- Each stage monitor needs its own power amp. Keep things tidy by using rack-mounted stereo amps.
- Graphic EQs are patched via the console, like the power amps they should be rack-mounted for easy access.
- If the lead vocalist uses in-ear monitoring, he/she will be acoustically isolated, so it's a good idea to feed audience pick-up mics into his/her mix to provide a sense of involvement.
- 'Side fills' are often used where monitoring is required over a large stage area, floor space is at a premium, and too many wedge monitors would simply clutter things up both physically and acoustically. Don't compromise on these speakers - they'll have to work hard to punch sound through to the performers.
- The Monitor Engineer's wedge lets him hear the total foldback mix or selected parts thereof.
- A good Monitor Engineer, who is "invisible" to the audience, will always position himself so as to see visual signals from the performers.

SECTION 5: Other Applications

**M O N I T O R  M I X**

FIG. 5.1




* If the mixer has a built-in splitter (e.g. Spirit Monitor 2 console), an on-stage splitter is not required.

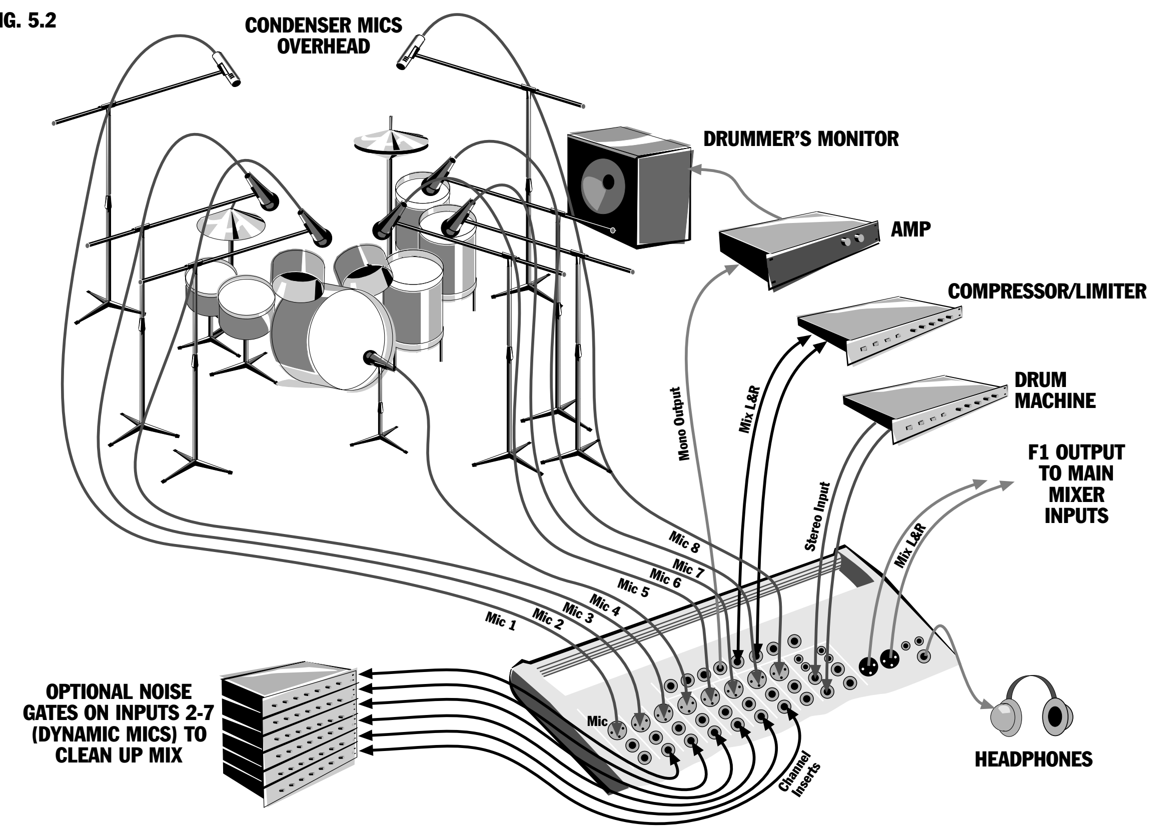
## B. Submixing

There are certain groups of instruments or performers (drums, backing vocals, multi-keyboards, etc) that can be logically grouped together - to save on input channels - via a small mixer, the output of which can then be controlled by just one pair of faders on the master console.

- If a mono output is available it can be used for a drum fill or for recording purposes.
- Output from the submixer goes to the FOH console and/or may be used for a small recording set up.
- Use the Aux Returns on the FOH console to return the sub-mix. This saves valuable input channels on the FOH console.
- In the case of a drum kit where many mics are in close proximity, the use of Noise Gates will prevent spill and clean-up the mix.
- Use a Compressor/Limiter to maintain a consistent level.

**SUBMIXING**

FIG. 5.2



# IN THE STUDIO

## A. Essentials & Ergonomics

Think about room layout and equipment. No, we're not going to plan your studio for you, but here are a few pointers:

- If you play keyboards, set them up so that you can reach the mixer.
- Position your effects and synth modules within arms length.
- If you use a computer, position the screen so as to avoid reflections. Do not position speakers near the screen unless they are magnetically compensated or shielded.
- If the room is too 'live', deaden it with drapes or soft furnishings.
- For best results, use dedicated nearfield monitors.
- Don't use large speakers in a small room - they'll sound wrong at low frequencies.
- Do use a well specified power amp (minimum 50 watts per channel).
- Don't compromise on a weedy amp: it will distort at high levels and may damage the speakers.

## B. Tape Machines & Recording Media

Basically, you'll need two types of tape machine: a **multitrack recorder** for recording the individual parts of the performance in readiness for mixdown onto a **2-track recorder** for mastering. There are both analogue and digital models available. The final choice must be based on individual requirements.

## C. The Console

Studio work presents additional problems for a mixing console in that it has to deal with a two stage process requiring very different skills.

1 Recording - Sound sources have to be captured on multitrack tape. This process will include ensuring that the cleanest strongest signal is being recorded to tape, without overload and distortion, optimising the sound of the recorded signal with EQ, signal processing and effects, monitoring the recorded sources, and creating a headphone mix for the musicians to ensure the best possible performance from them.

2 Mixdown - All the recorded sound sources as well as any "live" media coming from sequencers, drum machines or samplers must then be blended together using EQ, level, pan and effects and mastered down to a two-track device to create a "final mix". This process bears some similarities to mixing a band - minus the audience, the live performance and poor venue acoustics!

If you have seen any T.V. shows including footage of commercial recording studios you may be forgiven for thinking that good multitrack recordings are only possible using a mammoth console. This does not have to be the case! Professional sounding results can be achieved, albeit with some repatching between recording and mixdown stages, using a relatively small multipurpose mixer. However, to achieve professional results the mixer must be equipped with either (and preferably both):

- **Direct outs**
- **Groups/Subs**

When purchasing a console for both live and recording work, ensuring these facilities are available will save you having to buy a dedicated recording console until your requirements become more sophisticted.

## D. Simple Multitrack Recording

The diagram below shows a simple recording set-up using a multipurpose console equipped with direct outs and a pair of subgroups. The sound from instruments or voices is taken straight out to be recorded by the multitrack, with recorded signals being returned from the multitrack's channels into spare inputs of the mixer so they can be monitored. Alternatively, backing vocals or grouped instruments such as drumkits may be recorded to single or pairs of tracks by subgrouping them and connecting the mixer's group outputs to the multitrack device.

The engineer monitors both performances and previously recorded material through a monitor amp and speakers, with the performers getting their own separate foldback mix through the auxiliary sends .

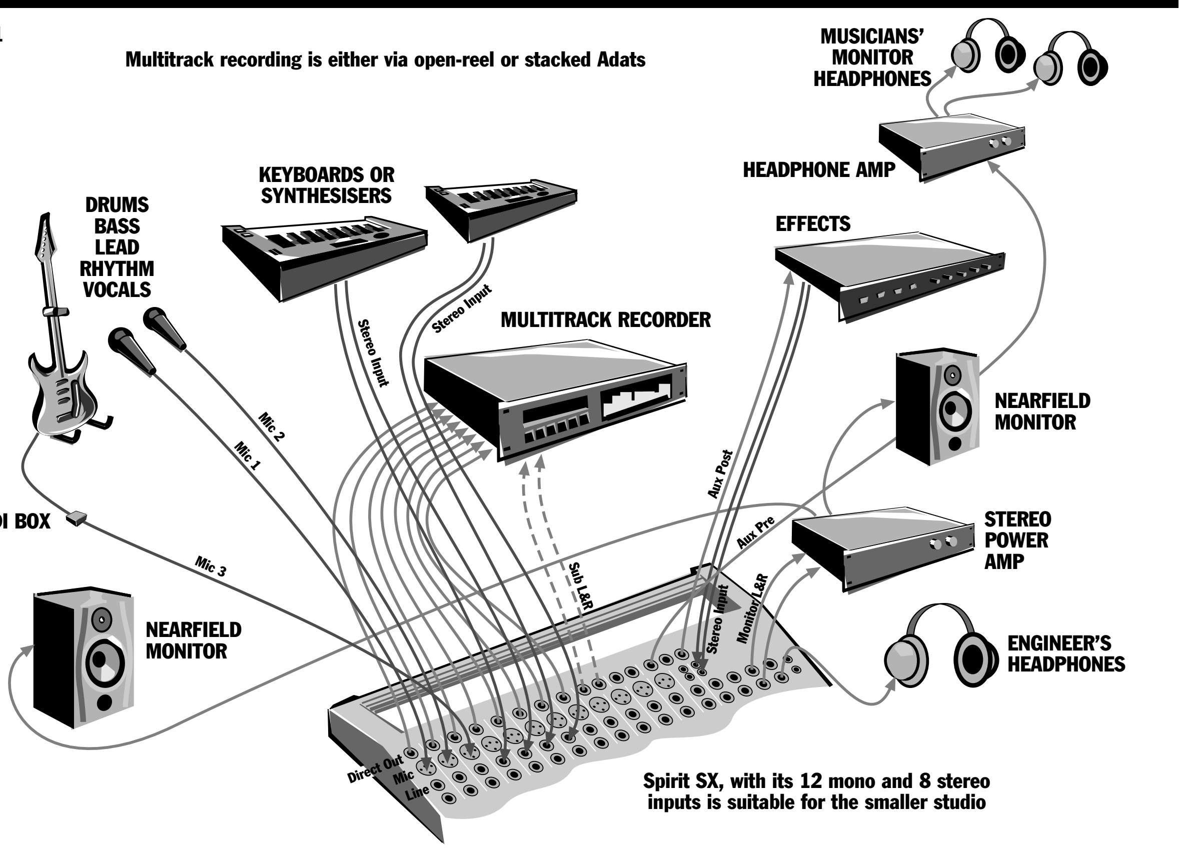
### Hints and Tips when Recording:

- If you are recording as a solo performer on a budget, you can avoid the expense of buying a separate amp to create a headphone mix. Plug your headphones into the console's headphone connector and use its monitor mix for your foldback. Alter channel fader levels as you wish to achieve optimum headphone levels for your performance.
- If your console is not large enough to cope with every multitrack send and return, connect only as many Direct Outs as you need per take. For example, if you are recording solo you will only be recording one instrument at a time anyway, so a maximum of only two direct outs will be required for stereo instruments, and one for mono ones. The same channel direct outs may then be repatched to adjacent multitrack tape ins to record new tracks. This should leave enough channels free to monitor all your recorded tracks.
- If you run out of tape tracks, group instruments together. For example a fully mic'd up drumkit can be recorded in stereo to two tape tracks via a pair of groups, or if you are really stretched you could do this with the entire rhythm section, including bass and rhythm guitar. However, it is then essential to mix the balance between the instruments accurately as, once recorded, they can never be individually altered again.
- If you have only one effects unit and you need it to create a variety of different sounds, it may be neccessary to record the instrument with effects included. Again, remember that once you have done this there is no going back, so wherever possible it is best to record "dry" and buy a second effects unit if you can. If you must record "wet", look at you

**M U L T I T R A C K  R E C O R D I N G**

FIG. 6.1

Multitrack recording is either via open-reel or stacked Adats




Spirit SX, with its 12 mono and 8 stereo inputs is suitable for the smaller studio mixer's block diagram and use outputs coming after the effects return for this purpose.

- Do not record in the same room in which you are playing unless your monitor speakers are muted. At the very least, your recorded track will pick up the mix from the monitor speakers, but more likely howl-round and feedback will occur which will damage your equipment. If you a recording a band, it is best to put them in an entirely different room altogether.
- Setting recording levels - for the best results, as it is important to set the highest record levels you can on your multitrack without getting overload or distortion. If you set levels too low, you will end up with a weak signal and background hiss. All multitrack recorders allow you to set record levels before a take. Consult the recorder's manual as to how best to achieve this.

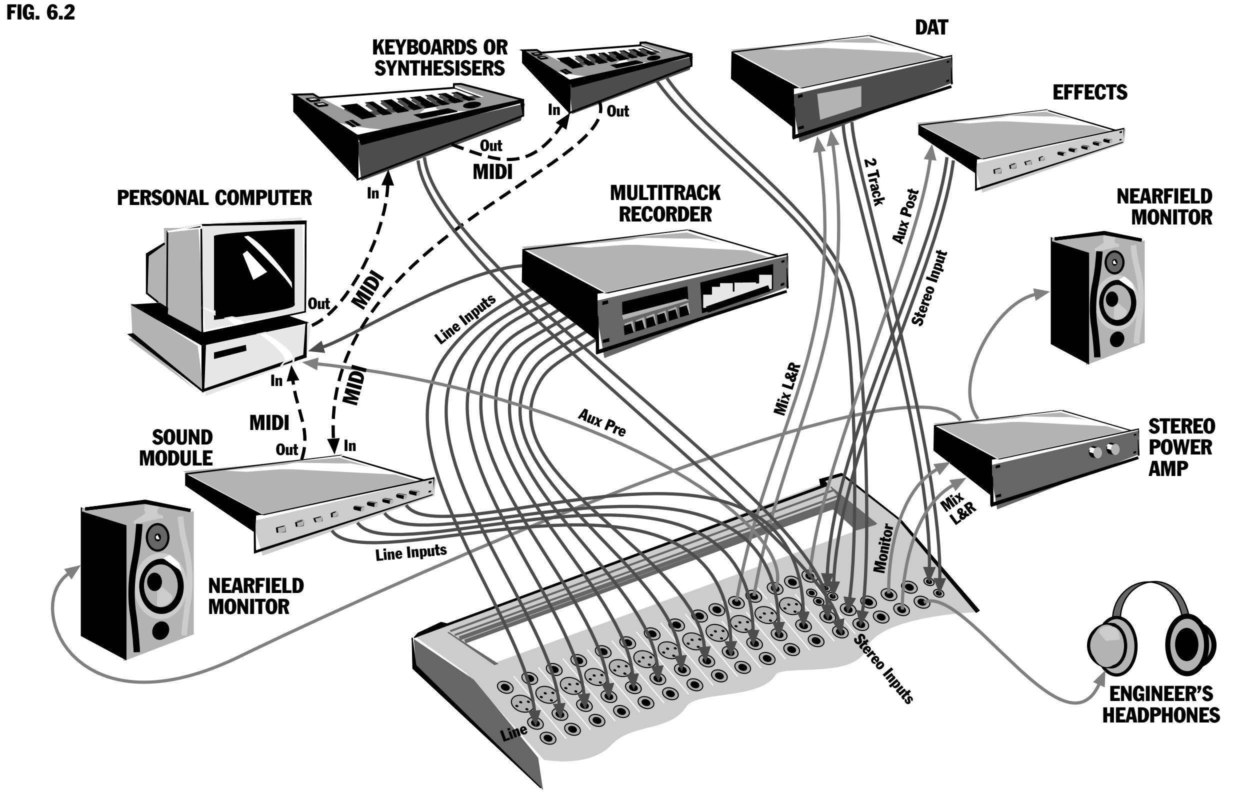
## E. Simple Multitrack Mixdown

The diagram below shows how a simple set-up will look for the mixdown process. Some repatching has occured to free up the input channels which were used as mutitrack tape sends. Tape returns can then be plugged into the mixer in sequence from channel 1 upwards, leaving any spare inputs for sequenced MIDI instruments. Effects, amps and speakers may be left as before.

*NB: Mixdown hints and tips may be found in "Creating a Mix" at the end of this section.*

**S I M P L E   M U L T I T R A C K   M I X D O W N**

**FIG. 6.2**



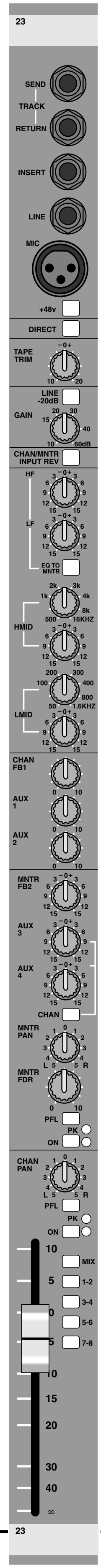
## F. Using a Dedicated "In-Line" Mixing Console

For recording projects beyond 8 track, a multipurpose console is usually inadequate, being unable to cope with the additional multitrack sends and returns and with all the repatching that is required between recording and mixdown. In such cases, a dedicated "in-line" recording console is necessary. An example of the input strip of such a console is shown here.

Virtually all of the features and facilities are identical to a standard mixer - except one: As well as including full channel input facilities and a direct out (here called a tape send), the strip also includes an extra input for a multitrack tape return as well as some basic rotary level control and pan facilities for that input. This second input is known as the **Monitor Input** or **Monitor Return**. Using this technique allows a signal to and from a multitrack to be handled by one input strip, saving space and avoiding the confusion of having to find corresponding send and return signals in different areas of the console.

The major advantage of using an "in-line" recording console is that repatching is unnecessary. This is because both channel and tape return inputs can be swapped (using the switch marked "Chan/Mntr Input Rev"), giving the signal coming from multitrack all the EQ, Auxiliaries and the linear fader of the channel input for the mixdown process. This also leaves the monitor input free for sequenced MIDI gear such as keyboards. If more facilities are required for these sound sources, then EQ and auxiliaries may be shared between the two inputs.

With two inputs per channel, a 16 channel "in-line"console actually has 32 inputs available. This high input count and compactness has made "in - line" consoles extremely popular with project studios, programming and remixing suites and commercially successful bands' home studios. With prices tumbling all the time, "in-line" consoles are now barely more expensive than standard designs.




### Multitrack Recording and Mixing with an "In-Line" Console

A more complex recording set up with an "in-line" console is shown opposite in Fig 6.4. Both multitrack ins and outs are plugged into the same channel strip, avoiding the need for repatching, whilst for sound proofing purposes, musicians are recorded in a separate room. Effects and signal processors are connected in an identical way to any other console via auxiliary sends and returns and insert points.

## G. Recording Instruments and Voices

### VOCALS

- Use a cardioid condenser mic positioned 9 inches (225mm) from the singer.
- A pop shield will reduce explosive 'p' and 't' sounds.
- If sibilance is a problem, change to a dynamic mic or move the singer back from the mic.

Recommended effects/processor settings:

**EQ:** Not normally required. But, if necessary, use the HPF (High Pass Filter) to reduce rumble.

**Compressor:** Attack as fast as possible;
Release around 0.5S, ratio between 4:1 and 8:1.

**Reverb:** Try a decay time of around 3 seconds and a pre-delay of 50mS.

### DRUMS

- Place mics 2 inches (50mm) from the heads of snare and kick drum.
- For the kick drum, place the mic inside - pointing directly at where the beater strikes the drumhead.
- To fully mic a kit, use separate mics on all toms and hats.
- Use condenser mics 5ft (1.5m) overhead, spaced around 5ft (1.5m) apart, to pick up the entire drum sound, cymbals and "ambience".

Recommended effects/processor settings:

**EQ:** Boost at: 80Hz to add weight to kick drums, 6kHz to add sizzle to cymbals or edge to a snare. Cut at 250-300Hz to reduce boxiness on a kick drum or low toms.

**Gate:** Fast attack setting to allow percussive transients to pass through. Precise settings will depend on microphone type and placement.

**Reverb:** Keep kick drum 'dry'. Try a percussion plate setting with a 2.5S decay time on other drums.

**FIG. 6.3**

## ELECTRIC GUITAR

- Some players prefer the sound of a valve amplifier, so be prepared to mic up the speaker cabinet using a cardioid dynamic mic.
- Experiment with mic positioning to achieve the desired sound.
- If preferred, the guitar can be DI'd via a recording preamp which incorporates an amp simulator.

Recommended effects/processor settings:

**EQ:** Boost at: 120Hz to add 'thump' to rock guitars, 2-3kHz to add bite, 5-7 kHz to add zing to clean rhythm sound. Cut at: 200-300Hz to reduce boxiness, 4kHz and above to reduce buzziness.

**Compressor:** Attack between 10 and 50mS; Release, around 0.3S, Ratio, between 4:1 and 12:1. Because of the noise generated by a typical electric guitar, use in conjunction with a gate or expander is advised.

**Reverb:** Plate or room, 1.5 to 4S; 30 to 60mS pre-delay.

## ACOUSTIC GUITAR

- Use the best mic that you can, preferably a condenser type.
- For a natural tone, position the mic between 12-18ins from the guitar, aiming at where the neck joins the body.
- If recording in stereo, point a second mic towards the centre of the neck, about 12-18ins from the instrument.
- Acoustic guitars sound best in slightly live rooms, if necessary place a piece of acoustically reflective board beneath the player.

Recommended effects/processor settings:

**EQ:** Boost: between 5kHz and 10kHz to add sparkle. Cut between: 1kHz and 3kHz to reduce harshness, 100 and 200Hz to reduce boom. In busy pop mixes you can cut the low end to produce a more cutting rhythm sound.

**Compressor:** Attack 20 mS; Release, around 0.5S, Ratio, between 4:1 and 12:1.

**Reverb:** Bright setting such as Plate to add vitality. Decay time of between 2 to 3S.

## BASS GUITAR

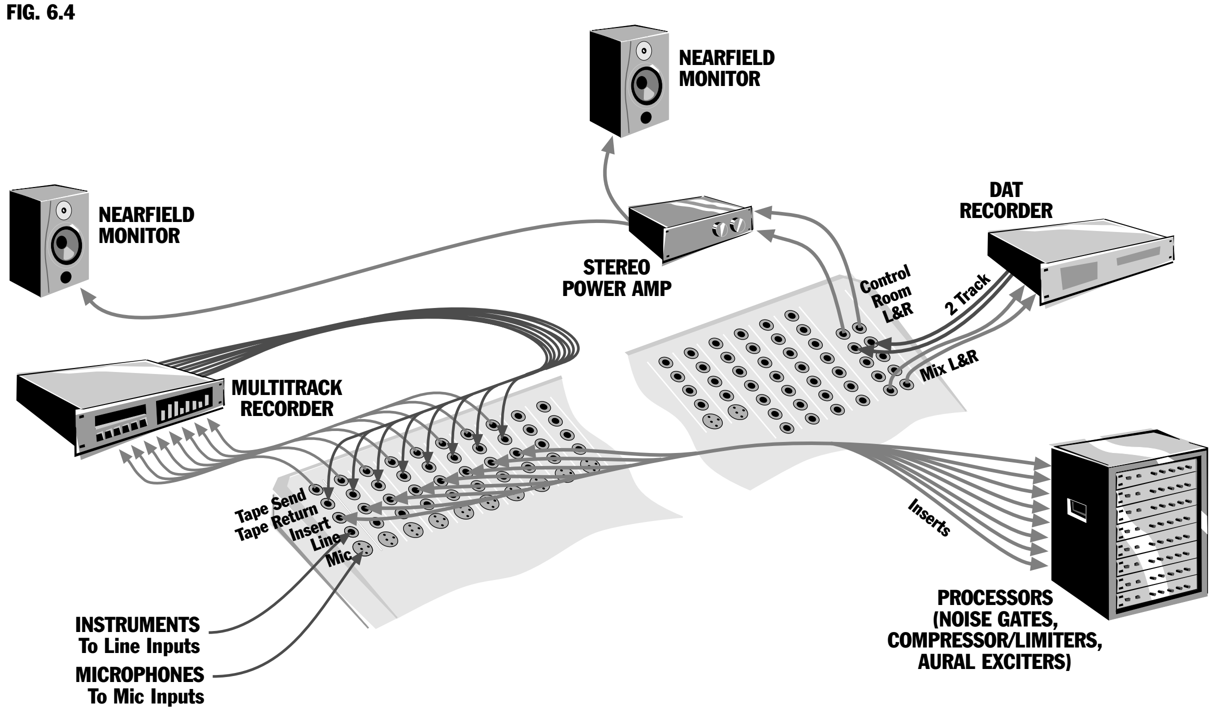
- Most engineers DI the bass via an active DI box and a compressor. This provides a clearer sound.
- Use the compressor to keep signal peaks under control.
- Check the player's technique; the harder the instrument is played, the brighter the tone.
- Consider the use of a budget graphic EQ.

Recommended effects/processor settings:

**EQ:** Boost: at 80-100Hz to add more weight and punch, between 2 and 4kHz to add edge. Cut: below 50Hz to

## MULTITRACK RECORDING & MIXDOWN

FIG. 6.4



reduce unwanted rumble, between 180 and 250Hz to reduce boxiness.

**Compressor:** Attack around 50mS; Release, around 0.4S, Ratio, between 4:1 and 12:1.

### KEYBOARDS

- Most electronic keyboards can be plugged directly into the line inputs of the mixing console.
- Bear in mind that the majority of contemporary synthesizers etc, have stereo outputs and will require two mixer channels.
- Most synthesizer sounds can be used without compression, though they do benefit from effects such as reverb or chorus.
- Overdriven keyboard sound may be created by feeding the signal via guitar recording preamp.

## H. Planning a Session

- You have a lot to remember during a session, so create a track sheet to keep a log of what instrument is recorded onto what tape track, plus other relevant information.
- Record rhythm sections first; drums, bass, and rhythm guitar.
- Add vocals, solos, and additional instrumentation as overdubs.
- Decide whether you want to add effects at the mixing stage or while recording. If you can, try to keep a copy of the original "dry signal" on tape. You may wish to remix at a later date!
- When recording vocals, ask the singer what instruments they most need to hear in the headphone mix.

## I. Creating a Mix

Go into 'neutral' before you start off -

- Set all the Aux Sends to zero.
- Set all EQ controls to their central positions.
- Pull all the faders down.
- All routing buttons 'up'.

**Organize your Subgroups**

- Put logical groups of sounds together.
- Route drums to a stereo sub-group.
- Consider grouping backing vocals.
- Group multiple keyboards.

**Metering**

- Use the PFL metering system for each channel in turn to optimize the gain setting.
- The PFL should just go into yellow band of the meter section, although peaking into the red area is acceptable.
- Check all the effects units for correct input levels.
- If fitted, use the Solo In Place function to check individual channels in isolation while retaining their original pan and level settings.

## J. Balancing the Mix

If you don't have a lot of mixing experience, it can help to set up the drums and bass balance first, then move onto the vocals and the other instruments. Don't worry about fine tuning the EQ or effects until your dry mix is somewhere near right.

- Satisfy yourself that the mix is working in mono. Check for Phase problems.
- Pan bass drums, bass guitar and lead vocals to centre - this will stabilize the mix.
- Spread other instruments across the stereo stage as required, including backing singers.
- EQ the mix as required.
- Now add stereo effects as necessary to add to the illusion of space and width.
- Check the balance of your final mix by listening to it from the next room through the adjoining door: for some reason, this often shows up whether the vocals are too loud or quiet.

### Hints & Tips

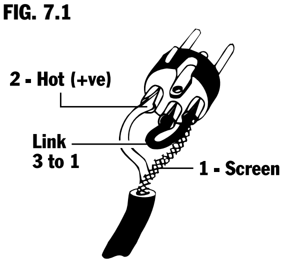
- Clean the heads of analogue tape machines before every session. Use cotton buds dipped in Isopropyl Alcohol.
- Check all instrument tunings before each take, because they have a tendency to change as the room warms up.
- Make a pop shield from stocking material stretched across a wire frame. This will minimise vocal "popping".
- Don't skimp on cables and connectors; these can be a source of noise.

# WIRING & CONNECTORS

Faulty connectors and cabling are some of the most frequent sources of noise and poor sounding systems. The following section should help you connect your system correctly. It's also worth spending a little time referring to all of your user manuals, as wiring conventions can vary between manufacturers - see diagrams.

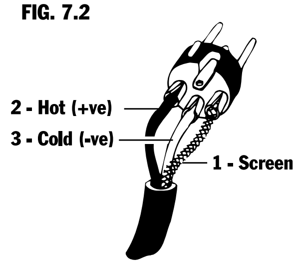
## BALANCED AND UNBALANCED MIC INPUTS

FIG. 7.1




UNBALANCED INPUT

FIG. 7.2




BALANCED INPUT

Soundcraft uses XLR sockets for its balanced mic inputs. The wiring convention for XLRs is: Pin 1 - Shield, Pin 2 - Hot (+ve) and Pin 3 - Cold (–ve).

Balancing is a method of audio connection which cancels any interference in a signal, to give low noise operation. This is achieved by using a 2-conductor mic cable, usually surrounded by a shield, in which the 'hot' and 'cold' signals are opposite polarity. Any interference picked up will be of the same polarity on both hot and cold wires and will be rejected by the mic input's Difference Amplifier. You may use unbalanced sources when wired as shown. However, do not use unbalanced sources with Phantom Power switched on. The voltage on Pins 2 & 3 of the XLR connector may cause serious damage.

## BALANCED AND UNBALANCED LINE INPUTS

Line inputs accept 'A' Gauge, 3-pole (Tip, Ring, Sleeve) 1/4 inch jack wired as shown in Fig. 7.3.

FIG. 7.3

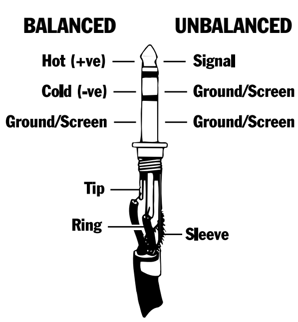
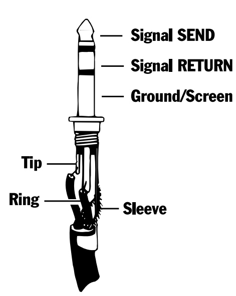



FIG. 7.4




*Note that for unbalanced operation the screen of the cable is wired to both the Ring and the Sleeve of the jack.*

## INSERTS

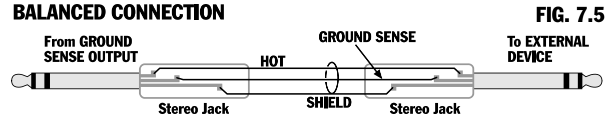
A Mixer insert point is a single, 'A' Gauge, 3-pole (stereo), switched jack socket (not unlike the headphone socket on a hi-fi amplifier). When a 3-pole jack is inserted the signal path is interrupted. The signal is taken out of the mixer via the plug tip, through an external piece of equipment and then back to the mixer on the ring of the plug. A special Y-cord is required which has the stereo jack at one end and two mono jacks, for the processor's input and output, at the other. See Fig. 7.4.

## GROUND COMPENSATED OUTPUTS

Ground compensated outputs may, to all intents and purposes, be treated as balanced outputs. Ground compensation helps avoid hum loops when the console is feeding into an unbalanced piece of equipment. Essentially, the Ground Compensated output has three connections, much like a conventional balanced output, except that the pin normally designated 'cold' acts as a 'ground sense' line enabling it to sense and cancel any ground hum present at the output.

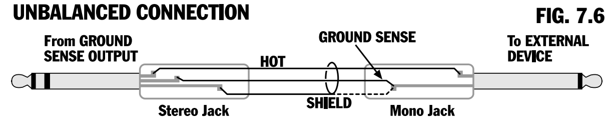
BALANCED CONNECTION

FIG. 7.5




UNBALANCED CONNECTION

FIG. 7.6




The convention for XLRs is: Pin 1 - Shield, Pin 2 - Hot, Pin 3 - Ground Sense. For jacks, the wiring convention is: Tip - Hot, Ring - Ground Sense, Sleeve - Shield.

For use with balanced destinations, the Ground Sense output may be treated as 'cold' allowing the connection to be made normally. Where the destination has an unbalanced jack input, a two-core (balanced-type) lead should be made up as shown. Unbalanced jacks may also be plugged directly into Ground Compensated Output jack sockets, but the benefit of hum rejection will be lost.

## IMPEDANCE BALANCED OUTPUTS

Impedance Balanced Outputs are configured as normal balanced outputs: Pin 1 - Shield, Pin 2 - Hot (+ve) and Pin 3 - Cold (–ve). See Fig. 7.2.

Impedance Balanced Outputs work on the principle that hot and cold terminals have the same resistance. When impedance balanced outputs are used with a balanced input, good rejection is achieved for both common-mode ground voltages and electrostatic interference.

*Note: The cold terminal can be either shorted to ground locally or left open-circuit for balanced and unbalanced operation.*

# GLOSSARY

**ACOUSTIC FEEDBACK (HOWLROUND)**
A whistling or howling noise caused by an amplified signal 'feeding back' into the amplification chain via a microphone or guitar pick-up.

**ACTIVE DI BOX**
A device which permits Direct Injection of signals from guitars, etc, into the console. Incorporates circuitry to adjust gain and provide impedance matching. Requires power and may be battery driven or sometimes 'phantom powered' from a console.

**AFL (AFTER FADE LISTEN)**
A function that allows the operator to monitor a post-fade signal. Used with Aux Masters.

**AMPLIFIER**
Device that increases the level of an electrical signal.

**AMPLITUDE**
Signal level, usually in volts.

**ANALOGUE**
Analogy (n.): correspondence or partial similarity, using physical variables. For example; an analogue tape recorder stores sound on tape in the form of a magnetic pattern which is a replica of the original musical waveform.

**ASSIGN**
On a mixing console, to switch or route a signal to a particular signal path or combination of signal paths.

**ATTENUATE**
To decrease the level of a signal.

**AUXILIARY SEND**
Level control feeding a dedicated bus for driving external effects or a foldback monitoring system. An output from the console comprising a mix of signals from channels derived independently of the main stereo/group mixes. Typically the feeds to the mix are implemented on rotary level controls.

**BACK-LINE**
Stage parlance for the row of instrument amplifiers and loudspeaker cabinets behind the performers, e.g. guitar amps.

**BALANCE**
Relative level of the left and right channels of a stereo signal.

**BALANCED**
A method of audio connection which 'balances' the signal between two wires and a screen which carries no signal. Any interference is picked up equally by the two wires, through common mode rejection at the destination differential balanced input resulting in cancellation of the interference signal. For balancing to be effective, both the sending and receiving device must have balanced output and input stages respectively.

**BANDWIDTH**
A means of specifying the range of frequencies passed by an electronic device such as an amplifier, mixer or filter.

**BARGRAPH**
A row of LEDs calibrated to indicate signal level.

**BOOST/CUT CONTROL**
A single EQ control which allows the range of frequencies passing through its filter to be either amplified or attenuated. The centre position is usually the 'flat' or 'neutral' position.

**BUS or BUSS**
A defined set of conductors along which signals may travel. A mixer has several busses carrying the stereo mix, the groups, the PFL signal, the aux sends, etc.

**CAPACITOR**
See Condenser

**CARDIOID PATTERN**
The 'heart-shaped' polar response of a microphone meaning that most of the sound is picked up from the front. Mainly used for stage vocals or in any situation were sound has to be picked up from a concentrated area, i.e. drums.

**CHANNEL**
A strip of controls in a mixing console relating to a single mono input or a stereo input.

**CHIP**
Integrated circuit; a multi-pinned device consisting of many circuits encapsulated in plastic.

**CHORUS**
Effect created by doubling a signal and adding delay and pitch modulation.

**CLIPPING**
Severe form of audio distortion which is the result of signal peaks exceeding the amplifier capacity. Normally caused by a limitation of the unit's power supply.

**CLONE**
Exact duplicate. Often refers to digital copies of digital tapes.

**CONDENSER**
Electrical component exhibiting capacitance (the ability to temporarily store electric current) and block direct current.

**CONDENSER MICROPHONE**
A type of microphone which picks up sound via a thin, flexible diaphragm placed in proximity to a metal plate - as opposed to the rigid diaphragm-and-coil system used by dynamic microphones. Condenser mics are very sensitive, especially to distant sounds and high frequencies. They have to be powered, which can be achieved by batteries, but for professional use a 48v DC PHANTOM POWER supply is provided from the console via the balanced mic cable.

**CONDUCTOR**
A thing that conducts or transmits heat or electricity.

**COMPRESSOR**
A device designed to control or reduce the dynamic range of an audio signal.

**CROSSOVER**
A passive circuit, normally built into a speaker system which divides the full-range audio signal from an amplifier in order to feed the individual drive units, ie: bass, midrange and treble.

**CUEING**
To put a piece of equipment in readiness to play a particular part of the recording material. Assisted on a mixing console by use of the PFL (Pre-Fade LIsten) facility.

**CUT-OFF FREQUENCY**
The frequency at which the gain of an amplifier or filter has fallen by 3dB.

**DAT (DIGITAL AUDIO TAPE)**
High quality cassette based recording format which stores signals digitally and therefore provides very high quality sound. Originally touted for consumer use, but now firmly ensconced as a professional tool.

**dB (DECIBEL)**
A ratio of two signal levels. Can be in Voltage, Watts or Current units.

**dBm**
Variation on dB referenced to 0dB = 1mW into 600 ohms.

**dBu**
Variation on dB referenced to 0dB = 0.775 volts.

**dBV**
Variation on dB referenced to 0dB = 1 Volt.

**DETENT**
In audio terms a click-stop in the travel of a rotary or slide control, normally used to indicate 'centre stereo' on pan-pots or 'zero boost/cut' on EQ controls.

**DI BOX**
A device allowing connections as explained below.

**DI (DIRECT INJECTION)**
The practice of connecting an electric musical instrument directly to the input of the mixing console, rather than to an amplifier and loudspeaker which is covered by a microphone feeding the console.

**DIGITAL DELAY**
The creation of delay and echo effects in the digital domain.The premise being that, as digital signals are resistant to corruption, the process will not introduce additional noise or distortion.

**DIGITAL REVERB**
Reverberation effects created as above.

**DIGITAL**
The processing and storage of signals with sound- information represented in a series of '1s' and '0s', or binary digits.

**DIRECT OUTPUT**
A pre-/post-fade, post-EQ line level output from the input channel, bypassing the summing amplifiers, typically for sending to individual tape tracks during recording.

**DRY**
Slang term for an original audio signal that has had no added effects.

**DYNAMIC RANGE**
The ratio in decibels between the quietest and loudest sounds in the audible range that the audio equipment will reproduce.

**DYNAMIC MICROPHONE**
A type which uses a thin diaphragm attached to a coil of wire arranged about a permanent magnet. Any variation in air pressure on the diaphragm will cause the coil to generate a minute electric current which then requires amplification.

**EARTH**
See GROUND.

**EFFECTS**
The use of devices to alter or process the sound to add special effects eg; reverb, normally as a mix of the original ('dry') sound and the treated ('wet') version.

**EFFECTS RETURN**
Additional mixer input designed to accommodate the output from an effects unit.

**EFFECTS LOOP**
Connection system that allows an external signal processor to be connected into the audio chain.

**EFFECTS SEND**
A post-fade auxiliary output used to add effects to a mix.

**ELECTRET MICROPHONE**
Type of condenser microphone using a permanently charged capsule.

**ELECTRONIC CROSSOVER**
An active device which divides the full range audio signal into several narrower frequency bands (eg:low, mid and high), which are then amplified and fed to the appropriate speaker drive units.

**ENCLOSED HEADPHONES**
Types that completely enclose the ears and therefore provide good exclusion of outside noise. Of particular use when monitor mixing or recording live on stage.

**EQ**
Abbreviation for equaliser or equalisation.

**EQUALISER**
A device that allows the boosting or cutting of selected bands of frequencies in the signal path.

**EXPANDER**
The opposite of a compressor, an expander increases the dynamic range of signals falling below a pre-determined threshold.

**FADER**
A linear control providing level adjustment. Favoured by professionals due to smoothness of activation and the ability to give an instant visual indication of status.

**FILTER**
A filter is a device or network for separating waves on the basis of their frequencies.

**FOH**
An acronym for Front Of House. In the entertainment world "House" is a collective term for the audience at a theatre, cinema, etc. Hence an FOH console will be situated "audience-side" of the stage. A "house" PA system refers to the main audio system responsible for the principal sound in the venue.

**FOLDBACK**
A feed sent back to the artistes via loudspeakers or headphones to enable them to monitor the sounds they are producing.

**FOLDBACK SEND**
A pre-fade auxiliary output used to set up an independent monitor mix for the performers.

**FREQUENCY RESPONSE**
The variation in gain of a device with frequency.

**FSK (Frequency Shift Keying)**
A method of synchronisation which generates a series of electronic tones related to the tempo of the music. These tones may then be record on a spare track of the multitrack recorder.

**FX UNIT**
Slang term for Effects Unit. Typical effects units are delays, reverbs, pitch shifters, and chorus units.

**GAIN**
Gain is the factor of how much the level of a signal is increased or amplified. Normally expressed in decibels.

**GATE**
A user-adjustable electronic device that switches off the signal path when the signal falls below a certain predetermined level or threshold.

Typically used to ensure silence between pauses in the signal during vocal passages or to prevent ‘spill’ between the close-proximity, multiple mics on a drum kit.

**GRAPHIC EQUALISER**
Device incorporating multiple narrow-band circuits allowing boost and cut of predetermined frequencies. Vertical fader controls are used which provide a ‘graphic’ representation of the adjustments across the frequency range.

**GROUND COMPENSATION**
A technique used to cancel out the effect of ground loops caused by connections to external equipment.

**GROUND**
Ground and Earth are often assumed to be the same thing, but they are not. Earth is for electrical safety, while Ground is the point of zero voltage in a circuit or system.

**GROUND LOOP**
A ground loop occurs when there are too many ground points, allowing small electrical currents to flow.

**GROUP**
An output into which a group of signals can be mixed.

**HEADROOM**
The available signal range above the nominal level before clipping occurs.

**HERTZ (Hz)**
Cycles (or vibrations) per second.

**HIGH PASS FILTER**
A filter that rejects low frequencies below a set frequency, typically 100Hz.

**IMPEDANCE**
The AC resistance of a circuit which has both resistive and reactive components.

**IMPEDANCE BALANCING**
A technique used to minimise the effect of hum and interference when connecting to external balanced inputs.

**INDUCTOR**
Reactive component that presents an increasing impedance with frequency. A coil in a loudspeaker crossover is an inductor.

**INSERT POINT**
A break point in the signal path to allow the connection of external devices, for example signal processors or to another mixer.

**K OHM, K Ω or kHz**
x 1000 ohms, x 1000 ohms and x 1000Hz respectively.

**LINE LEVEL**
Signals at a nominal level of -10dBV to +4dBu, usually coming from a low impedance source such as keyboards, drum machines,etc.

**mA (milliampere)**
One thousandth of an ampere, a measure for small electrical currents.

**MIC SPLITTER**
A device which divides the output from a microphone in order to supply two signals, for example; FOH console and recording mixer or monitor console.

**MIDBAND**
The range of frequencies to which the human ear is most sensitive.

**MIDI**
Musical Instrument Digital Interface.

**MIXDOWN**
The process of taking the outputs from a multitrack recorder, processing as required and combining all elements to create a stereo ‘master’.

**MONITOR LOUDSPEAKER**
Any high quality loudspeaker which is used to check the quality or status of the signal.

**MTC (MIDI Time Code)**
An interpretation of SMPTE allowing the time code to come in as part of the MIDI data stream.

**MULTICORE**
A cable with multiple cores allowing signals to be carried independently but within the same physical outer casing.

**MUTE GROUPS**
A method of combining the on/off status of a selection of channels under a single control button.

**NEARFIELD MONITOR**
A high quality, compact loudspeaker designed for use at a distance of three to four feet from the operator. Their use ensures that detrimental room effects are minimised.

**NORMALISE**
A socket is said to be normalised when it is wired in such a way that the original signal path is maintained unless a plug is inserted into the socket. The most common examples of normalised connectors are the INSERT POINTS found on mixing consoles.

**OSCILLATOR**
A tone generator for test and line-up purposes.

**OVERDUB**
To add another part to a multitrack recording or replace one of the existing parts.

**OVERLOAD**
To exceed the operating capacity of an electronic or electrical circuit.

**PAN (POT)**
Abbreviation of 'panorama': controls levels sent to left and right outputs. Allows positioning of signals within the stereo sound stage.

**PARAMETRIC EQUALISER**
A graphic equaliser in which the cut/boost, frequency and bandwidth are all adjustable.

**PASSIVE**
A circuit or component which does not amplify the signal or is not powered.

**PATCH BAY**
A system of panel mounted connectors used to bring inputs and outputs to a central point from where they can be routed using plug-in patch cords.

**PATCH CORD**
Short cable used with patch bays.

**PEAKING**
A signal of the maximum displacement from its mean (average) position.

**PHANTOM POWER**
The +48v DC voltage applied equally to the two signal pins of a balanced mic input to provide powering for condenser microphones.

**PHASE**
Phase is the fraction of the whole period that has elapsed, measured from a fixed datum. A term used to describe the relationship of two audio signals: in-phase signals reinforce each other, out-of-phase signals result in cancellation.

**PHONO PLUG**
A hi-fi connector developed by RCA and used extensively on semi-pro recording equipment.

**POLARITY**
The orientation of the positive and negative poles of an audio connection. Normally, connections are made positive to positive, negative to negative and this would ensure correct polarity. If this is reversed the result will be out-of-phase signals (see PHASE above).

**POP SHIELD**
A device used in the studio, consisting of a thin mesh placed between the microphone and vocalist in order to reduce the 'explosive' effects of 'P' and 'T' sound

**POST-FADE**
The point in the signal path after the channel or master fader and therefore affected by fader position.

**PRE-FADE LISTEN (PFL)**
A function that allows the operator to monitor the pre-fade signal in a channel before it reaches the main mix.

**PRE-FADE**
The point in the signal path before the monitor or master position and therefore unaffected by the fader setting.

**PROCESSOR**
A device which affects the whole of the signal passing through it, e.g. gate, compressor or equaliser.

**Q (Bandwidth)**
A measure of the sharpness of a bandpass filter. The higher the value of Q, the narrower the band of frequencies that passes through the filter.

**RESISTANCE**
Opposition to the flow of electrical current.

**REVERB**
Acoustic ambience created by multiple reflections in a confined space. A diffuse, continuously smooth decay of sound.

**RINGING OUT**
The process of finding the problem frequencies in a room by steadily increasing the gain of the system until feedback occurs. A GRAPHIC EQUALISER is then used to reduce the offending frequencies.

**ROLL-OFF**
A fall in gain at the extremes of the frequency response. The rate at which a filter attenuates a signal once it has passed the filter cut-off point.

**SEQUENCER**
Computer-based system for the recording, editing and replay of MIDI music compositions.

**SHELVING**
An equaliser response affecting all frequencies above or below the break frequency i.e. a high-pass or low-pass derived response.

**SHORT CIRCUIT**
The situation where two electrical conductors touch.

**SIBILANCE**
n. sounding with a hiss. When certain phonics are exaggerated, ie: s, sh.

**SIGNAL**
Electrical representation of input such as sound.

**SIGNAL CHAIN**
The route taken by a signal from the input to a system through to its output.

**SIGNAL-TO-NOISE RATIO**
An expression of the difference in level between the audio signal and the background noise of the device or system. Normally expressed in decibels.

**SMPTE (Society of Motion Picture and Television Engineers)**
Time code developed for the film industry but now extensively used in music and recording. SMPTE is a real-time code and is related to hours, minutes, seconds and film or video frames rather than to musical tempo.

**SOLO-IN-PLACE**
A function that allows the operator to listen to a selected channel on it's own but complete with all relevant effects, by automatically muting all other inputs.

**SOUNDCRAFT**
The name found on some of the best-value professional audio equipment around.

**SOUND REINFORCEMENT**
The process of amplifying or reinforcing on-stage sound (whether from already-amplified or acoustic instruments/voices) without overpowering the original sound. Suitable for smaller venues and often used solely to raise the level of the vocals above the back line and drums.

**SPL (Sound Pressure Level)**
Intensity of sound measured in decibels.

**STEREO**
Two channel system feeding left and right speakers to create the illusion of a continuous sound field. Stereo: from the Greek word for 'solid'.

**STEREO RETURN**
An input designed to receive any stereo line level source such as the output of effects or other external processing devices.

**STRIPE**
To record time code onto one track of a multitrack tape machine.

**SWEEP EQ**
An equaliser section (e.g. Midband EQ) which boosts or cuts a variable rather than fixed frequency.

**TALKBACK**
A system allowing the operator to speak to the artistes or to tape via the auxiliary or group outputs.

**TAPE RETURN**
A line level input provided specifically to receive the playback output of a tape machine.

**TRANSIENT**
An instantaneous rise in the signal level e.g. a cymbal crash or similar.

**TRIM CONTROL**
A variable control which gives adjustment of signal level over a limited and predetermined range usually for calibration purposes.

**TRS JACKS**
A 3-pole jack with Tip, Ring and Sleeve connection. Sometimes referred to as a stereo or A-gauge jack plug.

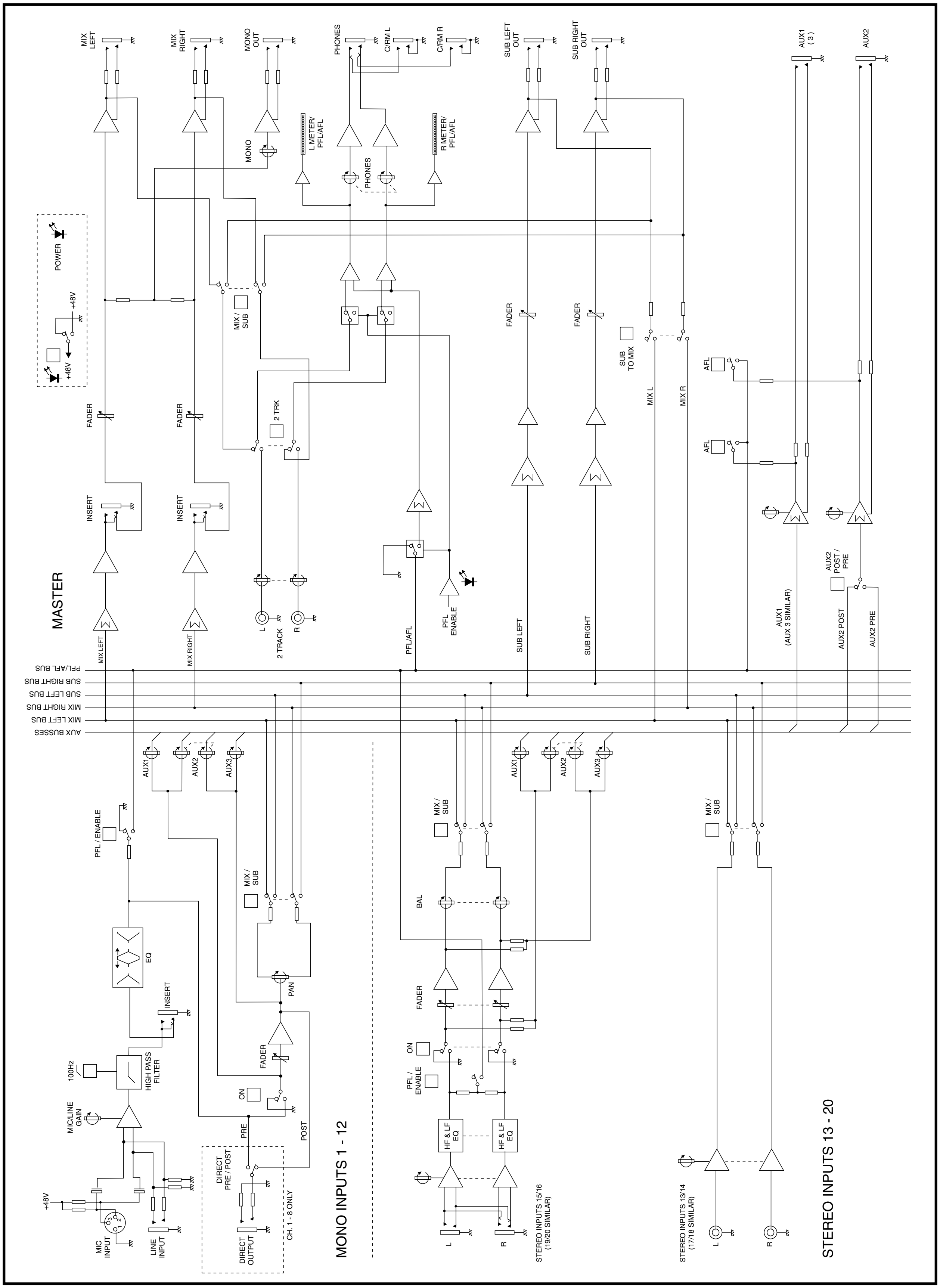
**UNBALANCED**
A method of audio connection which uses a single signal wire and the cable screen as the signal return.This method does not provide the same degree of noise immunity as a BALANCED connection.

**WET**
Slang term for a signal with added effects such as REVERBERATION, ECHO, DELAY or CHORUS.

**Y-LEAD**
A lead split so that one source can feed two destinations. Y-leads may also be used in console insert points in which case a stereo jack plug at one end of the lead is split into two mono jacks at the other.

**2-TRACK RETURN**
A line level stereo input on a mixing console designed to accept the output from a 2-Track recording device. May also be used as an additional effects return, depending on the internal routing of the mixer.

## A Typical Block Diagram **(Spirit SX shown)**



## FAQs



------------------------------------------------------------------------------------------------------------------------------------------------
-

### Can I use +48v Phantom supply with dynamic microphones connected to the mic XLR inputs?

It is possible to use BALANCED dynamic microphones when the +48v phantom is on. The microphone coil 'floats' up with the voltage preventing damage to the microphone by ensuring 'zero' volts across the microphone coil.

If the dynamic microphone is configured UNBALANCED (where pins 1 & 3 are connected together), then the dynamic microphone will be damaged. This is because one side of the microphone coil is connected to ground, with the other side connected to +48v. Thus, +48v will be across the microphone coil causing it to burn out.

**Always ensure the Mic lead is balanced when using +48v phantom power.**

back to the top

------------------------------------------------------------------------------------------------------------------------------------------------
-

### Do the Line Jack inputs have +48v on them when the +48v Phantom supply is on?

No. It is perfectly OK for line level audio sources to be connected to a Spirit console when the +48v phantom supply is active on the XLR input connectors.

back to the top

------------------------------------------------------------------------------------------------------------------------------------------------
-

### How do I connect my effects unit?

Spirit consoles have 'Post fade' Aux outputs. These outputs are designed to take a separate mix (the original 'DRY' signal), out of the console and into the input of an external effects device which modifies the signal in some way by adding chorus, reverb etc.

The output of the effects unit ( i.e. the'WET' signal) may be fed back into ANY input of the mixing console to be mixed with the original DRY signal on the mix buss. However most Spirit mixers have dedicated stereo FX Returns, specifically designed for this purpose.

back to the top

------------------------------------------------------------------------------------------------------------------------------------------------

-

## What do I use Inserts for?

Inserts allow you to send and receive a audio signal to and from the mixer.

They can be connected to various processors such as graphic equalisers, compressors, noise gates etc. To use a signal processor in the inserts, a special 'Y' lead is required. As the name suggests this consists of a single connector at one end with two connectors at the other. The single connector plugs into the mixer's insert point and allows signals to be sent both to and from it. The two connectors at the other end are plugged into the input and output of the signal processor. Go to the wiring and connectors section for a diagram of how this works.

back to the top

------------------------------------------------------------------------------------------------------------------------------------------------

-

## Can I plug a dynamic mic into the line input?

No. There is not enough gain to amplify a microphone in a line input. Line Inputs are not designed for microphones and although it is possible to use them, the results will not provide optimum.

Dedicated mic XLR inputs should always be used with mics. If you have a mic with a jack plug on it you should change it for a XLR connector.

back to the top

------------------------------------------------------------------------------------------------------------------------------------------------

-

## What do I use Line inputs for?

For plugging in "Line Level" instruments such as Keyboards, samplers or drum machines. They can also be used to accept the returns from multitrack tape machines and other recording media.

Keyboards, Drum Machines, CD Players, DAT Machines, Wireless Mic Receivers, all provide line level outputs. If some of your instruments are STEREO connect their left and right outputs to a spare stereo input. Alternatively connect to an adjacent pair of mono inputs and Pan the inputs hard left and right to create a stereo image.

back to the top

------------------------------------------------------------------------------------------------------------------------------------------------

-

## What are DI boxes used for?

A DI Box allows you to connect a guitar or bass directly to the mixer's input, rather than mic'ing up your guitar or bass amplifier. This technique is often preferred by musicians who require a "clean" sound. The best DI boxes are ACTIVE and require Phantom Power like condenser microphones. They should be connected to XLR mic inputs.

DI Boxes typically have the following features: Impedance matching, Transformer isolation, input attenuation, Low pass filter, Earth lift to prevent 'ground loops' and phase reverse. They are normally powered by a internal battery or by the mixing consoles +48v phantom supply.

back to the top

---

-

## Where do I connect a Direct Injection (DI)box?

DI boxes should be put into a mic XLR input. The mic XLR connectors provide +48v phantom which can supply DI boxes.

back to the top

---

-

## What do I use compressor/limiters for?

A compressor/limiter reduces the difference between the loudest and quietest passages of a signal. It works on a threshold system where signals exceeding the threshold are processed and those falling below it pass through unchanged. When a signal exceeds the threshold the compressor automatically reduces the gain.

How much gain reduction is applied depends on the 'compression ratio' which on most compressors is variable: the higher the ratio, the stronger the compression. Very high ratios cause the compressor to act as a limiter where the input signal is prevented from ever exceeding the threshold.

Compressors are the most commonly used processor and are particularly popular for maintaining constant vocal levels Live and in the Studio. This is because singers tend to vary their levels the most out of all instruments. Compressors help to achieve the sought after tight, "punchy" sound.

back to the top

---

-

## What do I use direct outputs for?

To send the sound direct from your channel out to a multitrack tape recorder, or to an effects unit when the channel requires its own special effect.

back to the top

---

-

## How do I choose the right mixer for my application?

Audio mixers come in many different sizes and at all price levels. There are several questions that need to be asked to narrow your search to the most appropriate models.

- What am I going to be using the mixer for - i.e. multitrack recording, live PA work or both?
- What is my budget?
- How many sound sources do I have?

As a guideline your mixer needs to have at least as many inputs as sound sources. If you are likely to be buying more equipment in the future you should budget for extra inputs.

What particular mixer facilities MUST I HAVE for my application? i.e. Plenty of EQ, auxiliaries, or Direct outs for recording.

back to the top

---

-

## How portable does the mixer need to be?

Will I be doing any location work where there won't be any mains power available?

back to the top

---

## What do I use the High Pass filter for?

As the name suggests this switch cuts out the very lowest frequencies of a sound whilst allowing the higher frequencies to "Pass Through". It's Particularly useful in live situations to reduce stage rumble or microphone 'popping', which can produce a muddy mix, or to 'clean-up' male vocals and filter out low frequency hum.

back to the top

---

## What do I use EQ for?

The equaliser section allows you to change the tone of the sound on each input. An EQ is normally split into "bands", which control a range of frequencies, in a similar fashion to the treble and bass tone controls on your Hi-Fi. Indeed a simple "2 band" EQ is little more than an input treble and bass control. The more bands an EQ has the more sophisticated it is. SX has a 3 band EQ, with a separate control for the middle audio frequencies. This control is also "swept" which provides even more sophistication. Simply described, a sweep EQ allows you to choose the exact frequency to cut and boost, rather than having it chosen for you, as on normal "fixed" controls.

back to the top

---

## What do I use Auxiliaries for?

Typically, these controls have two functions: First, to control the levels of the signal going to effects units, and second to create separate musician's "foldback" mixes in the studio or on stage.

- Effects use POST fade aux sends
- Foldback mixes require PRE fade aux sends

back to the top

---

## What do I use PFL for?

The PFL solo switch allows you to monitor an input signal independently of any other instruments that have been connected, which is useful for troubleshooting, or setting an instrument's Gain and EQ setting.

Pre-Fade Listen (PFL) is a type of solo that allows you to monitor your sound BEFORE THE FADER. In other words, when you move the input fader in PFL mode the level will not change, nor will you hear any effects. Because effects and volume are not a distraction, PFL solo is very useful for setting proper input levels.

back to the top

---

## What is the difference between Dynamic and condenser microphones?

Dynamic microphones utilise a robust design which uses a thin diaphragm attached to a coil of wire arranged about a permanent magnet. Any variation in air pressure on the diaphragm will cause the coil to generate a minute electric current which then requires amplification.

Dynamic mics are relatively inexpensive, rugged and require no electrical power to operate. They are ideal for all-round high sound pressure levels (SPL) and tend to be used for live applications. However, they are not as sensitive to high frequencies as condenser types.

Condenser Microphones pick up sound via a thin, flexible diaphragm placed in proximity to a metal plate - as opposed to the rigid diaphragm-and-coil system used by dynamic microphones. They need power to operate - the most common source being +48v DC PHANTOM POWER.

Condenser mics are very sensitive to distant sounds and high frequencies. Because of this sensitivity they are often used in studio recording situations.

back to the top

-----------------------------------------------------------------------------------------------------------------------------------------------------------------------

-

## How do I set up a basic mix?

- Press the PFL/Solo switch on the relevant input.
- Adjust gain/input sensitivity until meters read within the yellow ('3' to '6' on meter scale)
- Release PFL/Solo
- Repeat for all other inputs.

  NB: EQ affects gains settings. If you adjust the EQ you will need to re-check your gain level using the above method.

- Faders allow you to make fine adjustments to your sounds and act as a visual indication of the overall mix levels. It is important to keep your input faders aroundthe '0' mark for greater control. This is because fader scales are typically logarithmic and not linear so if your fader position is near the bottom of its travel then even a small movement will lead to huge leaps in level.Similarly try not to have your fader at the top of its travel because this will leave you no room to further boost the signal.
- Set your master outputs to '0' on the scale.

  There are three reasons for this:
  1. You have the maximum fader travel for fading out your mix.
  2. If your faders are set below '0' you will not be getting the full benefit from the meters because you will only be using the first few LED's on the meter scale.
  3. Your mixer is not an amplifier. So the master output faders should be set to maximum ('0' on scale). If extra output is required, then turn up your amplifier.

back to the top

-----------------------------------------------------------------------------------------------------------------------------------------------------------------------

-

## What is a VCA?

VCA stands for 'Voltage Controlled Amplifier' and is essentially an electronic level control element which is fitted to each input channel of a VCA-equipped console. You can imagine this element to act like a second 'hidden' channel fader, which has exactly the same effect on the signal as the regular channel fader that you move, except this one can be controlled by 'remote control'.

The way the 'remote control' capability is useful is that you can hook this electronic fader up to a control signal (which is the 'voltage' in VCA) which is actually genetated by another fader, located in the master section of the console. By moving this fader up and down you can then control the level of your channel from this master fader - in addition to being able to control it locally.

back to the top

-----------------------------------------------------------------------------------------------------------------------------------------------------------------------

-

## What would you use a VCA for?

What is the point of this? Well if you were to hook several channels' VCA elements up to the same control voltage master fader (or VCA master) you now have yourself a 'subgroup' capability, and are able to control all those channel faders with one master, whilst still being able to set the relative balance between channels on their local faders.

In a typical console there will be perhaps 8 of these VCA master faders, each of which sends out its own control voltage, and on each input channel there will be a way of deciding which master fader you want to use to control the VCA element in each input channel. So you can connect or 'assign' one group of several sources to one VCA master fader, and another group of sources to another master fader.

It will then be much easier to control a complex mix if you break it down like this into several subgroups and then only have to worry about the master faders whilst mixing.

back to the top

-----------------------------------------------------------------------------------------------------------------------------------------------------------

## What is the difference to using Audio Subgroups?

The result of this might seem to be the same as you can get by using tradditional 'audio subgroups' but there are three important differences -all of them are advantages in favour of VCAs:

1. With audio subgroups, you are routing each audio signal of a group of channels to a master audio fader, where they are added (mixed) together, and then back to the main mix bus, where they join with all the other channels that you haven't routed to subgroups. In just 'level control' terms, this gives you the same result as using VCAs....except when you consider what happens to any post-fade effects sends you have going from your source channels to a reverb or delay. If you were to fade your audio subgroup master all the way down, your source channels would disappear from the mix -or at least the 'dry' signals would -but the reverb sends are still sending from those source channels and you haven't been able to affect this with your audio subgroup master fader, so you still hear a great wash of reverb, even though you've faded your master out. So what is being illustrated here is that as you change your audio sugbroup level up and down during the mix, you are not maintaining the correct balance of effected vs dry signal. The subgroup cannot change the level of effects sends from the source channels. This is where VCA groups come in -in a VCA group you are controlling the source channel directly at source, by means of the cunning remote control system, so when you fade down your VCA subgroup, all the post-fade effects sends on the source channels are automatically turned down by the same amount, just as they would be if you went and adjusted all the local channel faders individually.

2. The second advantage of VCA groups is when you want to create a stereo group, ie, the sources you want to group are all panned differently into a stereo image (eg backing vocals or a brass section). You want your subgroup master fader to control the levels of the sources, but still maintain the stereo image you have set. The only way to do this with audio subgroups is to use two group busses as a stereo group, you then route the first group master to the left main mix bus and the second group master to the right main mix bus. This works fine, but you need to use two valuable group faders for one subgroup, and you have to push two faders up and down whilst keeping them at a matched level.

   With VCA grouping, you only need to use one group fader for a stereo group, as the routing of the source signals is still going directly to the main mix bus via the various pan settings of the source channels, and all you are doing with the VCA master is effectively pushing the input faders up and down using the voltage control.

3. The third advantage of using VCA groups is that you are not extending the signal path by routing the channel signals through another set of audio amplifiers which are needed for an audio subgroup, and so are avoiding the increase in noise and other degradations that go along with that. The VCA element in the channel itself does have a slight effect on the signal, but with modern chips and careful circuit design this is insignificant.

back to the top

-----------------------------------------------------------------------------------------------------------------------------------------------------------

## Why bother with audio subgroups any more if VCA groups are so great?

There are still a couple of reasons why audio groups are still essential on any live console. Firstly, if you want to apply compression or other dynamics processing to a group of signals, rather than each individual source (because you haven't got enough dynamics processors for everything). Routing to an audio group and then inserting the processor is the only way to achieve this. VCAs cannot do this, as there is no separate audio signal for the group which can be accessed for processing - the audio always remains part of the main mix with VCAs.

Secondly, if you want to use Matrix Outputs to create different mixes for feeding to differerent zones or areas of a venue for example, the sources must be routed to audio subgroups first, before each group is then routed with varying level, to the various zones (Matrix

Outputs). So what you can see here is that it is not possible to route signals directly from an input channel to a matrix output - you have to go via an audio subgroup first. This system of deriving matrix outputs has become standard functionality on any console you can buy which has a matrix capability, and it grew up this way partly because it is impractical to provide any more sends from channels other than the normal 'aux' sends (there just isn't space on the channel), but more importantly because it makes sense to organise sources into groups before routing to matrix outs, and the whole thing will be easier to control.

Finally, audio groups can come in useful if they have an associated output meter (most do), as a way of metering the combined level of a VCA group. VCA groups themselves cannot be metered -because there is no audio output signal from a VCA group (remember, the audio remains in the channels and goes straight to the mix bus). But if you have assigned a set of VCA groups and you want to meter them, a neat trick is to also assign corresponding sets of audio subgroups from the same channels. The subgroup master faders are left at unity gain and are NOT routed to the main mix bus -so their only function is to display the levels of the combined audio signals on their output meters (the grouped signals don't actually go anywhere). As you mix up and down on your VCA master faders, the audio group meters will show the corresponding variation in the level of the combined signals routed to them, which if you have assigned correctly, will correspond to your VCA groups (ie for example, drum kit assigned to VCA group 1 AND routed to audio subgroup 1, vocals assigned to VCA group 2, AND routed to audio subgroup 2, etc) In fact, it's a little more complicated than this, because if your source channels are panned in stereo, you will need to use pairs of audio groups for each VCA group, in order to meter them correctly in stereo, but this gives the general idea.

back to the top

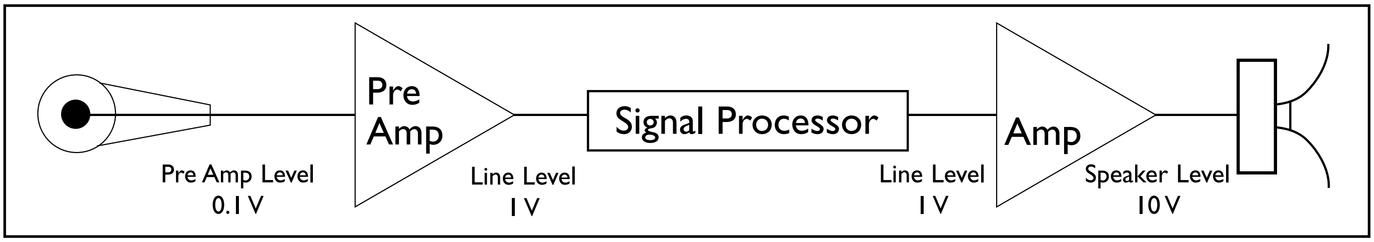
--------------------------------------------------------------------------------------------------------------------------------------------------------

# SIGNAL FLOW

**Marty McCann**



***You must understand Signal Flow in order to fully comprehend how a sound reinforcement system functions. Understanding Signal Flow lets you know just which components connect to which and in what order, so that you can hook up the sound system correctly each time.***
***I am going to teach you about signal flow by introducing you to the Audio Chain, or the sequence in which your sound reinforcement system's individual components, connect together.***




The audio chain begins with the microphone. A microphone is a device that changes energy from one form to another. A device that changes energy from one form to another is called a transducer. The microphone is a transducer that changes acoustical energy (sound) into electrical energy (voltage) . The acoustical energy may be the sound of someone's voice or that of a musical instrument.

The sound pressure or acoustical energy arrives at the diaphragm of the microphone and causes it to move or vibrate. The movement of the diaphragm converts the acoustical sound into an electrical signal. This electrical audio signal is now an electrical analogue of the acoustical sound. This electronic analogue is a corresponding electrical representation of the variations in loudness (level), pitch or timbre (frequency content) of the original audio sound striking the microphone's diaphragm.The microphone produces a very small electrical signal. A dynamic microphone that is picking up a loud voice or instrument may produce an electrical signal somewhere in the neighborhood of 0.1 volt.

Voltage is a measure of electrical pressure which determines the amount of potential to perform a task or accomplish some work. Since the microphone level is relatively small, too small to drive the power amplifier, it must be raised in signal level (to a higher potential or voltage) by means of the systems pre-amplifier (pre-amp).

A pre-amp can accept the small voltage that the microphone produces (pre-amp level of 0.1 volt) at its input, and increase or amplify the electrical signal to the next higher magnitude of audio signal level called a line level. Line levels are in the neighborhood of 1 volt when a loud sound pressure wave strikes the diaphragm of the microphone. When we use more than one microphone, we need more individual pre-amplifiers or something called the sound system mixer.

The second component in the audio chain is the audio mixing console. A mixer is nothing more than a number of individual microphone pre-amplifiers in a single package that provides signal gain (amplification) and summing (mixing) of the individual preamp channels and produces a line level signal at the output of the mixer.

The first mixer that I ever used in sound reinforcement consisted of four pre-amp inputs and one line level output. It had volume one, volume two, volume three, volume four, and master volume. It had no tone controls, no monitor or effects sends, just four individual pre-amp channel level controls and a master output level.

My first eight channel mixer consisted of two of these puppies, and my first sixteen channel mixer was made up of four of these units bused or connected together. Today's mixing consoles are very sophisticated with all of their features and functions. In addition to input gain, equalization, monitor and effects sends, we can find: signal inserts, assignment buttons, submaster and master sections, along with multiple auxiliary signal buses, PFL's, LED's, etc.

However, the primary purpose of the mixer is still to raise each of the individual channels, pre-amp signal levels (0.1 volt) to line levels (1 volt), and to sum or join these pre-amp channels together on a common mixing bus or assigned signal routing that appears as a line level output of the mixing console.

The third component making up the audio chain can be one of many signal processors. A signal processor does something other to the electrical audio signal than simply amplify or raise the level of the signal. Pre-amps and power amps amplify or raise the voltage level or gain of the entire audio spectrum of the signal. Signal processors may affect or alter the frequency response. They may gate, compress, limit, or expand the audio signal. They may perform modulated time delays or special effects, such as adding flanging, phasing, chorusing, reverb, delay, or echo to an audio signal.

The microphones and loudspeakers are transducers that change energy in the sound system from one form to another. The mixer (pre-amp) and power amplifiers raise signal level or increase gain. All other components making up the sound system fall then into a very general category known as signal processors, because they further process the electrical audio signal in some manner.

A graphic equalizer is a type of signal processor that takes the audio frequency spectrum (twenty cycles per second to twenty thousand cycles per second) and breaks these audio frequencies down into a number of discrete bands of frequencies, which can be individually raised (boost) or lowered (cut) in signal level. Some packaged PA sound systems may have five to nine bands of equalization on their graphic equalizer (GEQ).

In professional sound reinforcement there are one octave graphic equalizers (10 band EQ), two-thirds octave equalizers (15 band EQ), and one-third of an octave graphic equalizers (28 - 31 band EQ).

A parametric equalizer is a type of signal processor that enables the sound person to attenuate (cut) or increase (boost) the gain of a selected band of frequencies that can be wide or narrow in spectrum. The parametric EQ uses three controls to adjust these parameters. The frequency control selects the center of the frequency band. The bandwidth control adjusts the Q of the filter, or how wide or narrow the band of frequencies will be. The level control increases (boosts) or decreases (cuts) the signal level (gain) of the selected band of frequencies.

An electronic crossover is another form of a signal processor. The electronic crossover (x-over) takes the audio spectrum and divides it into a number of separate bands of frequencies called bandpasses. Each separate bandpass is then amplified independently in order to drive separate loudspeaker components, each of which reproduce a band of frequencies.

When bi-amping, the audio spectrum is separated into two bands of frequencies by the electronic crossover. All of the frequencies below the selected crossover point (x-over frequency) are called low pass (short for low frequency bandpass), those frequencies above the crossover frequency, are called high pass (high bandpass). Each band of frequencies is then amplified by its own power amplifier (low pass amp and high pass amp), resulting in increased clarity and performance. In sound reinforcement it is possible to operate a sound system 2-way (bi-amp), 3-way (tri-amp), 4-way (quad-amp), or 5-way (pent-amp?).

Compressors and limiters are types of signal processors that reduce or limit the level of the signal, thus reducing the dynamic range of the signal. Dynamic range is the difference measured in decibels between the lowest signal levels (very soft or pianissimo) and the highest (very loud or fortissimo) signal levels that the system can accommodate.

A noise gate is a signal processor that attenuates (reduces the signal level a great deal), or even turns off (gates) the audio signal passing through it, when that signal level falls below a minimum threshold that is adjustable by the user.

A sub category under signal processors are those processors known as effects processors. Reverb, digital delay, flanging, phasing, chorusing, and other modulated time delays are all performed by the effects processor. These effects can be available as individual processors or even in an all-in-one multi-effects unit. There are also effects processors called psycho-acoustical enhancers that generate amplitude and frequency dependent harmonics that are perceived as an increase in the apparent loudness of the signal.

Signal processors that are used in sound reinforcement are those types that can accept line level signals (1 volt on the average but up to 8 or even 10 volts maximum) at their inputs, and produce line level signals at their output. There are some signal processors that were designed for pre-amp or low level signals (0.1 volt) such as the output of an electric guitar or bass instrument. Usually you can hold these devices in one hand, and they operate off of nine volt batteries or a low voltage power supply.

These pre-amp level signal processors usually cannot accept line levels (1 volt or greater) at their inputs. They break up or distort when the line level signal overloads their input circuitry. Also, they cannot produce the maximum line level at their output. This means that they cannot drive subsequent line level components, resulting in reduced signal level and lack of headroom. Headroom is the difference between the average operating level and the point at which the signal is clipped or distorted within the device.

Some signal processors that are used for sound reinforcement can also be used with musical instruments directly. These units have switchable or programmable input and output operating levels, which enable them to accommodate either pre-amp or line level signals.

In the past, some signal processors have had different input and output jacks to allow the device to be used at either operating level. If the jack or switch is labeled -10 dB or even -20 dB, that would be the pre-amp mode. If the jack or switch position is labeled 0 db or +4 dB, that is the line level mode of operation. It is very important for the sound system operator to assign the unit's input and output for the correct operating level.

There is usually a difference in application between normal signal processors and those that do special effects. More often special effects signal processors are used in what is called a "side chain" operation. In a side chain patch, the signal is taken out of the mixer and sent to the effects signal processor. Then, from the effects processor's output, the signal is returned to the mixer itself (usually returned to a channel, although it can be returned via of an effects return or auxiliary return). Standard signal processors (non-special effect) are inserted directly in the line of the electrical signal flow.

Effects mixing buses in consoles are normally set up to send a separate mix of signals from any of the individual channels to the chosen effects processor. This "Effects Send" is post (after) the channel slider or level control. Increasing the level of the channel in the front of house (FOH) or main mix causes the level going to the effects processor to increase likewise. So, as the operator increases the level of the channel slider, there is also a corresponding increase in the level of the effects send signal from the channel. In these side chain applications, the effects processor is usually run strictly "wet" (all effect). This results in the effects mix tracking in level directly along with the dry signal of the channel that is routed directly to the house mix.

After the signal processors, the fourth component in the sound reinforcement audio chain is the power amplifier. The power amplifier, as the name implies, delivers power to the sound system loudspeaker in the form of higher levels of voltage and current.

The power amp accepts line levels (1 volt) at its input, and produces speaker levels at its output. Speaker levels are in the neighborhood of 10 volts when a loud voice or instrument is reinforced by the sound system.

The power amplifier's output voltage capability can be 40 volts for a 400 Watt amplifier (P = V x V / R, 40 volts x 40 volts / 4 ohms = 1600 / 4 = 400 Watts. A more powerful amplifier will produce even more than 40 volts at its output.

Up until now, our electronic audio components have dealt with pre-amp and line level signal or small voltage signal levels. When producing larger speaker levels, the power amplifier also sends a respectable amount of current to the loudspeaker.

Current is the rate of electron flow, measured in amperes (amps). Current is what actually causes work to be accomplished. Prior to the power amplifier, we have not tried to do any work with the audio signal, so we haven't had a high level of current flow.

Now that the power amplifier is driving the loudspeaker and converting this electrical energy into acoustical energy, the power amp must deliver significant amounts of current to the speaker. A 400 Watt amplifier will deliver 10 amperes of current to a loudspeaker load of 4 ohms (P = I x E, 10 amps x 40 volts = 400 watts).

Because it is the current delivered to the loudspeaker that gets the work done, the power amplifier's electrical power cord should be connected directly to the House AC Mains.

Do not use some light gauge electric "hedge clipper" type of extension cord. If you must use electrical extension cords to reach the power amplifier, make sure that they are the heavy duty, heavy gauge, grounded type of electrical power extension cord.

The fifth and final component in the audio chain is the loudspeaker. Like the microphone, the loudspeaker is a transducer that changes energy from one form into another. The loudspeaker changes the electrical audio signal back into an acoustical signal of sufficient loudness (audio level) to enable it to be heard by the distant listeners.

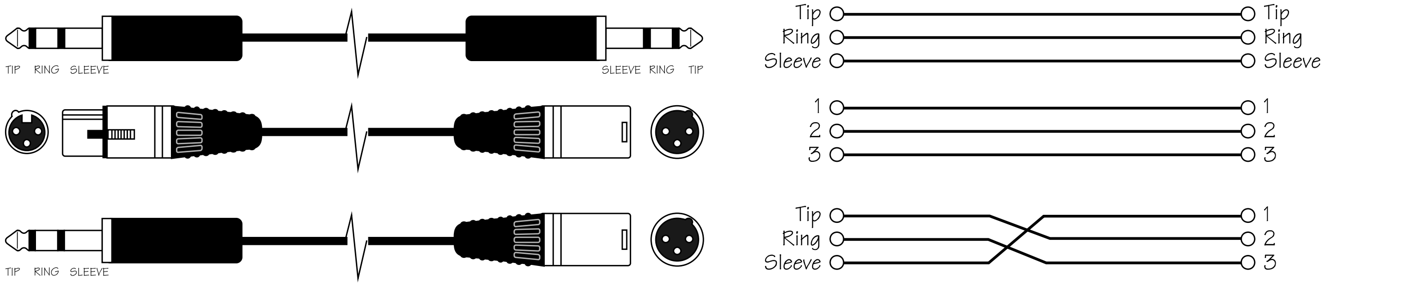
The loudspeaker must be capable of handling the high voltage and current produced by the sound systems power amplifier. The loudspeaker offers resistance or opposition to the flow of current from the power amp. Since this resistance to current flow varies with frequency, we use the term impedance when discussing the loudspeaker's opposition to current flow.

Impedance takes into account the types of resistance offered to a voltage that is alternating in its direction of current flow. Alternating Current (AC) encounters a different type of opposition to the rate of current flow through the loudspeaker's voice coil. Music or speech creates a signal of alternating current, which changes in direction twice for each audio cycle as opposed to a direct current or DC which does not change direction. The loudspeaker system can offer both an inductive and a capacitive form of opposition to the flow of current from the power amplifier.

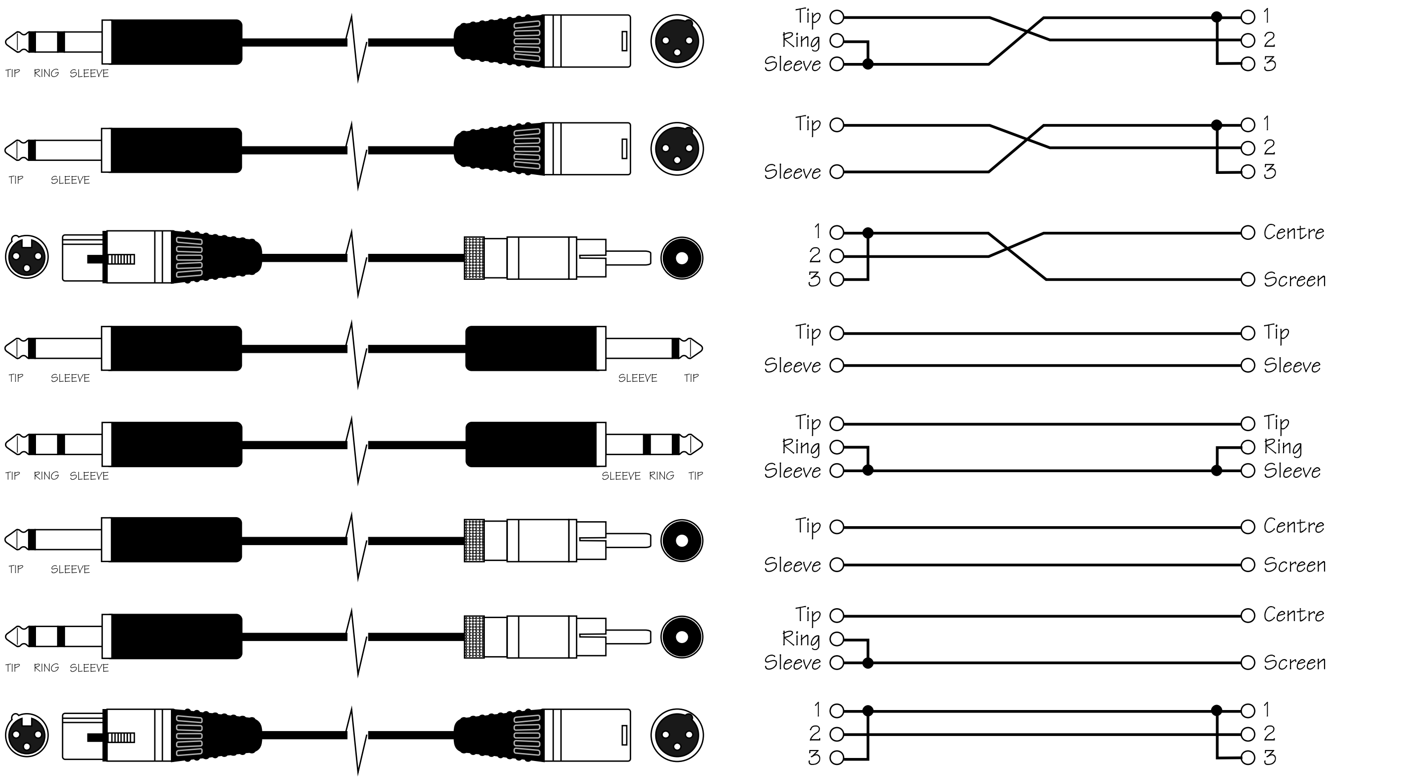
The five components that I have just covered make up the complete audio system or chain of audio components. The direction of the flow of the audio signal through this audio chain of components is called signal flow. When some people see a large sound system they are intimidated by its apparent complexity. The most complicated sound reinforcement system can be broken down into the five basic components that I have just covered. The large system contains the same five basic components, there are just more of them.

I hope this article on signal flow helps you to better understand your sound reinforcement system.

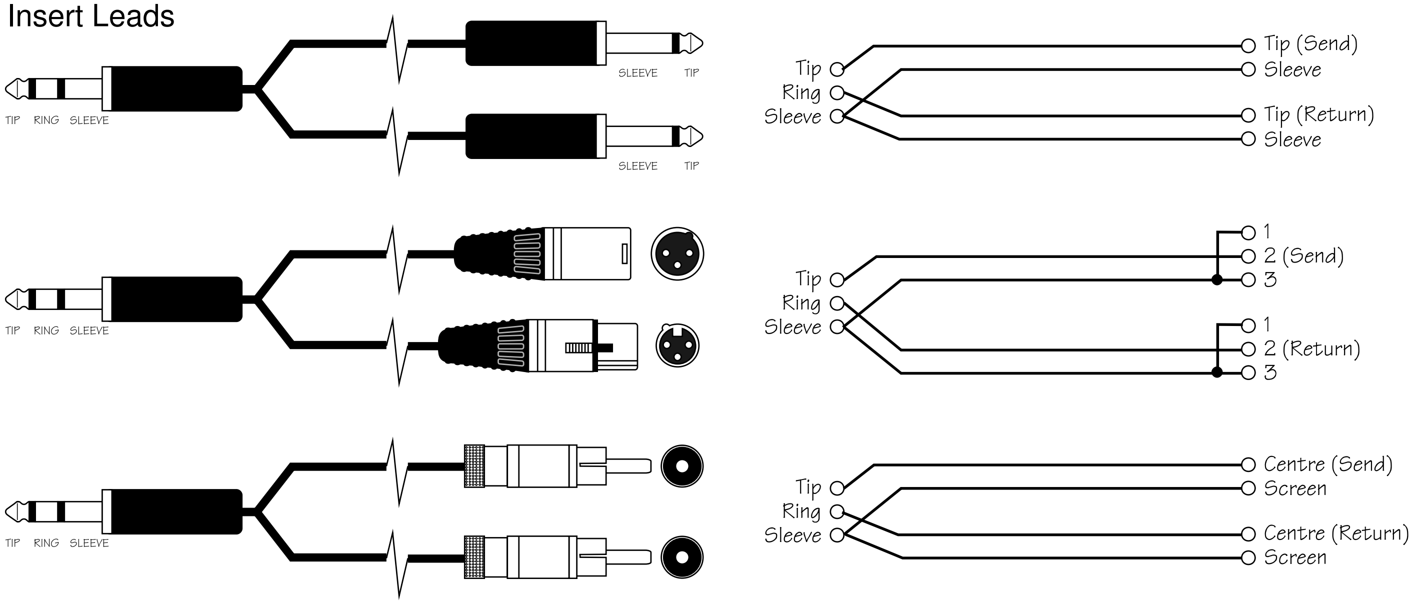
## Balanced




## Unbalanced




## Insert Leads



# Unity Gain & Impedance Matching: Strange Bedfellows

- **Unity Gain and Balancing**
- **Impedance Matching**
- **Cross-Coupled Output Stages**

**Dennis Bohn**
**Rane Corporation**

**RaneNote 124**


## INTRODUCTION

This paper discusses the pitfalls (often subtle) of our industry's failure to define and standardize what "unity gain" means, and the conditions necessary to measure it. It further discusses how people improperly use one piece of misinformation (impedance matching) to correct for this lack of standardization. All done, without knowing discrepancies exist between different pieces of equipment, and without knowing impedance matching is unnecessary, signal degrading, and wasteful.

For me, it began with a phone call. The caller said he wanted to know our output impedance so he could add the proper load impedance.

"Why would you want to do such a thing?" I asked.

"Because I want to maintain unity gain through each piece of signal processing gear," he replied.

That gave me pause. Then I laughed and realized what he was doing right, and what he was doing wrong.

The problem stems from another case of our industry working without proper guidelines and standards. This one involves the conditions used to establish unity gain. Lately, the popular trend of including unity gain detent points and reference marks only aggravates things.

This Note identifies and explains the problem. Once understood, the solution becomes easy – and it doesn't involve impedance matching.

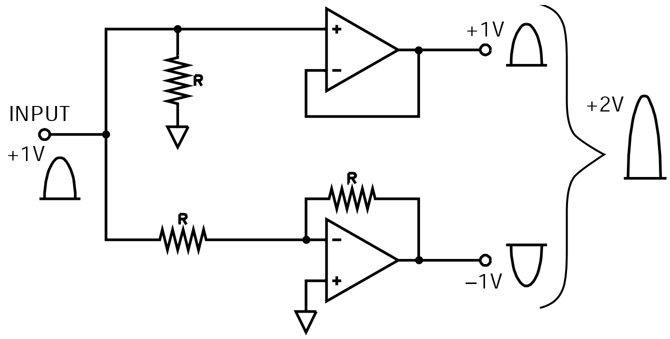



Figure 1. Differential line driving circuit.

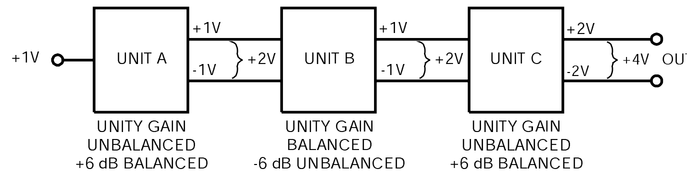



Figure 2. Example of signal increase due to different unity gain definitions.

## UNITY DOES NOT ALWAYS EQUAL ONE

It begins with an understanding of unity gain. Simple enough. Ask anyone and they will tell you unity gain means that if I put, say, 1 V in, I get 1 V out, i.e., a gain of one, or unity. Nothing could be easier. That is until that same someone asks the question,

"Is that unity gain balanced, or unbalanced?"

Herein lies the problem. Today we find that many (most?) pro audio signal processors have a gain difference of 6 dB between unbalanced and balanced out (exceptions to this are units with output transformers, or cross-coupled output stages – see Appendix). This x2 difference results from differentially driving the line. Figure 1 shows how an input signal drives one side of the line positively and the other side negatively (each line driving amplifier has a gain of one, but together they yield a gain a two). For example, a +1 V peak AC input signal drives one side to +1 V while simultaneously driving the other side to -1 V. This gives a balanced output level of +2 V peak (the *difference* between +1 V and -1 V). Alternatively, that same input signal drives an unbalanced line to +1 V peak. Thus, there is a 6 dB disparity between an unbalanced and a balanced output — a gain difference factor of two.

Here, unity equals two.

## NO STANDARDS

This brings us to the part about no standards. Without a standard defining the specified conditions for unity gain, manufacturers make their own decision as to what "unity gain" means. For one, it means 1 V in gives 1 V out unbalanced, and 2 V out balanced. For another, it means 1 V in gives 1 V out balanced, and ½ V out unbalanced. For yet others, it means 1 V in gives 1 V out (using transformers), or 2 V out (using cross-coupled stages), either balanced *or* unbalanced. Very confusing.

Figure 2 shows how this creates problems. Here different definitions result in a gain of 12 dB, *with all controls seemingly set for unity*.

## IMPEDANCE MATCHING

Impedance matching went out with vacuum tubes, Edsels and beehive hairdos. Modern transistor and op-amp stages do not require impedance matching. If done, *impedance matching degrades audio performance*.

Modern solid-state devices transfer *voltage* between products, not *power*. Optimum power transfer requires impedance matching. Optimum voltage transfer does not. Today's products have high input impedances and low output impedances. These are compatible with each other. Low impedance output stages drive high impedance input stages. This way, there is no loading, or signal loss, between stages. No longer concerned about the transfer of power, today's low output/high input impedances allow the almost lossless transfer of signal voltages.

What then, does impedance matching have to do with unity gain? Well, it shouldn't have anything to do with it. But because of different manufacturer's definitions, it is one way (brute force) of correcting gain discrepancies between products. *Impedance matching introduces a 6 dB pad between units*. Let's see how this works.

Look at Figure 3. Here we see a real world interface between two units. The positive and negative outputs of the driving unit have an output impedance labeled $R_{OUT}$. Each input has an impedance labeled $R_{IN}$. Typically these are around 100 Ω for $R_{OUT}$ and 20k Ω for $R_{IN}$. Georg Ohm taught us that 100 Ω driving 20k Ω (looking only at one side for simplicity) creates a voltage divider, but a very small one (-0.04 dB). This illustrates the above point about achieving almost perfect voltage transfer, *if impedance matching is not done*.

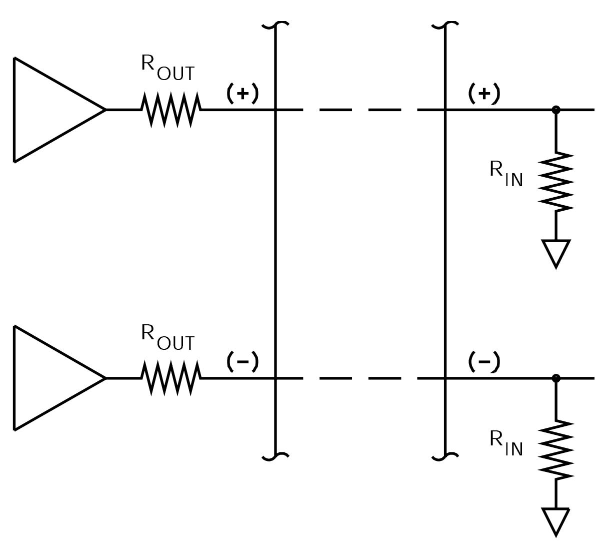



Figure 3. Balanced wiring interconnection between units.

If it is done, you lose half your signal. Here's how: impedance matching these units involves adding 100 Ω resistors (equal to $R_{OUT}$) to each input (paralleling $R_{IN}$). The new input impedance now becomes essentially the same as the output impedance (100 Ω in parallel with 20 kΩ equals 99.5 Ω), therefore matching. Applying Ohm's law to this new circuit tells us that 100 Ω driving 100 Ω creates a voltage divider of ½. That is, ½ of our signal drops across $R_{OUT}$ and ½ drops across $R_{IN}$, for a voltage loss of 6 dB. We lose half our signal in heat across $R_{OUT}$. Not a terribly desirable thing to do; yet, it does fix our unity gain problem.

Back to Figure 2. By selectively impedance matching only between Units A and B, we introduce a 6 dB pad. This cancels the 6 dB gain resulting from using balanced outputs with this unit. This changes the output of Unit A to ±½ V, or +1 V balanced. Since Unit B already is unity gain balanced, then we do *not* impedance match, and its output is also ±½ V. We do impedance match Unit C's output and now Unit C passes this +1 V signal to its output as ±½ V, and finally we get a true unity gain result from all three boxes. One volt in, produces one volt out — balanced.

## PREFERRED ALTERNATIVE TO IMPEDANCE MATCHING

The preferred alternative to impedance matching is ridiculously easy — turn the level control down 6 dB. Of course this means the unity gain mark, or detent position, loses its meaning, but this is far better than losing half your signal.

Many users do not view this issue as a problem. There are so many other variables that require turning level controls up or down that this just becomes part of the overall system gain setting. Most units have sufficient headroom to allow for an unexpected 6 dB of gain without hurting anything.

Besides, the unity gain mark/detent is only a reference point. The whole reason manufacturers give you level controls, is to allow setting the gain you need for your system. If it were important for them to remain at unity, they would not be there. They are yours. You paid for them. Use them.

## SUMMARY

Unity gain and impedance matching: a strange dichotomy. One solves the other, but badly.

Impedance matching is not necessary and creates many ills. It reduces signal levels and dynamic range by 6 dB (and possibly signal-to-noise by the same amount). The large currents necessary to drive the low matching-impedance usually degrades total harmonic distortion. And the extra current means excess heat and strain on the power supply, creating a potentially unreliable system.

Simply turning the level control down (or up, as the situation dictates), is the best solution for unity gain disparities.

**Rane's Standard:**

***"Unity Gain" is defined as 'balanced in' to 'balanced out'.* For unbalanced units, "unity gain" is modified to mean 'balanced or unbalanced in' to 'unbalanced out'.**

## APPENDIX: UNDERSTANDING CROSS-COUPLED OUTPUT STAGES

Cross-coupled output stages have been around for a long time[1]. So has their marketing rhetoric. Some of the many grand claims are even true. Understanding cross-coupled output stages begins with the following: *The only purpose of cross-coupling techniques is to mimic an output transformer under unbalanced conditions. They offer no advantages over conventional designs when used balanced.* Understanding cross-coupled *circuitry* begins with an understanding of output transformers (Figure 4). Here we see a typical configuration. The output amplifier drives the primary winding of the transformer (with one side grounded), and the secondary winding floats (no ground reference) to produce the positive and negative output legs of the signal. An output transformer with a turns ratio of 1:2 (normal), produces a 2 V output signal for a 1 V input signal, i.e., there exists a difference of potential between the two output leads of 2 V. The diagram shows how a 1 V peak input signal produces ±1 V peak output signals (relative to ground), or a differential floating output of 2 V peak. (Alternatively, two op-amps could differentially drive the primary; and use a turns ratio of 1:1 to produce the same results.)

So, 1 V in, produces 2 V out – a gain of 6 dB. Simple. Note that because the output signal develops across the secondary winding, it does not matter whether one side is grounded or not. Grounding one side gives the same 2 V output. Only this time it references to ground instead of floating. *There is no gain change between balanced and unbalanced operation of output transformers*.

Contrast this with the active output stage of Figure 1. Here, grounding one side reduces the output from 2 V to 1 V. Though this is a one time gain reduction (correctable by increasing the level 6 dB), it bothers some. Mark off points.

Of more concern is the 6 dB lost of headroom. A desirable aspect of differentially driving interconnecting lines is the ability to get 6 dB more output level from the same power supply rails. Most audio products use op-amps running from ±15 V rails. A single op-amp drives an unbalanced line to around ±11 V peak (+20 dBu). Using two op-amps to drive the line differentially doubles this to ±22 V peak (one goes positive, while the other goes negative), a value equal to +26 dBu. Mark off more points for loss of headroom.

Mark off even more points for potential distortion (depends on op-amps, and exact configuration), oscillation,

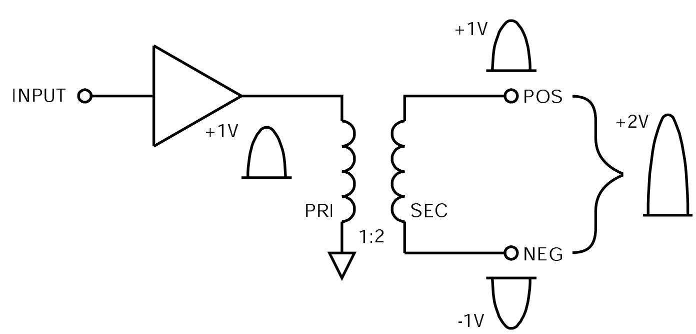



Figure 4. Balancing output transformer

and failure, resulting from asking one side to drive a short (the result of grounding one side for unbalanced operation).

These three things: 6 dB loss of gain, 6 dB loss of headroom, and the questionable practice of allowing an op-amp to drive a short, sparked creation of the cross-coupled output stage. It solves two out of three.

Cross-coupled output stages do two things active differential output stages do not. They maintain the same gain either balanced or unbalanced. And they protect themselves from having to drive a short. But *they still have 6 dB loss of headroom*.

A point not understood by many users. They believe that cross-coupled output stages behave exactly like transformers. Not true. They have the same headroom limitation as all op-amp designs operating from ±15 V power supplies. (Some equipment uses ±18 V, but this only results in a 2 dB difference for unbalanced.)

MCI's original design[1] appears in Figure 5. MCI used two op-amps, wired such that the opposite output subtracted from twice the input signal (not particularly obvious, but true). That way, each side's gain looks like a gain of one for balanced operation, i.e., 1 V in, gives ±1 V out. Yet shorting one side (running unbalanced) gives a gain of two (nothing to subtract).

Since cross-coupled and normal differential output stages use essentially the same parts (and therefore cost the same), a fair question is why don't you see more of the former? The answer lies in the perils of positive feedback.

Inherent to the operation of cross-coupled output stages is positive feedback. The subtraction process created by cross-coupling opposite outputs, has an undesirable side effect of being positive feedback. Because of this, op-amp matching, resistor ratio matching and temperature compensation becomes critical. If not done properly, cross-coupled stages drift and eventually latch-up to the supply rails. (This is why you see so many variations of Figure 5, with all sorts of excess baggage glued on. Things like capacitive-coupled AC feedback, fixed loading resistors, high-frequency gain roll off capacitors, offset trims, etc.) The difficulty in controlling these parameters in high volume production, leads most manufacturers to abandon its use.

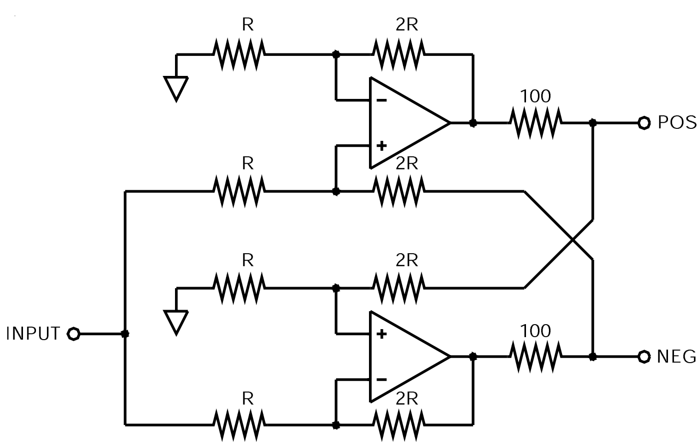



Figure 5. MCI cross-coupled output circuit[1].

Recently, Analog Devices helped solve these problems by putting all the elements into one integrated circuit[2]. Their monolithic IC version (which Rane uses in select products) operates on the same principles as MCI's, although Analog Devices uses three op-amps to drive the input differentially. Here precise control and laser trimming guarantees stable performance, and opens up a new chapter in cross-coupled output stage use.

## REFERENCES

1. T. Hay, "Differential Technology in Recording Consoles and the Impact of Transformerless Circuitry on Grounding Technique," presented at the 67th Convention of the Audio Engineering Society, *J. Audio Eng. Soc. (Abstracts)*, vol. 28, p. 924 (Dec. 1980).
2. "SSM-2142 Balanced Line Driver," *Audio/Video Reference Manual*, pp. 7-139 (Analog Devices, Norwood, MA, 1992).

Reproduced with permission from *S&VC*, vol. 8, no. 9, Sept. 20, 1990, pp. 10-20.

# Setting Sound System Level Controls

- Decibel: Audio Workhorse
- Dynamic Range: What s Enough?
- Headroom: Maximizing
- Console/Mic Preamp Gain Settings
- Outboard Gear I/O Level Controls
- Power Amplifier Sensitivity Controls
- Active Crossover Output Attenuators
- Using the RaneGain  Test Set



**Dennis Bohn**
**Rane Corporation**

**RaneNote 135**


## IMPORTANCE

Correctly setting a sound system's gain structure is one of the most important contributors to creating an excellent *sounding* system. Conversely, an improperly set gain structure is one of the leading contributors to *bad* sounding systems. The cost of the system is secondary to proper setup. The most expensive system set wrong never performs up to the level of a correctly set inexpensive system. Setting all the various level controls is not difficult; however, it remains a very misunderstood topic.

The key to setting level controls lies in the simple understanding of *what* you are trying to do. A few minutes spent in mastering this concept makes most set-ups intuitive. A little common sense goes a long way in gain setting.

A dozen possible procedures exist for correctly setting the gain structure of any system. What follows is but one of these, and is meant to demonstrate the *principles* involved. Once you master the fundamental principles, you will know what to do when confronted with different system configurations.

## DECIBELS, DYNAMIC RANGE & MAXIMIZING HEADROOM

Audio-speak is full of jargon, but none so pervasive as the *decibel.* Those unfamiliar or rusty with decibel notation, and its many reference levels, are directed to the sidebar offered as review or introduction. Mastering gain, or level control settings also requires an understanding of *dynamic range* and *headroom*.

**Dynamic range** is the ratio of the loudest (undistorted) signal to that of the quietest (discernible) signal in a piece of equipment or a complete system, expressed in decibels (dB). For signal processing equipment, the maximum output signal is ultimately restricted by the size of the power supplies, i.e., it cannot swing more voltage than is available. While the minimum output signal is determined by the noise floor of the unit, i.e., it cannot put out a discernible signal smaller than the noise (generally speaking). Professional-grade analog signal processing equipment can output maximum levels of +26 dBu, with the best noise floors being down around -94 dBu. This gives a maximum *unit dynamic range* of 120 dB — a pretty impressive number coinciding nicely with the 120 dB dynamic range of normal human hearing (from just audible to painfully loud).

For sound systems, the maximum loudness level is what is achievable before *acoustic feedback*, or system squeal begins. While the minimum level is determined by the overall background noise. It is significant that the audio equipment noise is usually swamped by the HVAC (heating, ventilating & air conditioning) plus audience noise. Typical minimum noise levels are 35-45 dB SPL (sound pressure level), with typical loudest sounds being in the 100-105 dB SPL area. (Sounds louder than this start being very uncomfortable, causing audience complaints.) This yields a typical useable *system dynamic range* on the order of only 55-70 dB — quite different than unit dynamic ranges.

Note that the dynamic range of the system is largely out of your hands. The lower limit is set by the HVAC and audience noise, while the upper end is determined by the comfort level of the audience. As seen above, this useable dynamic range only averages about 65 dB. Anything more doesn't hurt, but it doesn't help either.

**Headroom** is the ratio of the *largest* undistorted signal possible through a unit or system, to that of the *average* signal level. For example, if the average level is +4 dBu and the largest level is +26 dBu, then there is *22 dB of headroom*.

Since you can't do anything about the system dynamic range, your job actually becomes easier. All you need worry about is *maximizing unit headroom*. But how much is enough?

An examination of all audio signals reveals music as being the most dynamic (*big surprise*) with a *crest factor* of 4-10.

Crest factor is the term used to represent the ratio of the peak (crest) value to the *rms* (*root mean square* — think *average*) value of a waveform. For example, a sine wave has a crest factor of 1.4 (or 3 dB), since the peak value equals 1.414 times the rms value.

Music's wide crest factor of 4-10 translates into 12-20 dB. This means that musical peaks occur 12-20 dB higher than the "average" value. This is why headroom is so important. *You need 12-20 dB of headroom in each unit to avoid clipping.*

## PRESET ALL LEVEL CONTROLS IN THE SYSTEM

After all equipment is hooked-up, verify system operation by sending an audio signal through it. *Do this first before trying to set any gain/level controls.* This is to make sure all wiring has been done correctly, that there are no bad cables, and that there is no audible hum or buzz being picked up by improperly grounded interconnections (See the *Sound System Interconnections* RaneNote).

Once you are sure the system is operating quietly and correctly, then you are ready to proceed.

Turn down all power amplifier level/sensitivity controls.

Turn *off* all power amplifiers. (This allows you to set the maximum signal level through the system without making yourself and others stark raving mad.)

Set all gain/level controls to their *off* or *minimum* settings.

*Defeat all dynamic controllers such as compressors/ limiters, gate/expanders, and enhancers* by setting the Ratio controls to 1:1, and/or turning the Threshold controls way up (or down for gate/expanders).

Use no *equalization* until after correctly setting the gain.

## CONSOLE/MIC PREAMP GAIN SETTINGS

A detailed discussion of how to run a mixing console lies outside the range of this Note, but a few observations are relevant. *Think about the typical mixer signal path*. At its most basic, each input channel consists of a mic stage, some EQ, routing assign switches and level controls, along with a channel master fader. All of these input channels are then mixed together to form various outputs, each with its own level control or fader. To set the proper mixer gain structure, you want to maximize the overall S/N (signal-to-noise) ratio. *Now think about that a little:* because of the physics behind analog electronics, each stage contributes noise as the signal travels through it. (Digital is a bit different and is left to another Note and another day.) Therefore each stage works to *degrade* the overall signal-to-noise ratio. Here's the important part: *The amount of noise contributed by each stage is* (relatively) independent of the signal level passing through it. So, the bigger the input signal, the better the output S/N ratio (in general).

The rule here is to take as much gain as necessary to bring the signal up to the desired average level, say, +4 dBu, *as soon as possible.* If you need 60 dB of gain to bring up a mic input, you don't want to do it with 20 dB here, and 20 dB there, and 20 dB some other place. You want to do it all at once at the input mic stage. For most applications, *the entire system S/N (more or less) gets fixed at the mic stage.* Therefore set it for as much gain as possible without excessive clipping. Note the wording *excessive* clipping. A little clipping is not audible in the overall scheme of things. Test the source for its expected *maximum input level*. This means, one at a time, having the singers sing, and the players play, as loud as they expect to sing/play during the performance. Or, if the source is recorded, or off-the-air, turn it up as loud as ever expected. Set the input mic gain trim so the mic OL (overload) light just occasionally flickers. *This is as much gain as can be taken with this stage.* Any more and it will clip all the time; any less and you are hurting your best possible S/N.

(Note that a simple single mic preamp is set up in the same manner as a whole mixing console.)

## OUTBOARD GEAR I/O LEVEL CONTROLS

All outboard unit level controls (except active crossovers — see below) exist primarily for two reasons:

- They provide the flexibility to operate with all signal sizes. If the input signal is too small, a gain control brings it up to the desired average level, and if the signal is too large, an attenuator reduces it back to the desired average.
- Level controls for equalizers: the need to provide make-up gain in the case where significant cutting of the signal makes it too small, or the opposite case, where a lot of boosting makes the overall signal too large, requiring attenuation.

Many outboard units operate at "unity gain," and do not have *any* level controls — what comes in (magnitude-wise) is what comes out. For a perfect system, *all* outboard gear would operate in a unity gain fashion. *It is the main console's (or* preamp's) job to add whatever gain is required to all input signals*.* After that, all outboard compressors, limiters, equalizers, enhancers, effects, or what-have-you need not provide gain beyond that required to offset the amplification or attenuation the box provides.

With that said, you can now move ahead with setting whatever level controls *do* exist in the system.

Whether the system contains one piece of outboard gear, or a dozen, gains are all set the same way. Again, the rule is to maximize the S/N through each piece of equipment, thereby maximizing the S/N of the whole system. And that means setting things such that your *maximum system signal* goes straight through every box without clipping.

### RaneGain Test Set

The RaneGain (RG) test set is a handy tool kit based on techniques first developed by Pat Brown of *Syn-Aud-Con* for use in quickly setting sound system gain controls. It consists of two pieces: a self-contained, phantom-powered 400 Hz generator and a separate audio Transducer housed in an XLR connector. The RG Generator plugs into any mic input on a mixing console (or separate mic preamp) having phantom power in the range of 12-48 VDC, providing a convenient sound source. The RG Transducer plugs into the output of each unit and sounds a warning whenever the output level is clipped. See the RaneGain data sheet (*www.rane.com/ranegain.html*) for additional details.

## SETTING SIGNAL PROCESSING LEVEL CONTROLS

First, a sound source is connected to the mixing console (or separate mic preamp) to provide the *maximum system signal* output, then this signal is used to set the outboard units.

The most convenient sound source is one built into the mixer or preamp. If a built-in generator is available, use that; if not, use an external oscillator, such as the RaneGain generator or other test equipment. Connect the generator to an unused channel in the mixing console or to the input of the mic preamp. Carefully set the generator level and the channel input fader so the mic stage does not overload. Next, adjust the master output fader (or preamp output level control) for the largest level possible without clipping the output stage. Determine this maximum level using any of the four methods: *RaneGain Test Set*, *OL Light*, *Oscilloscope*, or *AC Voltmeter* described below.

- **RaneGain Test Set** Plug the RG Transducer into the console's (or preamp's) master balanced output XLR jack. Turn up the master output fader (or preamp output level control) until the Transducer first sounds; reduce the level until the Transducer stops. This is now the *maximum system signal* output.
- **OL Light** Adjust the sound source until the master output overload (OL) indicator just begins to light (or the output meter indicates an OL condition). This is now the *maximum system signal* output, although it is a conservative maximum since most OL indicators come on several dB before actual clipping.
- **Oscilloscope** Using the RG Transducer or OL light are fast and convenient ways to set levels. However, a better alternative is to use an oscilloscope and actually *measure* the output to see where excessive clipping really begins. This method gets around the many different ways that OL points are detected and displayed by manufacturers. *There is no standard for OL detection.* If you want the absolute largest signal possible before real clipping, you must use either the RG Transducer or an oscilloscope.
- **AC Voltmeter** If the RG Transducer or an oscilloscope is out of the question, another alternative is to use an AC voltmeter (preferably with a "dB" scale). Here, instead of relying on the OL indicator, you choose a very large output level, say, +20 dBu (7.75 Vrms) and *define that as your maximum level.* Now set everything to not clip at this level. This is a reasonable and accurate way to do it, but is it an appropriate maximum? Well, you already know (from the above discussion) that you need 12- 20 dB of headroom above your average signal. It is normal pro audio practice to set your average level at +4 dBu (which, incidentally, registers as "0 dB" on a true VU meter). And since all high quality pro audio equipment can handle +20 dBu in and out, then this value becomes a safe maximum level for setting gains, giving you 16 dB of headroom — plenty for most systems.

Outboard gear gain/level controls fall into three categories:

- No controls
- One control, either Input or Output
- Both Input & Output Controls

Obviously, the first category is not a problem!

If there is only one level control, regardless of its location, set it to give the maximum output level either by observing the OL light, or the RG Transducer, or the oscilloscope, or by setting an output level of +20 dBu on your AC voltmeter.

With two controls it is very important to set the *Input control* first. Do this by turning up the Output control just enough to observe the signal. Set the Input control to barely light the OL indicator, then back it down a hair, or set it just below clipping using your oscilloscope, or until the RG Transducer buzzes. Now set the Output control also to just light the OL indicator, or just at clipping using the scope, or just buzzing. (Note: there is no good way to optimally set an input control on a unit with two level controls, using only an AC voltmeter.)

For Rane digital audio products, like the Rane RPM series of Multiprocessors, where input A/D (analog-to-digital) metering is provided with software, setting the input level gain is particularly easy and *extremely important*: Using the maximum system signal as the input, open up the Input Trim box and simply slide the control until the 0 dBFS indicator begins lighting. This indicates the onset of "digital clipping," and is definitely something you want to avoid, so this is the maximum gain point.

## SETTING POWER AMPLIFIERS

If your system uses active crossovers, for the moment, set all the crossover output level controls to maximum.

Much confusion surrounds power amplifier controls.

First, let's establish that power amplifier "level/volume/gain" controls are *input sensitivity* controls. (no matter *how* they are calibrated.) They are not *power* controls. They have absolutely *nothing to do with output power*. They are sensitivity controls, i.e., these controls determine exactly what input level will cause the amplifier to produce full power. Or, if you prefer, they determine just how *sensitive* the amplifier is. For example, they might be set such that an input level of +4 dBu causes full power, or such that an input level of +20 dBu causes full power, or whatever-input-level-your-system-may-require, causes full power.

They do not *change* the available output power. They only change the required input level to produce full output power.

Clearly understanding the above, makes setting these controls elementary. You want the *maximum system signal* to cause full power; therefore set the amplifier controls to give full power with your maximum input signal using the following procedure:

1. Turn the sensitivity controls all the way down (least sensitive; fully CCW; off).
2. Make sure the device driving the amp is delivering max (unclipped) signal.
3. Warn everyone you are about to make a LOT of noise!
4. Cover your ears and turn on the first power amplifier.
5. Slowly rotate the control until clipping just begins. Stop! This is the maximum possible power output using the maximum system input signal. In general, if there is never a bigger input signal, this setting guarantees the amplifier cannot clip. (Note: if this much power causes the loudspeaker to "bottom out," or distort in any manner, then you have a mismatch between your amp and speaker. *Matching speakers and amps is another subject beyond this note.*)
6. Repeat the above process for each power amplifier.
7. Turn the test signal off.

## ACTIVE CROSSOVER OUTPUT LEVEL CONTROLS

Setting the output attenuators on active crossovers differs from other outboard gear in that they serve a different purpose. These attenuators allow setting different output levels to each driver to correct for efficiency differences. This means that the same voltage applied to different drivers results in different loudness levels. This is the loudspeaker *sensitivity* specification, usually stated as so many dB SPL at a distance of one meter, when driven with one watt. Ergo, you want to set these controls for equal maximum loudness in each driver section. Try this approach:

1. Turn down all the crossover outputs *except for the lowest frequency band*, typically labeled "Low-Out." (Set one channel at a time for stereo systems.)
2. If available, use pink noise as a source for these settings; otherwise use a frequency tone that falls mid-band for each section. Turn up the source until you verify the console is putting out the maximum system signal level (somewhere around the console clipping point.) Using an SPL meter (*Important: turn off all weighting filters; the SPL meter must have a flat response mode*) turn down this one output level control until the maximum desired loudness level is reached, typically around 100-105 dB SPL. Very loud, but not harmful. (1-2 hours is the Permissible Noise Exposure allowed by the U.S. Dept. of Labor Noise Regulations for 100-105 dB SPL, A-weighted levels.)

Okay. You have established that with this maximum system signal this driver will not exceed your desired maximum loudness level (at the location picked for measurement). Now, do the same for the other output sections as follows:

1. *Mute* this output section — *do not turn down the level control; you just set it!* If a Mute button is not provided on the crossover, disconnect the cable to the power amp.
2. Turn up the next output section: either "High-Out" for 2-way systems, or "Mid-Out" for 3-way systems, until the *same maximum loudness level* is reached. Stop and mute this output.
3. Continue this procedure until all output level controls are set.
4. Un-mute all sections, and turn off the test source.

*Congratulations!* You have finished correctly setting the gain structure for your system. Now you are ready to adjust EQ and set all dynamic controllers. Remember, after EQ-ing to *always reset the EQ level controls for unity gain*. Use the Bypass (or Engage) pushbuttons to "A/B" between equalized and un-equalized sound, adjusting the overall level controls as required for equal loudness in both positions.

## SUMMARY

Optimum performance requires correctly setting the gain structure of sound systems. It makes the difference between excellent sounding systems and mediocre ones. The proper method begins by taking all necessary gain in the console, or preamp. All outboard units operate with unity gain, and are set to pass the *maximum system signal* without clipping. The power amplifier sensitivity controls are set for a level appropriate to pass the maximum system signal without excessive clipping. Lastly, active crossover output controls are set to correct for loudspeaker efficiency differences.

## REFERENCES


1. Murray, John & Pat Brown, "A Gain Structure Guide," *LIVE SOUND! International*, pp. 18-24, Mar/Apr 1997. Thanks to John and Pat for inspiration and some content for this RaneNote.
2. *The Syn-Aud-Con Newsletter*. Various issues; you need them all — subscribe: 1-800-796-2831.