HEY SOUNDMAN!

LIVE AUDIO ENGINEERING BASICS

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Chapter 1 - WHAT IS SOUND?

"Everybody form a line!"
- Bruce Springsteen's 'E-Street Shuffle'

Sound is *linear*.

Sound travels from one point to another. Like water and light, sound flows from the source to the receiving vessel. Along the way, it can be guided and shaped in a way that enhances its purity and clarity.

The best sound engineers are people who naturally organize their world into simple, linear patterns. Life makes the most sense when you handle tasks and challenges one step at a time in an appropriate order. Attention to the specific order of a series of connections is crucial to all aspects of quality sound engineering.

Imagine if a mail delivery person just had a big sack full of random envelopes that were not sorted by address and was just driving back and forth all over town. Or if a builder tried to construct the roof before the foundation was poured. The job would not get done very well.

Sound engineers maintain the line of communication from sender to receiver. It is important to be able to perform tasks and solve problems by visualizing successful results as the completion of a logical series of steps.

Sound is possibly the most sensitive of the five senses, and definitely the most readily manipulated. The seemingly natural sound we hear every day while passively enjoying recorded music, film and other multi-media is often the result of a great deal of human interaction. We mostly prefer to just enjoy the show and pay no attention to the man or woman behind the console. The sound engineer works like an invisible conductor in the space between the source and the vessel.

Sound engineering is the art of making connections. We hook stuff up.

They say the shortest distance between two points is a straight line. Sound is on that line, and sound engineers are the ones who straighten it out.

So let's get it straight…

"I'm picking up good vibrations"
- The Beach Boys

It's not a NASA shuttle launch, but sound engineering does involve a helpful awareness of a few basic properties of entry-level physics.

The scientific principles we will need to discuss are pretty much summarized by the paper cup telephones we played with as kids.

Sound is molecules connecting through solid matter, and sound is waves passing through the air. When we speak into a dixie cup on one end and it comes out of the cup at the other end, it seems like magic. But there are simple physical principles at work which explain how it happens. These connections are all linear in nature.

Sound is created as a vibration.

It starts with the manipulation of an instrument, like striking a drum, picking the strings on a guitar, or the lungs moving air through our vocal chords. This is point A at the start of the line. The ears of the audience are waiting at point B. We are both the senders and receivers.

The initial vibrations being created move at certain speeds and patterns. These **sound waves** travel through air and are received by our ears and our bodies, which translate the wave patterns to our brain as information. The brain organizes and interprets the information into different meanings.

EXERCISE 1

Close your eyes now and just listen for a moment. You are probably hearing a combination of sounds in the background of whatever situation you are in right now. Try to separate the individual sounds you are hearing from the most prominent to the least. Now try to listen for something in the background you were not aware of at first. Consider which sounds are more noticeable and which sounds are more subtle. What makes a sound seem loud? Or soft? What makes these sounds jump out or stay in the background? Pick a particular sound and attempt to relate it in whatever descriptive terms come to mind.

Different combinations of sounds convey different messages and meanings to the brain. Sometimes our brain's interpretation of sound wave information is everyday noise, like busy traffic or dramatic weather. Sometimes it is interpreted as meaningful content like a train whistle or a cat's meow. All too often, some would say, the most noticeable sound in our ears is human speech. In the best case scenario, the information reaching your ears is the elusive and divinely-inspired combination of sound waves that we humans define as "music."

It is important for sound engineers to have a musical ear. Among many reasons is that sound engineers need to be able to recognize the difference between all the various vibrations that combine to make music.

There are some very strong connections between the science of sound and the art of music. Foremost of these areas is the concept of "pitch," which is a recognition of the differences in vibration of low to high tones.

Low tones are slow vibrations, like rumbling bass notes and deep booming thumps that resound at the foundation of a structure. Higher pitched sounds like crashing cymbals, claps and snaps are the result of faster vibrations at the brighter end of the range.

Low tones are more robust and tend to travel farther. Higher tones are more fragile and dissipate faster. The sounds of musical instruments, like drums, guitars, and even the human voice are widely-mixed expressions of low-range, mid-range, and high pitched "**frequencies**."

A musical ear naturally recognizes the variations and interactions of these diverse and complex frequencies. A technician with a trained ear can describe varying frequencies in certain measurement terms. The most common frequency unit is Hertz (Hz), which is one cycle per second. A 100 Hz sound wave travels at a rate of 100 cycles per second.

A person with musical training could also recognize these same tones as musical notes, such as "middle C." For example, a string vibrating at 440Hz produces the tone that western music notation calls "A."

The lowest pitches in human hearing are around 20 Hz, and the highest are around 20,000 Hz. The boom of a bass drum is between 75 and 100 Hz. The cymbals crash between 3,000 and 6,000 Hz. We commonly abbreviate these higher multiples as 3k and 6k.

The recognition of these frequencies is a huge part of sound engineering.

EXERCISE 2

Try this exercise as a helpful gauge to keep in mind while learning to recognize audio frequencies. Just wet your lips and whistle. Drop the tone of the whistle as low as you can without losing the tone. The lowest pitch that most adult mouths can whistle is around 1,000 hz, or 1k. Your highest whistle note, *unless you are a bird, is in the 2 or 3k range.*

Now sing a high note with your full voice (like the note at the end of the phrase "Oh say can you seeeee!") Good singers can generally hit their highest full voice notes within the scale that starts at middle C, which falls roughly in the 250 to 500 hz range. If you can hit the high C at 523 hz, that's some impressive pipes.

You can also make higher-pitched sounds with your mouth, such as the "S" sound in words like "sound systems." This is good to know, because the "S" sound often causes a subtle unwanted hiss in the 6k to 8k range with some singers. Easy to remember, since "six" and "seven" start with "ssss…"

As you achieve more experience, you will begin to recognize frequencies in smaller sections. Most of us can already discern the difference between general lows and highs. Even if you are not sure of the specific frequency, recognizing a sound as being in a general range is helpful. Finer areas of the frequency band, such as those described with compound terms such as "Low/mids" and "Mid/highs," correspond with smaller areas of the overall sound scape, to use another commonly mixed metaphor. Low sounds rumble like thunder starting at 20 Hz. Your voice is in the middle between 100 and 500 Hz, the high mid range starts with a whistle around 1000 Hz, and the highest sounds hiss and rustle above 4k.

Eventually, the smallest bands of the broad frequency spectrum in between low and high will become more readily identifiable, and some day you may be able to recognize each specific tone in the frequency band with your ears alone. You will also be able to visualize the frequencies on a linear scale in your mind's eye, as well as on the screens where the graphic representations of these bands of sound are displayed on modern digital mixers and workstations.

PROS ONLY

Practice making sounds with your own vocals that run a wide spectrum of frequencies. Being able to generate various tones with your own voice into microphones is a crucial skill for tuning rooms and ringing out monitors. The monotone of the cliche "Check-one-two" serves this purpose only minimally. If you can sing notes up and down the musical scale, whistle and hiss up to the high end and drop down to Johnny Cash bass notes while checking microphones, you are way ahead of the curve.

Sound engineers need to have a musical ear and be smart, resourceful creatures. Modern devices have given us the ability to use convenient digital apps to help us recognize these variations in pitch and dynamics.

Experimenting with these tools and toys is a great way to develop a more highly trained musical ear. More now than ever, you certainly don't need perfect pitch to be a good sound engineer any more than you need perfect pitch to be a musician, but it helps.

EXERCISE 3

You can download an "RTA" frequency-analyzing app on your phone (short for Real Time Analyzer). Try whistling into it, and verify with the graphic waveform how close to 1k your lowest whistle note is. This can be a helpful quick reference because this mid-range frequency is often an area of concern in vocal monitor mixes. 1k is a very good frequency to learn to recognize, with or without an RTA device.

GLOSSARY ONE

Many words used commonly in sound engineering jargon have multiple meanings, and many concepts are referred to by multiple terms. We will add to the glossary as we go:

Sound waves
Pitch
frequency
hertz
tone
low range
mid range
highs
instrument

CHAPTER 2 - IT'S ALL RELATIVE

"Turn it Up!"
- Lynyrd Skynyrd

Sound engineering and music share another important measurement scale, which is **dynamics**. Just as musical notes are low or high in pitch, volume is the scale of the sound from quiet to loud. Sound volume level is measured in **decibels** or dB.

The range of volume perceptible to humans starts with a slight whisper just above 0dB. Conversational speech is around 60dB and rock concert sound at the front of house position is around 100dB. Noise gets painfully loud above 120dB, like thunder cracking or a jackhammer blasting.

Volume is defined and measured through a variety of approaches with some generally-agreed standards. One such rule of thumb is that every 10 dB increase roughly represents a doubling of volume.

Quiet and loud are perceived by people in subjective ways. For example, the dense sound of a distorted guitar strummed constantly can make a band seem much louder to the ear than a snare drum played on occasional beats. That drum may be louder in decibels the moment it is struck, but the density of the other instruments in the music is what makes the perception of volume higher.

You can also think of volume in the more literal sense, as the physical property of how much space is being taken up in a vessel, like "the weight by volume" of a package. Sound waves take up space in the overall bandwidth in the same way.

Another strange but true example of "perceived" volume is that people will often think a band is louder just because their flashy stage outfits and big speakers make the music "look" louder to them. It is a worthwhile goal for a sound engineer to never have anyone complain about the volume, but once in a while it happens before the band even plays a note.

And every sound engineer knows there is nothing louder than a leaf blower at 8am the morning after a late night load out.

The main mixing tool which the sound engineer possesses is the ability to manipulate the relative dynamic volume levels of the various elements in the music. The second tool is the ability to emphasize or limit the specific frequencies within those discreet audio signals. We are blending lows, mids and high frequencies at different relative volumes like a master chef mixes spices and ingredients in a perfect dish.

I guess it must be lunch time...

PROS ONLY:

It helps to raise the noise floor before a band starts in a room *full of unsuspecting people by increasing the recorded house music gradually. Ear drums expand and contract to compensate for volume the way our pupils react to light, so you can help your audience get used to live music by easing their ears into it instead of unleashing it on them suddenly.*

"Take Me to the River…"
- Rev. Al Green

The changes that happen to the sound in between the instrument and the ear are what the engineer is controlling with the help of several systems. The primary overall system at our command is the PA system, which consists of the circuits, components and hardware used at the various stages of the engineering process.

It all starts with a sound created by the performer, which is then turned into a "**signal**." A signal is sound that has been converted from a wave into the form of an electrical impulse traveling through wires and circuits. Microphones and other so-called "**analog**" devices transform sound waves into these electrical signals. The signals then travel through the system, being processed in various ways at numerous stages, until the signal is turned back into sound waves by the speakers which cause the sound to travel through the air to the audience's eager eardrums.

This process of **"transducing"** sound waves into electrical energy is based on a consistent, measurable physical reaction of magnets and coils. We don't need to comprehend a deeply technical understanding of the science involved, unless we are going into electrical engineering like an electronics repair person, which is not the goal of this beginner-level class. However, a fundamental comprehension of these properties can be very helpful to tracking the linear flow of sound.

What we need to know from the start is that sound is transferred from waves in the air into electrical energy that flows through wires, by a process involving the magnetic properties of the elements in the hardware. This transfer occurs at both ends of the system, and often throughout the signal path also, depending on the sophistication of the audio hardware. The microphones and the speakers both contain magnets and coils that translate mechanical energy into electrical signal and vice versa.

It's a simple binary process, meaning that the molecules are either positively or negatively charged, and they behave physically in response to the interaction of other molecules charged the same or opposite. Like a river, the signal stream "flows" from sender to receiver on a magnetic "**current**" of energy.

GLOSSARY TWO

dynamics
decibels
analog
signal
decibels
current

CHAPTER 3 - IS THIS THING ON?

SOUND SYSTEM =
SOURCE + MIXER + AMPLIFIER + SPEAKERS

Necessity is the mother of invention. Most of us started to learn how to do sound out of desperation. We found ourselves in a band playing in a garage or coffee shop with some archaic borrowed equipment, and we just started plugging things in and turning knobs through trial and error to get it to make sound. Eventually the way the stuff works started to make more sense to our curious minds, and we began to grasp the concept and try to improve upon it. Before you know it, like it or not, you're a sound engineer.

The most basic sound system consists of a "**source**," such as a microphone or any device creating a signal, plus the cables that carry the signal, plus the mixer and processors that adjust and direct the signal, plus the amplifier that adds power or "**amplitude**" to the signal, and the speakers that turn the amplified signal back into sound waves.

PA stands for "Public Address" system. A simple PA system, such as one used in a practice space or a small cafe, uses a few mics, a mixer and a pair of full range active or passive speakers. "Active" speakers combine the amp and speaker in one self-contained unit. "Passive" speakers require a separate power amp.

On the most basic PA system, there are 4 to 8 input paths, or "**channels**" of signal being sent to one mono mix for the main speakers and one mix for the stage monitors, for a total of two mixes, connected to the main speakers and monitors through two amplifiers, often contained in one powered unit or "head."

"Monitors," or "wedges," are the speakers on stage aimed at the performers for them to hear their own preferred mix of vocals and instruments. This is a different, discreet balance of inputs than the mix sent to the main speakers for the audience.

A slightly more advanced system might have a set of stereo mains and one or two separate monitor mixes. In such a set up there are three or four amplifiers. A stereo mix for the mains through two amplifier channels, and a monitor mix on one or two amplifier channels. Most commercially-available stand-alone amplifiers have two channels of equal power, referred to as a "stereo" amplifier.

A bigger system might add more monitor mixes, allowing each member of the band to have a custom stage mix. A system with stereo mains and 4 discreet monitor mixes uses 6 separate power amp channels: One stereo amp for mains and two stereo amps for monitors. In a larger venue, there might also be extra speakers for the audience nearest the stage called "fills" and speakers in the rear of the hall and other areas called "delays."

CROSSOVER POINTS

Main speakers can also be divided into lows, mids and highs using a **crossover** unit, a processor that divides and balances a signal into separate low, mid and high frequency signals. These are sent to separate amps and speaker components: one amp for low speakers right and left, one for mid speakers right and left, and one for high end "**horn drivers"** right and left. The horn is the plastic conical baffle that spreads and projects the sound waves out from the tiny high end driver.

Many PA systems also have a separate amp and speaker cabinets or powered speakers for the low end subwoofers, or "**Subs**." Subs require the most amplifier wattage power and high-end drivers require the least.

A PA system with an active external crossover system works more efficiently because the internal crossover circuits in speakers "cut off" some of the powered signal in the process of dividing it, therefore wasting a portion of the amplifier's effort.

Crossovers allow the manual adjustment of the frequency points where the lows, mids and highs are separated. Moving these settings up or down can often help smooth out the overall response of the speakers. Subs are usually cut off around 100hz, and the lower you move this point, the tighter and less rattling a sub cabinet will sound. The mid/high cut off point is usually between 1k and 3k, and depending on the size and quality of the horn driver, adjusting this point will often make a difference in the clarity of the vocals.

A typical PA system in a professional venue is actually made up of 12 or more discreet sound systems, each consisting of a source, amp, and a speaker:

Left Low/Right Low
Left Mid/Right Mid

Left High/Right High
Mon 1/Mon 2
Mon 3/Mon 4
Mon 5/Mon 6
front fill L / front fill R
delay L / delay R
patio / kitchen

PROS ONLY

Subs can also be connected to a special feed direct from the mixer instead of the main crossover, which gives the engineer access to all of the mixer's detailed controls for adjusting the low signal. This will require the mixer's controls to remove the high frequencies and/or a self-contained separate crossover in the signal to the sub amplifier.

MIX IT UP

As sound engineers we are all born with the main tools required for the job: our ears, our eyes, and our hands. Let's look now at the primary piece of external equipment we will need to have handy at our fingertips.

The **mixing board** (or console, or desk, or mixer…) is like the control panel surface of the sound system. If the PA system was a race car, the rack full of power amplifiers is like the engine, the speakers are the wheels pushing the music forward, and the point where the engineer interacts with the entire sound system is like the driver in the cockpit. The mixer is the interface where we use the controls to steer the system, give it more gas or put on the brakes.

Sounds are generated by the musical instruments on stage, where they are collected by microphones, turned into signals which are transferred along a system of cables, to arrive at the mixing board in a big, unsorted mass where we as engineers are in charge of sorting them out and presenting them in a more defined, coherent form.

Engineers work with each signal one at a time, straightening them out and sending them where they are supposed to go. The mixing board is the connecting point for all those signals.

You can visualize all these signals interacting with the mixer as a grid of "**inputs**" and "**outputs**." A number of signals come in to the grid, where the signals then get directed to a number of outputs leaving the grid. It's like a busy airport, where the planes come in from all directions and get sent out in many other directions.

A 24 channel board has up to 24 inputs, or "**channels**" coming in at a time, and these signals can each be sent out to up to 8 or more outputs, or "**sends.**" A single engineer in an average size venue (around 250 capacity) or smaller is often in charge of mixing the front of house main mix as well as 4 or more stage monitor mixes, plus various external mixes.

MIX INPUTS

A working sound engineer has a preferred way of connecting or "patching" the various inputs to the mixer in a consistent pattern, show after show. This is called a standard "**patch**." It can be abbreviated in sharpie marker on a strip of masking tape and remain attached to the board for reference on an analog mixer. Digital boards have color coded tags that display the input description on each channel, and you can save the default channel settings for each instrument on "scenes" stored internally.

An effective standard patch on a house mixer will be able to accommodate the inputs of 99% of bands that come through. Some channels should be left as undesignated spares to allow flexibility. Some bands will have larger drum sets, more guitar amps or more DIs. Larger venues with bigger consoles will add more drum mics, such as overheads, snare bottom, inside kick drum head, and more DI lines.

Larger venues also have a monitor desk where a second engineer on the side of the stage takes a split set of all the stage signals and mixes only the stage monitors for the performers, leaving the "**Front of House"** engineer the freedom to mix only the mains for the audience.

An example of a standard 24 channel mixer patch for a typical club venue is something as follows:

1 - K = kik drum
2 - S = snare
3 - H = hi hat
4 - R = rack tom
5 - F = floor tom
6 - spare
7 - G< = guitar L
8 - G> = guitar R
9 - B = bass
10 - DI<
11 - DI<
12 - DI
13 - DI>
14 - DI>
15 - spare
16 - spare
17 - V< = L vocal
18 - V = Ctr vocal
19 - V> R vocal
20 - V>> RR vocal
21 - V^ = drum vocal

22 - spare
23 - spare
24 - TB= talkback mic

THEATRICAL STAGE DIRECTIONS

You need to know the basic traditional stage directions for communication with musicians and other production staff. "Stage right" is to the right of a person standing on stage facing the audience. "House right" is the right side of the stage from the audience. "Down stage" is the edge of the stage nearest the audience and "Up stage" is in the rear. "Back stage" is where the "green room" is.

You will also want to arrange your input list in a left to right fashion, with the vocal mic and guitar amp on stage right (your left) on the input list before the vocal mic and guitar amp on stage left.

The same applies to the order of the monitor mixes. Mix one will be the stage right wedge, mix four will be at stage left, mix 5 the drum wedge.

Now let me see your backstage pass…

PROS ONLY

Labeling your board inputs is crucial to mixing, and labeling the various XLR cables on stage is also super helpful for patching, especially in a festival situation where the inputs will change several times throughout the event.

Wrap a short piece of masking tape around the mic end of the XLR cable and write on it the number of the input it is connected to. Then make sure the cables are never disconnected at the snake throughout the show. No matter what you plug that cable into, a vocal mic, a direct box, or a drum mic, it will always be on the channel indicated on the label. (Keep an eye out for *bands switching the order of the vocal mics while they set up. You may have to go and unplug your XLRs and put them back in the proper order on the vocal mics so it doesn't cause confusion.)*

There is nothing more frustrating than picking up the end of an unlabeled XLR cable on stage in the middle of a show and trying to trace it back to the snake through all the gear sitting in the way (and usually on top of the cable) to determine what channel it is plugged into.

EXTRA TECHS

In rushed and crowded festival situations, it is advantageous to have more than one sound crew member. This extra hand can oversee the on-stage patching between acts while allowing the front of house engineer to mostly remain at the console instead of trying to get back and forth through the crowd.

On touring crews in bigger venues, a third tech might also be employed to run a separate monitor console. Professional concert venues are often also contracted to supply extra loaders and stage hands for moving band equipment on and off stage, and in and out of the venue.

GLOSSARY THREE

source
gain
channel
amplifier
amplitude
active
passive
Patch
input
output
send
sound system

signal chain or path
pre amp
line level
flat
unity
downstage

CHAPTER 4 - CHAIRMAN OF THE BOARD

In keeping with our linear theme, the mixing board consists a grid of **channel strips**, identical columns of buttons and knobs which correspond to each input. At the top of each channel strip you will find the gain knob controlling the signal input, and at the bottom is the fader that sends the signal to the master mix. In between are the sections for EQ and auxiliary sends, and other various controls depending on the sophistication level of the board.

When used properly, the surface of the mixer itself is a graphic display of lines and dials that visually represent a good deal of information about what is happening with the mix. For example, you can easily see which knobs are turned all the way down to the "infinity" mark, which means all signal is cut, and you can see which knobs are at the "zero" mark, which means no gain is being added or reduced. There is often a notch point or a "detante" in a pot where the knob has a little hitch when it is at the mid point, making it easy to quickly reset to zero.

The row of faders at the bottom shows the relative position in the mix of each instrument. You can see in an obvious graphic way which vocal is set higher in relation to the others, or which guitar has been boosted for a lead solo.

PREAMP TRIM

The first knob, or "pot" at the top of each channel strip is called the "**input gain**", or "trim." This is the first and most important control you have over those unruly, unstable and wildly variable signal levels hitting the input of the board. When the signal is attenuated to a user-friendly level, it is being processed as a nice, smooth wave. Adjust this pot until you have a stable, steady green level signal showing on the meter. Ideally, there is a mark on the board indicating "zero" or "**unity gain.**" Your first order of business is to use the gain control get the signal to be at an average unity level.

THE CHANNEL STRIP:

input gain
pad
phantom power
high pass
EQ high
EQ mid
EQ sweep
EQ low
aux sends
FX sends
pre/post
pan

mute
PFL
assigns
fader

The signal might also be coming in too weak from a microphone to be useful. The same gain pot also controls a "**pre-amp**" a tiny amplifier at the input of the channel which can add a very small amount of voltage, or "**gain**" to boost the signal to a healthy level. This is helpful with vocals, and with acoustic instruments which often contain inconsistent electronics. These instruments can be the trickiest to manage. If you have to adjust the pre gain up or down more than a few dB, it is wise to see if the signal can be adjusted on stage.

Boosting pre gain too much makes the signal vulnerable to painful peaks. For example, you might add a bunch of gain to a weak signal from an acoustic guitar, and then when the musician suddenly wiggles a bad cable, the signal comes in super hot and peaks the system and everyone's ears. For this reason, it is useful to make your initial gain adjustments with the channel muted until you are confident the signal is stable.

Microphones are generally designed to produce a level of voltage that is roughly at unity gain when a source such as the typical sung vocal is introduced. This means that if we start the show with the vocal channel at unity gain, we can feel somewhat secure that it will be fairly audible. In order to bring a softer vocal up to a level in the PA that carries it above a band, we need to adjust the input gain slightly above unity gain on the knob to get it to register at unity on the meter, so it can be mixed at the proper ratio to the other instruments.

"Line level" and "Mic level" are terms used to distinguish between the 2 most common standard signal strengths. Line level is a signal averaging about one volt in power that is output from a variety of sources such as stereo equipment, some DJ mixers and other multimedia components, and usually via 1/4 inch or RCA style 2 conductor connectors. It usually has already gone through a pre-amplifier phase.

Mic level is an even lower signal strength produced by microphones and carried by 3 conductor XLR cables. Mic level signal requires a preamplifier to produce a line level signal.

"Instrument level" is in between mic and line level, and usually comes from an electric instrument like an electric guitar or keyboard. Mixers often have a button near the gain pot at the top of the channel called a "**pad**" which reduces the input signal level by a certain amount. This is a useful way to significantly cut the level of a signal that is coming in too strong, so the trim knob can be reserved for finer gain adjustment.

The pre fade level meter or "**PFL**" is a meter that is usually accessed by pushing a button near the channel mute button. When engaged, it shows on the main meter the level of signal at the channel input. This is a helpful way to see the input level without relying on hearing the mains. It operates the same when the channel is muted. This gives you the opportunity to detect a signal that is excessively hot or weak before you un-mute. This is a very useful safety measure for adjusting gain on direct instruments with unpredictable signal strength before making them audible.

As you can imagine, adjusting the pre-amp gain control on each channel is a super sensitive and crucial primary aspect of the overall mixing process.

PHANTOM POWER

Dynamic microphones like the Shure SM58 use diaphragms and coils to create signal. Condenser mics contain capacitors (formerly called condensers) and require phantom power, which comes from the mixer through the XLR cable when the 48v "**phantom power**" button is engaged on the channel.

The capacitor in a condenser mic stores this small voltage charge and uses it to convert sound waves to signal. The conductor wires in the XLR cable are carrying both phantom power from the mixer to the mic and stage signal from the mic to the mixer at the same time. Condenser mics are very sensitive to more subtle sounds, and more vulnerable to instability than dynamic mics.

CHANNEL EQ

In the next section of each channel strip, the channel "**EQ,"** you will see pots for high, mid, and low, and sweep knobs for **parametric** EQs, which allow the user to set the variable frequency being adjusted. There will be a "**high pass filter**" or "low-cut" button that cuts off frequencies below a certain point, which is useful for vocal microphones and other signals where the inclusion of the lower frequency sound waves is undesired.

These lower sound waves in a signal are often the result of "**bleed**," or the leaking of the sound of other surrounding instruments into mics that are not intended to carry it. Filtering your channels helps to isolate the sound of the instrument that is intended to be represented on each channel. If there is kick drum coming through a guitar mic for example, it will make it more difficult to regulate the level and sound of both of those instruments in the mix. Filtering is a very effective technique for bringing clarity to a murky mix.

AUX SENDS

"Aux send" pots are for increasing the signal of a particular input channel to various auxiliary mixes such as stage monitors. With these controls, a front of house engineer can establish a number of custom monitor mixes discreet from the front of house mix.

"Pre fade" and **"post fade"** buttons dictate whether a signal to the aux mixes is affected by the channel fader level. On analog mixers, the first 2 or 4 aux mixes are usually preset as "pre fader" and the last 2 are switchable. There are also FX mixes which are always post fader. Pre and post fade can also affect whether or not the channel EQ section is engaged on an aux mix, depending on the mixer's specifications. It's good to be aware of exactly what point in the signal path that your mixer's pre/post intersection occurs.

Pre fader aux mixes are helpful for monitor mixes that you do not want to be changing as the channel faders move up and down in the main mix. Post fader aux mixes are helpful for mixes such as outputs to recording devices and other zones where it is desirable for the aux mix to change as the main mix changes.

SUB GROUPS

Most professional mixers have "**sub groups"** or "**busses**" that can divide several channels of the mix into groups which in turn are combined to make up the **master** mix. These sub groups are created by pushing buttons near the channel fader which "**assign**" the channel to the sub groups and or the master.

The assign buttons on each channel select which master busses or groups the channel is included in, such as **sub groups**, the master main mix, and the mono send. Keep in mind, if no assign buttons are selected, the channel is effectively muted in the master mix. This can be easily overlooked and cause confusion. It can also be a deliberate technique for sending a signal only to the aux sends without hearing it in the mains.

Some newer analog mixers have "VCA" groups, which assign a master group fader to a set of selected channels and adjust their gains together remotely, without actually combining their signals to one buss.

PRO TIP

You can use the assign buttons to make a group of channels *that only go to a buss that feeds the subwoofer. This really clears up a mix by removing these instruments from the main speakers where they can interfere with vocals and other instruments and gives you the ability to adjust the overall level of the sub with a single fader at front of house.*

"**Pan"** is the control on a channel that moves the signal in the main mix stereo field from right to left. You should be judicious about using the pan control for creating a stereo field in your main mix. Consider the fact that very few people in the audience are going to be hearing the main speakers in a true stereo position. If you are doing a mix that places a lot of instruments at various degrees in the stereo field, like a studio engineer might create a recording mix, you are potentially making the mix imbalanced for most of the audience. There are just a few instruments that benefit from stereo separation.

The toms in a drum kit sound great when panned in a way that moves in stereo, and this effect can be appreciated throughout the room. Stereo keyboards on two channels can be panned to separate the low keys from high notes.

The biggest common mistake is panning guitar amps the way they appear on stage. If anything, pan the guitar amp on the right over to the left and vice versa so the audience gets a better balance. Always, without exception, place kick drum and bass in the center.

Panning can also be useful for separating signals into different sub mixes or busses. A single assign button will often assign a signal to a stereo pair of groups at once, such as group 1/2, or group 3/4. Sub group masters have pan controls also. So you can send some channels to just the left or right side using the pan controls, then use the sub group pan control to put them back in the middle of the main mix.

For example, engage the group 1-2 buss assign button on a group of channels. Pan all these channels to the left, in effect putting them all in sub group 1. Then use the sub group 1 master to move the level of all these channels up or down, and use the sub group 1 pan control to put the whole group back in the center of the main mix as desired.

MUTE

The channel mute buttons have red lights indicating the signal is cut off to the mains and all aux sends. When the channel is muted, the PFL meters still function, so you can test your input levels visually before sending signal to the mixes. Some mixers have mute groups which allow you to assign a group of channels to be muted by one master mute button.

PROS ONLY

The mute button can be a good trouble-shooting tool. If there is a strange hum or feedback and you are not sure what channel it is originating from, you can quickly mute channels one by one until it stops, thereby identifying the source.

TALKBACK

The talkback mic is connected at front of house for the engineer to use for announcements and communicating with the band through the monitors. The talkback mic may have a special input that you can assign to mains and/or auxes, but it is more handy to use a spare channel on the console, such as 24. When talking to the band, be sure to not send your voice to the mains, unless you are addressing the crowd.

Introducing bands on the talkback in the mains is a nice professional touch at a show, especially if you use the classic rock concert intro:

"Ladies and gentlemen, please welcome,
from Austin, TX, Polymer Records recording artist,
Johnny X and the Ys!"

MOTORIZED FADERS

Fancy analog boards and modern digital boards have faders that move on their own as if touched by ghosts. These fader moves can be programmed to fade channels up and down automatically for studio mixes, or they can move into different level positions when different mix scenes are selected. For example, on a digital board, the faders might be displaying the main channel input levels in one scene, and then when another scene is selected, they move into position to represent the auxiliary output masters.

MASTERS

The master section of the mixer consists of the master sends for the sub groups, the master sends for the various aux mixes, the master sends to the effects, and most importantly, the master fader to the mains. If everything is going haywire, remember you can always kill the master fader.

PRE SHOW ROUTINE

It is usually the sound engineer's job to turn on the system before the venue opens. If you turn on the power amps before turning on the console, you will create a loud pop in the speakers when the power from the console spikes the outputs. This abrupt "**square wave**" is harmful to the system. Turn on the console first, then the power amps. When powering down, reverse the process by turning the power amps off before the console.

An experienced engineer shows up at least 30 minutes before the band loads in and sets up the stage. Placing mic stands, running XLR cables, and checking vocal mics is much easier without trying to share the stage space and weave around band members setting up instruments and claiming their territory.

Bands should be asked to provide their input list and stage plot in advance so you can have the stage set beforehand with the proper number of vocal and instrument mics in the general locations needed. This "**advance work**" saves time and effort for everyone involved.

When the advance information has not been provided, you should set up in a default mode. 99% of bands can be accommodated with the basic club stage patch set up. Most bands consist of drums, bass, 2 or 3 guitar amps and 3 or 4 vocals. After that, the next most common instruments are keyboard and acoustic DIs. Sometimes the drummer sings. If you are set up for this default configuration, you will be ready for almost anything. It's like a "zone defense" in sports. You have all the general bases covered.

Run your vocal mic cables from the snake along a path between the monitor wedges and vocal mic stands and coil the slack at the end by the mic stand. This way, when the mic stand moves or the singer picks up the mic, there will be extra cable to follow the mic around. The same applies to instrument mics. The mics should be placed in the general area where they will be used, waiting for the band to set up. Monitor wedge speaker cables should also have slack coiled near the speaker for the same reason.

When the musicians arrive, you get out of the way and start setting up your mix at the console. When they are all set up on stage and ready to play, you hop on stage for a minute to point mics at speaker cabinets, position wedges in front of vocal mics, and clip on the drum mics.

AMP MICS

You will need to determine where speakers are mounted in guitar amp cabinets. Sometimes you can see through the grill cloth, or feel the outline of the speaker mount. You can also look or feel in the back of an open cabinet. Your cabinet mic, usually a Shure SM57 or a Sennheiser e906, should be positioned close to the grill but not touching, aiming at a space in between the edge of the speaker cone and the center. If your mic is on the edge or at the middle of the cone, it may sound too harsh or too dull.

SOUNDCHECK

There are two ways that sound checks are conducted. At a professional venue, where advance tickets are sold, doors open before the show after the stage has been set up and checked. At smaller clubs, bands often load in, set up and check while the club is open, where sometimes the public is already in the space, drinking, chatting and even dining.

At the minimum, you always need to do what is called a "**line check.**" The band is set up, you make a cursory confirmation that each instrument and microphone is passing signal. You can actually check many of the channels with no audio passing into the mains. You can visually determine using the board meters if signal is reaching the mixer and the noise made by the band before the show or between acts is contained to a few minimal notes played on each instrument.

The sound check done at a professional venue before the doors open to the public is a very different process, with much more attention paid to each instrument, individual monitor mixes given extra time, and a few songs performed (or rehearsed) with mains on at full volume. The expectation of most traveling bands selling advance tickets and arriving earlier in the day is to be accommodated with a full sound check.

PROS ONLY

With all the channels muted, you can (very) cautiously boost the input gain on each channel enough to see signal from the ambient room noise and verify that the mics are hot, which serves as a quick visual line check. You can also attach a mic to the direct input lines to verify that those cables are good.

GLOSSARY FOUR

console
channel
input
unity
pregain
high pass filter
bleed
parametric
assign
pan
input gain
trim
preamp
gain structure
sub groups
square wave
advance work
line check

Chapter 5 - SHOWTIME!

ONE-TWO-THREE-FOUR!
Dee Dee Ramone

In a perfect world, every live show would involve a good, thorough sound check before the doors open. But now, let's assume the people are here, the band is tuned up, and it's time to jump right in to the mix. A band is just about to start the show, there are hot mics on the stage, and we have been thrown in front of a mixing board at the last minute to save the day.

This is where Leslie Nielsen pops into the cockpit and says very seriously: "Good luck, and we're all counting on you…"

Ready? (or not), the band is counting off the first song, so here we go!

Those red lights on the buttons by the faders mean "mute." If the vocal channels are muted, better unmute them fast. The master fader better be up at unity also, and if the vocals are grouped, the sub group faders need to be unmuted and faded up too.

When a band starts playing, job number one is getting the vocals clearly audible in the mix. If the band is loud and headroom is an issue, you will need to adjust the input gain and the channel fader to boost the vocal above the band. Until you can hear vocals clearly, you shouldn't be concerned with anything else coming through the mains.

If you can't hear the vocal, neither can the audience, and you are seconds away from someone demanding that you to turn up the singer, usually while you are working on that exact challenge. Everybody's a sound man.

If the input gain, the channel fader and the master fader at unity don't get the vocal prominent enough to hear, your PA system may be inadequate. Turn up the master fader as high as you need to.

Once your vocal is loud enough, contrary to many venue engineers' methods, it's not smoke break time. It's not a real mix until the drums and instruments are well represented. Although it may sound reasonably mixed in the console position or in front of the stage where the audience is getting a good dose of stage sound, the mains must provide a complete mix to the entire venue.

PROS ONLY

There are some tricks for getting extra gain out of vocals to get them over a loud band. First, a little extra input gain can help, as long as your monitor does not become unstable and create feedback. You have to be careful with adding input gain during a live mix. You may need to simultaneously drop the aux send to the monitor on the channel while you increase input gain. You can also push channel faders and or the master fader above the unity mark to achieve headroom. Another effective way to put some gain in a vocal signal is inserting a compressor and boosting the make-up gain. Assigning the vocals to a sub group and inserting a compressor on the group with added make up gain is also a good way to push vocals.

KICK IT!

After getting the vocals on top, the first thing that helps make a club mix sound more like a professional concert is kick drum. When you have these 2 elements prominent, it starts to sound like a real show. Before you turn on the kick channel, look at the meter on the channel strip. Your eyes are the first of your senses you are going to use in the mixing process. If that meter is showing a few green LEDs pulsing with the beat, you are good to go, but if it's a cautious yellow or angry red, do NOT un-mute that channel!

If the kick drum is hitting red on the input meter, the signal is too hot. It is distorting, also known as **overloading**. The sound wave is coming in too loud. The circuit is over capacity, and the sound wave is making spikes that are being physically chopped off by the circuit, which is called "**clipping**." A red light indicates a "**clip**."

If that clipped signal were made audible, the clip is perceived as a loud and unmusical pop that is highly unpleasant, and also very harmful to speakers and eardrums. Think about the difference between sliding off a hill and dropping off a cliff. That clipped wave hits the ear like a slap to the face.

Give some attention during the first song to the level and character of the kick, the most often overlooked but central element of a band's sound. It's the foundation of the music, like a heartbeat. Finding the sweet spot on the kick drum will bring a band's sound together like nothing else at your command.

Start at sound check by experimenting with the high pass on the kick channel and the low mid sweep to isolate the core frequency in the bass drum tone and shape it to fit the music. Once the band starts, find a level that is audible but doesn't distract. You should feel it more than hear it. If it's missing from the mix, there is a general feeling of something lacking.

These two elements in the music, vocal and kick drum, should be dealt with from the get go in a live mix. Other instruments like direct acoustics and keyboards often need attention next, since they may not be very loud coming off the stage.

Guitar amps are usually last in line for your attention. At some point in the first song, there will be a lead guitar lick or solo that will indicate if the guitar needs to be given some help. Guitar amps are often the loudest thing in the room. You may have to mix the whole band, including drums, in relation to the level of the guitar amps. The sound engineer's job sometimes involves letting musicians know if it will be helpful overall for them to turn their amp down.

You should have guitar amps in the mains even if they are loud on stage, because the mains project over the head of the crowd down front, while the guitar amps are blocked by the audience.

I once asked a guest engineer in an elevated mixing booth to turn up the guitar. I couldn't hear it loud enough in the middle of the crowd, but the amps were blasting right at him.

He yelled down at me, "Are you hearing the same mix I am?"

I said, "No, I'm not, actually."

GAIN STRUCTURE

Live show mixing can be a delicate balancing act, diminishing some elements to enhance others. The general term for achieving the right levels at the various stages of the signal chain to get the most out of a signal is your "**gain structure**." You will see a lot of variety in different engineers approach to this endeavor. Many adhere to the dogged pursuit of unity gain. Others insist on adjusting input and master levels to make the mix faders all sit in the sweet spot around zero, because boards are supposedly designed to sound best that way.

It's important to walk around the room occasionally and make sure the mix sounds just as good in other areas of the venue besides the mix position. Sometimes the mixing console is in a part of the room that is less representative of the true mix. You may have to compensate for these inconsistencies while you are at the console. For example, there may be a bass trap in the corner where the mixer lives, and when you walk out in the middle of the room the low end goes away, so you need to add more kick and lows than your ears want you to when you are at the board.

It's all relative, and therein lies the sound engineer's freedom to their own interpretation of balance and expression of musical taste.

Trust your ears...

PROCESSING

We've got the signal coming into the board at a nice useable level and coming through strong in the main mix. The next phase of the signal path from stage to listener is the "processing" phase. If we take a less is more approach and allow a signal to pass straight through "as-is," that is called a "flat" signal. All the frequencies in that signal are being reproduced and amplified to the listener just as the microphone picked them up.

But let's say that we want to change the quality of that signal before we present it to the audience. And why would we? Well, that can be a somewhat philosophical question.

The answer is simple. Our job is to make the music sound as good as possible.

For a variety of reasons, that flat signal might not sound as pleasing musically as would a processed signal that we can create by adjusting the frequencies that it contains. In other words, the purpose of the mixing board and the engineer are to make all the signals blend in a more pleasing way than what is coming off the stage unprocessed.

It's tempting to think that a flat signal is the most pure and accurate representation of the sound of the instrument on that channel. This is not actually the case or a variety of reasons. The hardware and circuitry of the system itself, the speakers and the architecture of the stage and the room can reinforce or subtract frequencies in the signal that color it, distort it, and otherwise obscure the pure sound of that instrument. No system provides a perfect reproduction.

It is up to us to use our ears and the processing tools in the system to make adjustments that restore it to as close of a representation as possible. It's like a famous sculptor who says he is removing the excess marble to find the statue that is already inside the raw uncarved block.

Besides gain, there are 3 other basic ways that signals are processed by the engineer: Equalization, Compression, and Effects.

PARAMETRIC EQ

Each channel strip on the most basic mixing board has at least a couple of pots controlling the levels of low and high frequency bands of each signal being input. This section of the channel strip is called the equalizer, or EQ. Simple channel EQs consist of separate gain knobs for low, mid and high, and some mixers have finer divisions of the frequencies, such as low-mid and high-mid.

An analog parametric equalizer has the same controls on the channel strip dividing the signal into at least 3 bands, and it also has knobs that "sweep" the frequency being controlled from lower to higher. You turn a pot to pick the frequency point you want to adjust and then you control the level of the selected frequency with another pot.

When lowering the gain of this frequency, this point is sometimes called a "notch." Increasing that frequency might be described as a "bump." The width of this notch or bump in the frequency spectrum is sometimes controlled by an extra knob or an outer-ringed pot called the "Q."

Parametric equalization really makes a lot more sense when you can see it represented on a graphic display. Newer digital touch screens allow you to actually move the frequency, gain and Q width around on a visible frequency spectrum in real time with your finger tips.

OUTBOARD EQs

Outboard rack-mounted equalizer units can also be routed into the signal chain after the outputs of the board and before the amplifiers. This is where the independent signals with custom mixes for the house main speakers and the individual stage monitor wedge speakers get equalization that is adjusted for those specific speaker applications.

A graphic equalizer is a physical interface as well as a graphic visual representation of the various frequency bands. The most common graphic equalizer is a 31 band, which represents the tactile controls for a single audio signal divided conveniently into a straight line of sliders that correspond with specific frequencies. Like a loaf of bread, the range of the sound is being sliced into physically adjustable sections from lowest pitch to highest.

EXERCISE

Talk and or sing into a microphone while listening to a fairly loud amplified signal of your vocal channel in the system speakers. Put all the EQ controls at the flat position and all the sliders on the graphic EQ on the mix at unity. Does it sound like a true and pure representation of your own voice to you? What is the system adding to or lacking in your voice? Use the channel EQ to make it sound more like your actual voice. Now adjust the graphic EQ on the send to make the speaker sound more true. Bypass the graphic EQ and compare the flat signal to the processed signal.

OCTAVES

These graphic EQs are also called "1/3rd octave" EQs. The width between a frequency and its doubled amount is known as an **"octave,"** and the 31 band graphic EQ divides each of the 10+ octaves that fall within the human hearing range of 20hz to 20k into 3 discreet "1/3 octave" bands per octave. (Octaves are so named for the 8 whole tones in a musical scale).

The bands of a 31 band graphic EQ are made up of 10 corresponding octaves of just 3 basic tones. In other words, every graphic frequency band is one of only three musical notes, none of which are exactly in tune. 100hz and its multiples are a little above the G note, 125hz is a little above B, and 160hz is just shy of E. All the bands in the graphic EQ are near to an octave of these 3 notes.

When mixing DI acoustic guitar in a monitor mix, it's helpful to consider how the frequencies of the open strings correspond with the bands of a graphic EQ. The challenge is that the A string at 110hz and the D string at 147 fall in between the standard EQ bands at 100, 125, and 160. The E string at 82 is closest to a band at 80. So while you can't specifically cut the overtone frequencies of open strings that resonate in the monitor, you can make general cuts of bandwidth clusters to eliminate ringing. You can also identify and regulate the “wolf note” of acoustic instruments, which are designed to have a natural sympathetic ring that falls between the frequencies of open strings, such as Ab on upright bass. For all these reasons, It is very handy when performers have on stage pre-amp units with sweepable EQs that can provide the specific notch of a problem note before sending to the mixer.

It is often helpful to also cut the neighboring octave multiples of a troublesome frequency, since pitches tend to resonate with their corresponding octave tones. So for example, a perceived surplus of 200 hz might also benefit from also cutting the 400 and/or 100 hz bands.

In the following grid, notice the columns of multiples in each octave row, and how some of these doubled octave multiples are rounded down to correspond with multiples of 10 of frequencies in other columns.

For example, the next octave above 160hz is rounded down to 315hz, which is 31.5x10. The octave of 630hz is rounded down to 1.25k, which is 125x10. These overlapping patterns will help you memorize all the frequency bands in a graphic EQ so you can start to match them to the frequencies you recognize by pitch.

31 band 1/3 Octave Frequencies:

20-25-31.5
40-50-63
80-100-125
160-200-250
315-400-500
630-800-1k

1.25k-1.6k-2k
2.5k-3.15k-4k
5k-6.3k-8k
10k-12.5k-16k
20k

COMPRESSION and GATING

The signal on a channel can also be diverted through an "insert point" on the back of the mixer to go through external cables to rack units or through internal digital circuits for other types of processing, most commonly compressors and gates. Compressors attenuate the gain level of inputs based on the user's preferences, like valves that limit the signal strength based on a ratio of gain reduction to signal. "Limiters" provide a hard gain level ceiling to a signal. Limiters can be very useful in protecting the system from overloading.

Gates cut off the signal until it reaches a certain threshold set by the engineer. This is useful for filtering external instruments out of a signal during the space between the specific parts being played. Drums are the most commonly gated signal. The gate can determine how long the tail of the drum is allowed to ring out before the gate shuts off the signal. This can result in a dramatic or subtle dynamic effect depending on the threshold setting.

Compression is the processing tool that can most help a band sound better. It can be used to make a bass guitar line sound more consistent or make a kick drum sound steadier.

The volume of a snare drum varies wildly from sharp loud cracks to soft brushing. Never gate the snare, because you might cut out some grace notes, but a good amount of compression helps keep the loud notes in check while bringing out the softer touches.

Used properly, compression smooths out the rough edges without being too obvious.

PROS ONLY

Experienced engineers often insert compression on sub groups, especially on analog boards with a limited amount of compression channels available. For example, a stereo compressor inserted on all the drums and another on all the vocals can help keep the various peaks of an active mix in check and also provide some extra headroom by utilizing the make up gain on the compressor, which adds back in overall signal gain that is being lowered intermittently by the compression process.

EXERCISE

Connect a portable source such as an iPod to a direct box using a headphone jack 1/8 inch to 1/4 inch stereo adapter and plug the input into a stereo channel on the mixer. Play some good dynamic music you are familiar with and work on achieving unity gain. Then send the signal to the mains and adjust the channel EQ controls until you like the overall balance.

Now set the sweepable mid pot to a relatively high peak, like 6db, and sweep the mid frequency point up and down, noticing what parts of the music correspond with the higher and lower pitches. You should hear something like a wah wah or flanger pedal effect.

Now insert a compressor on the channel and experiment with the various settings to recognize what the threshold, release, and gain are doing.

Now send the signal to each monitor to see if the signal has continuity throughout the system. Now hook up a microphone and do all the same techniques with the signal of your own voice.

Once you have run through these steps, you have effectively done what professional touring engineers do to test out an *unfamiliar PA system.*

EFFECTS SENDS

The post-fader aux send setting is mainly useful for sending signal from each channel to effects like reverb and delay. When the channel fader is lowered, the delay unit receives less receiving signal, so there is no excess of "wet" signal remaining in the mix inadvertently.

Effects sends are routed into effects processors for adding various textures like echo and reverb. Delays and reverbs have controls for length of the echo, the rate of the repeat, and the ratio of "wet" to "dry" signal. The rate on a "tap" delay can be set by pushing the tap button multiple times in rhythm with the music.

The effects return puts the effected signal back into the main mix at the desired proportion. If you have spare input channels, it can be useful to patch effects returns into spare channel inputs, where you have more control over the returning signal and can send to more outputs than just the mains. This also allows you to add effects to aux send channels, such as when an artist requests reverb in their monitor. Just be careful not to send effects back into themselves creating an internal feedback loop. And be sure to mute the effects returns between songs.

Aux sends sent to recording equipment can also be set to post fader if you want the mix to the recording device to be affected by adjustments to the front of house mix.

EXERCISE

Hook up a mic and send the signal to an effects unit and bring up the return on a channel or an effects return channel to the main mix. Make a pop or click sound with your mouth and experiment with the various parameter controls on the effects unit. Set up and save a reverb setting that sounds good for vocals. Set up a second reverb setting that sounds good for the drums, and set up a tap delay that you can access easily for the vocals.

These three effects will serve you well on a regular basis.

VOCAL PROCESSING

A voice is like a fingerprint or a snowflake. There are no two exactly alike. Every singer has unique frequencies in their voice and subtle characteristics that sound like no one else. The sound system is going to bring out those differences, and in some cases exaggerate them. The engineer can use gain, EQ and processing to enhance the pleasing tones and diminish any qualities of a singer's voice that the system is emphasizing too much.

There are usually a couple of narrow frequencies to be found in vocals that benefit from being cut to make a voice sound more purely human and less like a microphone through a speaker. Soft compression can also help smooth the peaks in a voice that tends to hit some notes harsher that others. And a bit of reverb can bring out the personality in an otherwise dry voice.

INSTRUMENT EQ

Finding the right level for an instrument in the overall mix is an ongoing process as the music changes from song to song. The EQ curve of the instrument channel also affects how the instrument fits in with the overall texture of the mix. Your musical ear really comes into play in the finer adjustment of each instrument's channel EQ. Like the vocals, you are trying to bring out the musical aspect of the signal and filter or cut out unwanted noise and overtones.

For example, acoustic bass can really benefit from cutting the deepest frequencies out and enhancing the low mids where the actual notes being played are most recognizable to the ear. Digital piano is often a little heavy in mid range and can use a little enhancing on the highs to bring out definition. Upper notes on lead guitar can be brought out by a little bump around 1 to 2k. Acoustic electric instruments like guitars and fiddles sound best as full range as possible with specific boomy lows cut out carefully.You will develop your own preferred techniques as your personal mixing style evolves.

A standard trick of the trade is to enhance some of the highest frequencies above 8k on vocals and other bright instruments such as acoustic guitar and hi hats for added "**presence.**"

Enhancing the highs can add articulation and help the signal cut through if used subtly, but it is easy to overdo and create a disconcerting artificial effect like tape hiss.

Don't let presence become unwelcome.

LESS THAN ZERO

After the show, leave the controls on the board the way you found them in a venue with multiple engineers. Sometimes they want the mixer left with everything turned down, which is called "**zeroing** the board." This is a mutually agreed upon condition of the house gear by the staff. Pre amp trims turned down and all channels muted is usually rule number one so you don't booby trap the next engineer who turns on the system. Outboard rack EQ units are usually left with cuts in place until an engineer chooses to restore them to flat and start over.

The stage itself has a zero state also. Some busy venues leave the stage partially patched overnight with XLRs coiled and ready. Mic stands can be put away or left in place by preference of the venue.

There is an old boy scout adage about leaving a campsite in the condition you found it. This is a a great rule for a sound engineer. If you are coming in to someone else's regular venue, there is no professional courtesy that is appreciated more than the house engineer returning from his or her days off to find the stage and console in the exact condition as they left it.

Conversely, nothing is more aggravating than discovering that things have been disconnected or moved from where they were when you left.

GLOSSARY FIVE
signal
source
clipping
Gain
bus
headroom
presence
gain structure

Ch. 6 - MORE ME!

"There's ringing in my ears!"
Peter Frampton

Let's talk about a subject that will turn up eventually like an ugly weed, or a nasty pimple, so we might as well nip it in the bud right from the get go.

It's called "**feedback**." They should name a comic book villain after it. You think it's dead, and then it comes back to torment you. It is the enemy and must be eliminated.

Feedback and "**overtones"** are the tricky gremlins that the engineer chases around like a gopher on the golf course. They are the stuff of nightmares.

Feedback results from sound waves that return back into a microphone from the speakers which are already recreating the sound from that same microphone, resulting in a cascading loop that produces a painful, non-musical squeal or rumble. These feedback events are alarming and disconcerting to musicians and audience alike. Like a spill or a leak in a boat, they can quickly get out of control and bring a show to a disastrous halt.

Experienced engineers can hear a squeal of feedback or an unnatural overtone in a mix and quickly adjust that frequency on an equalizer unit (or EQ) to restore stability to the mix. A highly-skilled engineer tunes these problem frequencies out of the system in advance by using their tools and skill set to "ring out" the system when the venue is closed, helping to avoid feedback incidents during the gig altogether.

A more subtle form of feedback is **overtones,** or "ringing." These are resonant tones audible in the mix that are not produced directly by the musicians but are instead resulting from sound waves being magnified by the structure of the room, or hardware like mic stands and often the stage itself. picking up vibrations and reinforcing them. Overtones add muddy tones to the mix that obscure the music, and when they are eliminated, clarity and balance of the overall sound are enhanced.

MIXING and TUNING MONITORS

The most common source of feedback is vocal monitors. The singer wants to hear more of their own vocal over the sound of the band. The input gain is turned up at the channel to get "**headroom**" in front of house. The engineer is walking a fine line between achieving the necessary gain and boosting the sensitivity of the microphone to the unstable point where the microphone starts to regenerate the tone from the wedge which is aimed right back at the singer.

It's tricky business, folks. There are many variables to juggle. The design of the microphone pickup pattern effects its feedback tendency. The height and angle of the microphone makes a difference. You can set it up where it's nice and loud with no hint of feedback, and then the singer shows up and changes the mic stand height or angle and gets blasted with a 2k ear bullet. High end feedback loops can even result unexpectedly from hats and glasses on the singers' heads.

The quickest fix for sudden surprise feedback is to grab the input gain and nudge it down just enough to make the mic behave without losing all the volume in front of house. Once again, the input gain pot is where the signal is at its most vulnerable point to become unstable. Once the feedback is decreased, you can look at the graphic EQ of the aux send and notch the frequency down that you have identified as the problem, which should allow you to bring your gain back up without the spike in the problem frequency. Identifying the frequency is the work of your perceptive ear and your digital tools.

Until you have developed a finely-tuned talent for naming those ringing frequencies, you can try whistling or humming the offending pitch into an RTA app to narrow it down, or just guess until you find it. Never boost frequencies during a show to find the feedback point. You should only use this crude technique when you are alone in a closed venue. This painful process is called "**ringing out**" a system, and doing it with customers and artists present is simply obnoxious.

A common mistake by beginners is to try to adjust the channel strip EQ to eliminate monitor feedback. This is ineffective because monitor sends are typically set up to as "pre fader" which means the signal goes to the aux mix send before the channel EQ is applied to the signal. If the "post fader" option is chosen, which usually adds the channel EQ to the signal before the aux send, the resulting dilemma for a front of house engineer mixing monitors on the same console is as follows:

When a fader is adjusted to change a signal level in the main mix, it changes the level of the post fader aux mix send to the monitor also. Changing master monitor mix levels during the show is poor form and causes distress to the performer. Imagine if your vocal were to drop in level dramatically in your monitor while you are trying to sing. That would really throw you off, like the lights going off while you are reading.

Ideally, the fine tuning monitor mix for each performer's wedge occurs at a sound check before the venue opens, or during a quick line check between acts. The engineer communicates to the band with a front of house "talkback" mic sent to the monitors, and the artist informs the engineer what to add or omit from their personal monitor mix. A second engineer or more astute band member might also listen to the monitor and suggest EQ adjustments to the engineer at the console.

EXERCISE

Sending the desired channel quickly to the desired monitor mix takes some practice. Learn how to have a fast response to commands such as the following:

"More center vocal in mix 2"
"Can I get more bass vocal in mix 1?"
"More drum vocal in center mix"
"More stage right vocal in stage left monitor"
"Bass guitar in drum wedge"
"Lead vocal in all mixes"
"All vocals only in their own wedge"
"Roll off some 2k in mix 3"
"Split the difference…"

MONITOR SEND EQ

Muddy overtones in monitor mixes are picked up in the main speakers and are also audible in the house off the back of the monitor wedge cabinets, which can muddy up a main mix as well. Monitors should sound transparent, meaning your voice simply sounds like your voice, not obscured by the added sound of a boxy, boomy speaker cabinet.

Artists sometimes ask for the monitor EQ to be adjusted to make their direct acoustic instrument sound better. This is disadvantageous since the monitor is set up to sound ideal for the vocal. The channel EQ doesn't change the monitor send. Switching the monitor send to post fader so the channel EQ will effect the monitor send is tempting but ultimately frustrating since the front of house level and EQ changes will effect the monitor balance. The best solution is to work with the instrument controls and the artist's onstage processing units to make the instrument sound better before it goes to the mixer input.

It's generally not good to insert compression on monitor sends. If a vocalist is trying to project their voice louder to hear themselves above the band but the compression is keeping that signal from getting louder, it's frustrating. A soft limiter on monitors is ok just to protect the speakers from blowing, especially if a full range program such as a DJ is being sent to the wedges. Delay and reverb sent to monitors is also best to avoid, since it makes the signal more unstable and prone to overtones and feedback.

DJ BREAKSALOT

DJ signals can do the most damage to speakers and monitors. They often send distorted signals from unstable, unpredictable sources. Venue managers mistakenly assume DJs don't need a sound engineer, but they really do. Give DJs plenty of headroom to be as loud as necessary on stage without clipping their output.

It's such common practice for DJs to peak their mixer outputs that they casually call it "rainbowing." Be sure to send the DJ's program heavily to the subs, and always insert a limiter or

compressor on DJ channels.
Always. For real.

PROS ONLY

Drummers and some other players sometimes want to hear kick drum in their wedge. It helps them not play as loud and stay in time. It can be difficult to make a kick drum in a drum wedge loud enough to be heard over the actual kick drum. Big concert stages often have a sub cabinet and wedge for the drum monitor, but in a small club, if your drum monitor is too powerful, the kick drum sound fills the whole stage and just makes the main mix more boomy.

In order to make the drummer content, try a couple of Jedi mind tricks. Put the kick up loud in the mains and subs while he checks his wedge, so he feels the thump in the house. Make sure the kick is not gated while the drummer checks, so the full wave of the kick signal is ringing out of the monitor. A gated kick just sounds like a weird click in a small wedge. If you can't get it loud enough for them, you may have to just apologize and say, "That monitor is meant for vocals. It doesn't really have enough power to handle very much kick."

I BROUGHT MY OWN MIC

Singers sometimes bring their own vocal mics. They often have extra gain and some enhanced frequencies compared to the house mics, so be sure to start with the channel muted, setting the gain lower and bringing it back up gradually. Singers should be aware that PA systems need some re-calibration for foreign vocal mics, so it's not an ideal scenario without a proper soundcheck. House monitors are set up for the house vocal mics. If their mic is a condenser mic with higher output, it could cause unwanted feedback before the higher gain is compensated for.

Singers will sometimes bring their own effects units for their vocal mics. These are often pedals not designed for balanced signals and can cause problems with unpredictable gain and monitor instability. Proceed with caution.

In-ear monitors also show up occasionally. You can attach the transmitter at the monitor aux send return point on the stage snake, or you can use a parallel output from the back of the monitor power amp (sometimes a 1/4 inch jack), depending on whether the artist prefers the wedge to remain on or off.

Wireless microphones are basically mandatory for rappers, auctioneers and wedding hosts. The receiver units will need to be located near to AC power and to the vocal mic inputs on the stage snake. The most common problem with wireless gear is a low battery, so keep spares handy. It once took me hours to untangle a huge wad of cables after I mixed a set with several rappers circling around each other with wired mics.

MICROPHONES

The typical microphone case lives on stage in a convenient area, usually near the amp rack. It should contain several vocal mics, instrument mics, condenser mics, and various drum mics. It also can be a repository for small spare parts like mic clips, small mic stand parts and spare small drum hardware parts such as cymbal felts, wingnuts and washers.

The Shure SM58 and SM57 are industry standard dynamic mics, first introduced in the 60s, that you will see on every professional stage, due to their predictable performance and durability. They are basically the same mic with different screens.

The 57 is best for instruments because the streamlined design puts the diaphragm closer to the source, and the 58's screen has built in protection from wind and breath noise making it better for vocals. They are both cardioid pattern unidirectional mics, which makes them pick up a source in front of the mic better than the sides and back. This helps keep unwanted sound from interfering with the sound being mic'd.

There are many upgrades available in the microphone market. It is best to avoid mixing mic types on stage. When you have several different types of microphones on stage, you have to balance a variety of different frequency responses and gain outputs. When your microphones are uniform, you will have an easier time getting the system EQ stabilized, especially the vocal monitors.

DRUM MICS

Clip-on drum mics made by brands such as Sennheiser and Audix have become industry standard. The Shure Beta 52 is a workhorse kick drum mic, and you will also need a couple of condenser mics for cymbals. In a small club, overhead mics are unnecessary since cymbals will fill the vocal mics whether you like it or not.

The intricate patterns of a swinging hi hat beat can benefit from a close condenser mic, aimed at the edge of the hats, in a position where the least amount of snare drum is hitting the hat mic directly. On a festival schedule with backline drums, it's better to have a snare drum mic on a small boom stand instead of a clip on, so the drummer can switch snares without the engineer having to unclip the snare mic every time.

DIRECT BOX

One helpful device usually kept in the mic case is a **direct box**, or "DI" (for direct injection), which converts the weaker high impedance signal from instruments with magnetic pickups to a low impedance, balanced mic-level input signal, and stabilizes the signal with the addition of a 3rd balanced conductor. DIs usually have a built in adjustable pad to cut the signal, and a ground lift which can be turned on or off to cut out a hum in the signal.

A DI also acts as an adapter from the quarter inch output of the instrument to the 3 conductor XLR cable that is generally connected to the mixer. The quarter inch inputs on a direct box act as both an in/out pass thru to the amp or as a pair of inputs that are combined evenly to the mono XLR out. Some DIs also need phantom power.

SNAKES

Snake boxes consist of a group of mounted XLR inputs and a large composite cable for running several XLR signals down a common path to the board. There are snakes that can contain all 24 inputs running to front of house and smaller "stage snakes" that can be attached to the main snake to run from other sections of the stage, such as the drums. The snakes have female XLR inputs on stage, and a fan out of male XLRs at the end where they connect to the mixer. They also can have a group of up to 8 mounted XLR male connections for the **"returns"** from the mixer that attach to the monitor and main amplifiers on stage.

MIC CABLES

You should have a crate of XLR microphone cables easily accessible, with enough cables to connect to however many channels your board is, plus a few spares. These cables should vary in length, so you can cover the appropriate distance without an excess of slack. Drum sets and guitar amps can connect to the drum snake at the rear of the stage with 8 to 12 foot cables. Vocal mics should be connected with 25 foot cables. You can always connect extra cable to the end of a mic cable that is too short. But using a cable that is too long for the necessary run makes for a messy stage.

CABLES & CONNECTORS

Cables used in sound systems come in several varieties and it is important to know the difference. A balanced XLR microphone cable carries a ground wire plus 2 conductors in reverse polarity that are combined and phase-corrected at the input in order to cancel out interference and allow longer runs without noise.

XLR (ground, left, right) cables have male and female ends which helps keep outputs from being plugged into inputs and vice versa. An unbalanced, high impedance 1/4 inch instrument cable has a ground and one conductor and is useful for guitar and instrument connections, but more susceptible to noise.

Speaker cables come in various gauges and the thicker they are the more power they can handle. Never use a 1/4 inch instrument cable in place of a speaker cable, or use a speaker cable to carry line level signal. Instrument cables are too small in thickness and too high in impedance to carry speaker signal, which can cause damage to the amplifier.

Speaker cables consist of 2 conductors that are attached with various types of connectors to amplifiers and speakers. "Speakon" connectors lock into place on the speaker cabinet input panel. When they are pulled out of the connector they can be repaired quickly with a small phillips screwdriver and a flashlight. Repairing and assembling Speakon cables is a very important skill to learn.

1/4 inch speaker cables plug in more easily to speaker cabinets and will yank out more easily if the cabinet is moved too far or the cable is tripped over. This can actually be an advantage over the Speakon, which is often broken and needing repair after being kicked or yanked, as opposed to just being temporarily disconnected.

Banana plugs on speaker cables attach to the output poles of an amplifier. Speaker cables can also be connected with bare wire to these poles on the back of the amp. Banana plugs can be flipped and or stacked by design to allow for intentional phase reversal, and for connecting to 2 positive poles for "**bridging"** a stereo power amplifier into a single higher wattage amp. Amplifier poles are red and black for positive and negative, and when bridging you connect the two wires to both red poles.

SELF-CONTAINED SPLIT SYSTEMS

Some bands will show up with an entire set up of mics, stands, wedges an/or in-ear monitor rigs, and a monitor mixer. This requires the stage to be cleared of most of the house equipment, and the insertion of a "split" snake system before the stage lines go to front of house, so the signals can also go to the monitor rig. In some cases, the front of house mix is also made with the band's custom console, and you will just need to patch in a stereo pair of inputs to the house PA.

If you have any aux sends such as delay speakers, fills, and subs other than stage monitors, you will need to go through 2 channels on your house console. If not, the guest mixer main drive outputs could go straight to your house amp rack processor inputs, but you will not have the added benefit of inserting any limiters, EQ or other processing to protect your PA.

This type of arrangement also facilitates multi-track live recording projects. Be sure to allow extra soundcheck time. If a band attempts to do any of this during a short festival break, they are on the wrong end of a tough learning curve.

WRAP IT UP

Cables should be wrapped and put away with the "over/under" technique, and secured with velcro ties. There are many variations on cable wrapping technique, and it seems like everyone has their own style. It's not about speed as much as it is about the cable coming unwrapped easily without making knots. You will find that the repetitive process of wrapping cables after a long show is kind of a meditative practice in feeling a sense of closure. Practice makes perfect.

GAFFER TAPE

Never use duct tape to tape down cables. Gaffer tape is designed to not leave residue on the stage or on your cables.

When taping down cables, never run the tape down the length of the cable. It will roll up on itself when you try to remove it and become impossible to unwrap. Run it in 5 inch strips across the cable at 1 foot intervals, like stitches. It uses less tape, it sticks to less of the cable, and is way easier to pull up. You will thank me later.

MIC STAND MAINTENANCE

Mic stands are simple mechanical systems that often become unusable because a small piece of hardware has been lost or damaged. A good sound engineer keeps spare mic stand parts and knows how to assemble a mic stand from a set of small pieces. A whole show can come to a halt because of a tiny bolt or washer in a mic stand that has become stripped. Having these spare parts handy and testing the functionality of your stands on a regular basis is a huge part of having a professional stage presentation.

Performers like to fidget with stands in the awkward down time between songs and over-tighten the bolts that keep the stands straight, so it is on an ongoing project to keep your stands functional. Investing in high-quality stands that wear out more slowly is more cost effective in the long run. Look for brands such as "Atlas" that cost more but last much longer. The cheap brands can often break on or before the first use. There are brand new low-quality stands on display in the store right now that already have stripped threads, so buyer beware.

I am a "mic stand whisperer." I can take the parts of 6 supposedly "broken" stands and make 5 perfectly good ones. Someday I am going to start an emergency mic stand repair service called "Frankenstand".

HEADPHONES

Some live engineers use headphones. The phones plug into the console and normally contain the main mix. When the PFL, or sometimes a "CUE" button is selected, the individual channel or send can be heard in the headphones without affecting the main mix. They need to be good isolating headphones and cranked up real loud to hear over the room sound. Headphones can help you dial in effects, diagnose feedback in monitor sends, and check the quality of input sounds.

EARPLUGS

It's hard to mix with earplugs in. It's good to wear them in loud places when you are not mixing. My last hearing check showed a slight hearing loss around 3k from being on small stages with cymbals crashing right by my ears for many years.

The irony is, the frequencies where you have suffered hearing loss are the ones that aggravate your ears the most. That's why I don't care very much for cymbals. I always want to duck 3.15k in the main mix EQ, but I know it's probably just me.

GLOSSARY SIX

sound system
signal chain or path
pre amp
line level
flat
unity
sends
returns
direct box
condenser mic
pre/post fader
dry signal

wet signal
feedback
cue
bridging

Chapter 7 - FEEL THE POWER

"We're gonna blow a 50 amp fuse…"
- The Rolling Stones

A sound engineer should have a basic knowledge and a healthy respect for electricity. It is the fuel that keeps the engine running, and if there is a problem with the power, the whole system can fail. Electrical power problems are the kind of thing that causes sound engineers to have nightmares, so the more you know, the better you will sleep.

The amount and density of electrical energy is defined in many terms, the most common of which are "**volts**" and "**watts**."

The electrical power flowing within the component circuits of a sound system is widely variable. The power from a household electrical outlet is 110 volts, which could give a person a serious shock. The electrical power of the signal in a connected microphone cable or guitar patch cable is just a fraction of a single volt, which is normally too small to feel by touch.

The power from a headphone jack is less than 2 volts. A 9 volt battery can power a small radio. You can test a 9 volt battery with your tongue, and you can also test a speaker by connecting a 9 volt battery to the speaker connectors and seeing if the coil moves.

The power in an amplified signal going from amplifier to speaker varies widely on the voltage scale depending on the level of the signal. You can get a mild shock or see a spark from a loose speaker cable hooked up to an amplifier.

If the electrical power in an outlet or faulty equipment on a stage is wired improperly or has caused a short circuit, it can cause a partial amount of voltage to travel from the microphone through the performer or engineer's body, which can be alarming and painful, but only rarely deadly. Testing mics for stray current can be done with the back of the hand or a guitar neck to avoid the much more painful lip shock.

BREAKERS

The AC electrical circuits originate from fuses or "breakers" in discreetly placed wall panels in the building which act as a safety measure to disconnect the flow of electrical current when it exceeds a specific capacity. That capacity of current is measured in a unit called "amperes," commonly abbreviated as "amps" or "A."

A household outlet is usually 20 amp capacity. If the current exceeds that amount, the breaker will trip and power will be lost. The voltage on the standard US household circuit when it is connected is 120 Volts.

The wattage being produced by the sound amplifier creates a "load" on the electrical circuit. If this load is too great, it will surpass the capacity of the breaker and the circuit will break.

The general formula is that the potential wattage of the amplifiers should not surpass the voltage of the AC power (usually 110v) multiplied by the amperage of the breaker (20a) or the circuit will fail. An average sized power amp with two speakers connected per channel is using about 2500 watts of power on paper, which would be pushing the limit of a 20 amp circuit by the standard formula.

However, power amps are almost never steadily running at their peak wattage capacity, because music tends to have spaces and lulls between the beats. In order to trip a 20A breaker with one 2500w amplifier, the amp would have to be constantly creating a huge, peaking sound wave at full blast on all channels, which rarely happens.

In general, each 20 amp circuit will handle system amplifiers running at about twice its estimated limit, up to around 5000 watts. If the music is dense with steady, heavy low end peaks such as electronic DJ music, this might be approaching the critical point, but the average rock band will be fine.

It is wise to distribute the amplifier loads evenly across the available circuits. Try to put the mains and front of house console on one circuit, the subs and monitors on another. Consider in advance what will happen if a circuit trips. The subs use the most power and are usually the first circuit that gets overloaded. If the mains are on a different circuit, the subs might cut out, but the mains with the vocals are still on and the show is not completely derailed.

Be mindful of where the crossover is getting power. If it's on the circuit with the subs, you will lose the main signal too. If the monitors go with the subs, but the back line and the mains still work, the show might limp along for a few more songs, instead of the buzz kill of the main PA signal suddenly cutting out.

Let's say you have been provided just two 20 amp circuits at a mobile event job. Always make sure to request "**dedicated**" circuits that are not attached to any other appliances or lighting. Margarita machines have blown countless stage breakers. You should be okay putting two 1000 watt main speakers plus the backline guitar amps and the front of house console on one circuit, and two 1000 watt subs and four 500 watt monitors on the other circuit.

DIRTY POWER

AC electrical power is often contaminated by surges, spikes and interference. A **power conditioner** is a rack mounted AC distribution unit with several outlets that filters and stabilizes AC power. It can help your equipment run more efficiently and shield against damage done by faulty power.

If it seems like the provided AC power might be inadequate, there are ways to mix the band more conservatively to save power. Some power conditioners have voltage meters which will dip when the power is being pushed toward its limit. Diminish the amount of low end signal in the main mix by using high pass filters and lowering the faders on kick, bass, floor tom and other dense, power-taxing signals. Insert limiters on the heaviest peaking signals to keep the peaks from spiking the power amps. Turn the overall mix volume down.

There are of course different voltage power outlets, such as those for large appliances and those in foreign countries. Plugging your equipment into these is a one way ticket to the repair shop. And there are different size circuit breakers. When in doubt, seeking the expertise of a licensed electrician to provide and test the AC power may be your safest solution for some jobs.

A good **generator** can be one of the cleanest and most consistent power sources. A gas generator no bigger than a couch is usually plenty to push a decent size PA system at an outdoor event. Be wary of any type of "solar powered" or alternative energy electrical power. The conversion systems in these alternative methods can be inconsistent and prone to harmful spikes and circuit failure.

Some household outlets have "GFCI" circuits in the outlet itself, usually found in restrooms and outdoor plugs. These are a different kind of safety breaker designed to detect inconsistencies in the ground voltage that can occur when current is accidentally passing through water or a foreign object, like when a hair dryer falls in the bathtub, or a plug connector is in a puddle.

It is a good idea to avoid GFCI outlets for your mobile sound equipment. They can be tripped by vintage amplifiers and instruments that are not actually that dangerous, but just unstable enough to blow the GFCI and cause the power to be cut and the show to be delayed unnecessarily.

The point of all of this is that there is a maximum amount of wattage that an electrical circuit can handle. Depending on the volume of the music, the size of the room, the size of the band and many other variables, you should be able to run a decent sized sound system off of a couple of 20 amp circuits.

I have operated hundreds of PA systems on 1 or 2 circuits. I have also blown more breakers than Clark Griswold in *Christmas Vacation*.

AMPLIFIER POWER

Amplifiers in sound systems utilize the powerful alternating current or "AC" power from a standard electric wall outlet. This 110 Volt current alternates between two conductors in a standard 60hz wave. These AC circuits also use a third conductor connected to the earth called a "ground" which acts as a default destination for excess or stray current.

Sound system amplifiers combine this AC electrical power with their internal circuits of tubes, transistors, and transformers to boost the very low voltage signal in an attached microphone cable up to a higher powered signal. This boosted wave signal travels through thicker cable connected to the poles of a magnet that moves the coils in a speaker.

The energy that results from the amplifier powering the speaker is measured in units called watts.

SPEAKER RATINGS

Speaker coils, like light bulbs, contain a resistance to the voltage from the amplifier, which is measured in units called

"**Ohms."** The resistance results in a physical expression of power, which is measured in "**Watts.**" A light bulb filament expresses this resistance by glowing with light, and the speaker coil expresses the resistance by vibrating the speaker, producing sound waves.

Household light bulbs are 40 to 100 watts. PA Speakers and amps range from 100 to several thousand watts.

Most single speakers are rated 8 ohms. Adding a second 8 speaker on the same amp drops the "ohmage" rating to 4 ohms, and adding 2 more speakers drops the resistance to 2 ohms. The lower the ohm load on an amp, the higher the wattage rating.

Speakers are mounted in cabinets designed to enhance their sound reinforcing properties and efficiency. Speakers can be wired together in "series" or "parallel" configurations, which affects the ohm load in different ways.

The patient people who manufacture and repair the amplifiers know these complex electrical formulas very well, but live audio engineers mainly need to know that amplifiers and speakers should be matched by the wattage and resistance ratings that are in their posted specifications.

The wattage ratings of speakers are often exaggerated and inconsistent. Very simply stated for the audio engineer's practical use, the more speakers you attach to one channel of an amplifier, the more wattage the amp should be rated. A mismatch or overload can cause circuit breakers to fail, and create distorted signals that cause speaker coils to fry.

PROS ONLY

*"**Bridging"** an amplifier by linking the two channels to form one channel is a way to achieve more wattage, but it uses more electrical power and makes the amp more prone to overloading. Bridged amps usually cannot handle a resistance load under 4 ohms. Use bridging with caution. Doubling the wattage of an amp does not make it twice as loud. The best way to get more volume out of a system is to add more speakers, not just to add more*

amplifier watts.

STAGE POWER

Your stage should have at least two or three circuits accessible for the sound system amplifiers and several power strips or quad box outlets arranged behind the guitar amplifiers. Bands will often ask for power at the front of the stage for pedals.

It is a good idea to have some 3 prong plug adapters handy. When guitar pedal boards or keyboards and guitar amps are on different circuits, there is often a 60 cycle ground hum in the speakers generated by a slight AC voltage that is leaking between the two circuits. Lifting the ground on one or the other of the plugs with an adapter will quickly correct this issue, as will using an extension cord to put the amp and the instrument on the same circuit.

It's also good to have your front of house console connected to the same circuit as the main speaker power amps for this same reason. A 60 cycle hum can emerge in the mains from voltage traveling between two separate circuits.

GLOSSARY SEVEN

dynamic
signal
volts
watts
circuit
ohms
acoustics
stage right
downstage
series
parallel
ground hum
ringing out

bridging

CH. 8 - EXTRA CREDIT

BUILDING AND MAINTAINING A MOBILE SYSTEM

I did thousands of mobile event production jobs by myself with a pickup truck and and a 4 wheel dolly. The equipment that my dolly and I transported during those decades altogether probably equaled the weight of the great pyramids of Egypt.

I eventually put together a self-contained system that included a mixing console, 5 passive wedges, a rack of amps and two 2x18 subs. The modern paradigm for small venue and mobile event sound systems is even more streamlined now, with a digital board and an iPad mixer with a digital stage box. No more snake, no processor rack, no mixing console. The convenience of a digital board is the self-contained processing, the mobility of the mix position, no awkward mix station in the middle of the room, being able to mix vocal monitors on stage with iPad in hand, and the convenience of bringing up your saved basic scratch mix instead of starting with a completely flat board.

GOING MOBILE

Here's a list of necessary and helpful items for a basic mobile or small venue PA company. It's good to have a printed check list you can refer to while loading the van. It's easy to get distracted while loading up and forget some small item, such as an IEC power cable or iPod adapter cable, which the show cannot go on without.

I hate to admit I have forgotten to load the mixer or the microphone case more than once.

MOBILE PA COMPANY CHECKLIST

2 Main speakers, active or passive
4 or 5 stage wedge monitors
1 or 2 Subwoofer speakers
Amp Rack with monitor and main amps
Sub Amp with Crossover
24 channel Mixer
Microphone case
Mic cable case
Case with DIs, patch cables, adapters, etc
IEC Power cables for all amps
Speaker cables for subs and monitors
24 ch. Snake
8 ch. drum snake
1/4 to xlr aux snake for amp patch
6 Vocal Mic boom stands
6 Short instrument mic stands
Hi hat mic stand
kick drum mic stand and mic
set of drum mics
Speaker stands or sub poles
Extension cords, power strips

iPod with cable and AC adapter
Dolly Cart
drum carpet
Gaffer tape
Masking tape
Sharpies
Flashlight
Business Cards
Directions to gig
cell phone, charger
Speaker straps

OH SHIT BAG

It's good for a mobile system engineer to have a case or bag that you travel with that contains a variety of extra small items that are easy to forget. These are the small items that break the camels back, like the horseshoe nail in the old proverbs.

You can order these cheap items by the dozen online so you don't have to run to the closest Radio Shack and pay retail price at the last minute:

IEC power cables
1/8 inch Y adapter cables for iPod, etc.
banana plugs
speakon speaker cables
short xlr patch cables
XLR to 1/4 patch cables (You will need more male than female)
insert cables for analog processors
phone charger
iPod charger
rca to 1/4 bullet adapters
AC power tester for outlets
9 volt and flashlight batteries

IDIOT CHECK

When you load out of a mobile production event at the end of a long day of loading gear in and out, it is tempting to just close the van door and drive away. Do yourself a favor and go back one more time after you think you are done. Nothing is worse than realizing hours or days later that you left something small like an extension cord or a drum carpet at the venue, and it is often very difficult to arrange to get access later to recover it before it disappears, not to mention unprofessional.

PEOPLE SKILLS

The stereotype of the surly sound engineer is an unfortunate reality. Many veteran engineers are overextended and stressed and have developed a thick skin to avoid conflict. They are often territorial and resentful of any interference to their necessary routine.

It is good to keep perspective on who the show is about and try to always say yes to the artist's requests, as frustrating as they may be. There are many typical pitfalls that sound engineers fall into after too many shows. Be aware of these bad habits and take measures to avoid them as they inevitably come up. The first problem to emerge is usually inattentiveness. Make an effort to keep eye contact with your artists, especially between songs and at the beginning of the set when they often need attention. You can approach the stage after a song or two and ask them directly if they need anything adjusted. Routine mixing techniques can also be a problem. If you mix every band the same way, you will miss the opportunity to mix with an ear for style. Country western bands are playing music from an era where kick drum was not very prominent, so it's just bad form to have hip-hop level low end in a honky tonk mix.

It's always important to have a tight relationship and communication with your venue managers and security. They will rely on you in situations where announcements need to be made and schedules adjusted. You will need their assistance sometimes as well, like when an unruly patron who is harassing you requires ejection. If a customer touches the board, even as a joke, they are out the door before you can say, "What does this button do?" Answer: It's the "customer ejection" button.

If there is a guest engineer with a band, be gracious and hospitable. Help them get started and then stand by politely in case they need a hand. Ask if you can get them a drink.

You will need a reliable device, not your phone, to play appropriate music between sets and keep the energy lively. You are going to become a DJ by default whether you intend to or not.

You can also make your job more enjoyable by recommending good talent. It sucks going to work every night and mixing acts that you are not impressed with. It makes the job more like mundane labor. Mixing bands that you enjoy hardly seems like work at all.

DRUGS & ALCOHOL

Another typical pitfall in the business is relying on substances to relax after a long stressful day's work, and sometimes during, since we work in bars and dance halls where legal and illegal chemicals are easily available. Abuse and addiction are occupational hazards.

Just as driving while impaired is dangerous because you are behind the wheel of deadly machines, doing sound under the influence can have disastrous repercussions. Your response time is slowed and your ability to trouble shoot is impaired. Your senses are skewed and you have an altered perspective on the way it sounds. It is just a bad idea to mix under the influence.

OTHER AUDIO ENGINEER FIELDS

There are many other more technically sophisticated fields of sound engineering that are also highly worthy of pursuing as a career.

The job that is concerned with assembling the front end of the system by combining the ideal configuration of various models and sizes of sound reinforcement components, like amplifiers and speaker cabinets, is a very important resource for any professional venue. Marketing and selling this equipment can be a very rewarding line of work also.

Studio recording engineers work with much more sophisticated audio equipment at a much finer level of audio processing. Studio engineers and live audio engineers share some skills and techniques in common, but are really performing two very different jobs. With rare exceptions, they tend to be two very different types of characters in general. Most engineers tend to strongly prefer one or the other discipline.

There is also an advanced stage of audio engineering called "**acoustic design**" which is concerned with the effect on the sound waves within the room as they interact with the architecture while traveling between the speakers and the audience. Sound waves travel in a line, and they also are bent, reflected, absorbed, and dissipated. An expert on acoustics can be a great asset to a new venue being built with a high-quality listener experience as a priority.

All of this advanced audio technology works together to enhance the musical experience for musician, technicians, and audience.

PHASE

Advanced engineer jobs like installation and acoustic design will be very concerned with the behavior of sound waves created by speakers in a room. When sound waves are converted to electrical energy, the waves are represented by positive and negative peaks and valleys. You can see these in graphic form on devices that display sound waves on screens.

In the observable physical world, these waves are represented by the movement of the speaker as it vibrates. When the speaker pushes out of the cabinet, the top of the wave is cresting. When the wave is at the low negative point, the speaker is withdrawing into the cabinet.

The term "phase" is used to define the synchronization of sound waves and signals. If speakers are out of phase, they are moving in and out at opposite times. Speakers often become out of phase in systems due to mistakes in installation. It is very easy to flip the polarity of a speaker wire inadvertently. When speakers are out of phase, the sound waves are in conflict and the result is a disorienting sensation to the untrained ear. It might be described as hollow, thin, or just plain weird. When conflicting sound waves combine to negate each other, they are said to be **"canceling**" each other. This effect can sometimes be used intentionally, such as in "noise-canceling" headphones.

EXERCISE

Connect an iPod and play a song with a heavy, slow and steady bass drum beat. Open up the front screen on a subwoofer cabinet and observe the speaker cone moving in and out with the beat. Connect a second speaker cabinet to the same source amplifier and observe the difference when 2 speakers are moving together on one power source. Now flip the polarity on one speaker by connecting the speaker cables backwards at the amp. This is easy with banana connectors or bare wire leads. Notice the change in sound when the speakers are out of phase. How do you describe the phenomenon? Perceive the difference between one speaker, two speakers in phase and two speakers out of phase.

SIGNAL PHASE

The term "phase" also applies to mic and instrument signals.

Professional mixer consoles often have a channel strip control that flips the phase of a signal in the signal path. Inverting the phase of an instrument signal can be helpful in a variety of ways. One common example is a kick drum. Due to the high volume of the kick drum sound wave, the abruptness of the signal, and the drums' rear position on the stage, the sound wave created by a kick drum head itself can often be in conflict with the sound waves coming out of the mains and subs.

Reversing the phase on a kick drum can often help align it with the system. By trial and error, a sound engineer can experiment with flipping the phase on the kick drum or other instrument signals and discern with a well trained ear which phase position "in or out" is best overall.

Phase issues also occur due to mic placement. The signals produced by mics placed underneath or behind drums and behind open guitar cabinets will be out of phase with mics on top and in front. These phase effects can be useful sometimes, but generally when one of these signals is flipped to be in phase with the other, the sound is improved.

Another good application for the phase button is to help minimize feedback in the monitors. Since the sound waves from vocals and acoustic instrument monitors is aiming backward at the stage, flipping the signal phase on a signal that is strong in the monitors can sometimes help stabilize these signals on stage.

PHASE ALIGNMENT AND DELAY

On larger stages, the instrument amplifiers and drums are far enough back from the mains to be in conflict with the sound waves coming from the main speakers, so the audience is often being projected with sound waves from different sources that are not in "alignment." The same applies to satellite speakers in large rooms or outdoor spaces that have auxiliary mixes for the rear of the crowd. Digital processors and consoles can be engaged to align these "delay speaker" signals with the mains so that the sound waves blend in sync instead of conflicting or "phasing," creating a delayed, echo effect.

Other forms of sound wave distortion and interference that happen in the space after the sound leaves the speakers are "comb filtering" and "interference," which occur as the sound waves from different sources collide and cancel each other out. Experienced engineers will minimize these problems through careful configuration and placement of the cabinets and will put processors in place during installation so that the whole system is dispersing sound waves the most efficiently and in the best possible alignment throughout the hall.

SYSTEM PROCESSORS

Modern high end systems utilize digital units in the amp rack that perform the function of speaker crossover plus an array of system processing functions. A unit such as the DBX Driverack has a pair of stereo inputs and multiple stereo outputs for lows, mids and highs. Each of these 6 output channels has discreet functions for EQ, compression, limiting, phase delay, along with other processes and presets that can help fine tune a PA system.

SYSTEM TROUBLESHOOTING

There are many engineers working in venues today who have very good ears for mixing and a strong grasp of the techniques and skills necessary to mix a show effectively. But there are only a handful who have learned the nuts and bolts details of how systems are put together under the hood and are able to track down the inevitable problems with systems hardware that result from the typical wear and tear and chaos of a live music venue.

Patch cables get unplugged, speaker cables get kicked, mic cables go bad, and all at the worst time, in keeping with the sadistic law of that evil bastard named Murphy.

An experienced engineer can shift into troubleshooting mode with the focus of a NASCAR pit crew and the resourcefulness of McGyver to keep a deconstructing show on track.

The specific trouble is almost always, "Why are we not hearing the signal?" You need to put on your best linear thinking cap and trace the entire signal path from the source to the speakers to figure out where it is being lost.

The main technique for this real time troubleshooting in the field is the **"scientific method."** This is the method of experimenting with each of the variables in a system one at a time until the culprit issue is found.

The first step is making sure your signal source is reaching the mixer. You can see this with the PFL meter on the channel. The source signal from the instrument is usually the problem. Common examples are bad guitar cables on stage, or a pickup in an instrument with no output due to dead batteries, etc. If there is no signal reaching the mixer, utilizing the scientific method, you can test a source on stage by removing a cable from the direct input and plugging it directly into an amplifier that is already proven to be working.

Troubleshooting is also like a basic flow chart:

Does his guitar amp work?
YES.
Then - Plug your guitar cable into his amp.
Does your cable work now?
NO.
Then - Try a new cable.
Does it work now?

YES.
Then - Throw your cable in the trash
and get a new one.

This technique applies to every part of the system, and once you learn how to successfully implement it, you will find it useful on a daily basis.

MONITOR TROUBLE

The other most common application of this troubleshooting method is for monitors. Let's say no signal is reaching a wedge, we'll call it mix 1. Send the same signal to another wedge, for example mix 2. When you hear that signal, disconnect that speaker cable from the mix 2 working wedge and connect it to the non-working mix 1 wedge.

Is there signal in mix 1 wedge now? If not, your mix 1 wedge is probably fried, because the cable worked in that wedge but not this one.

If there IS signal in the mix 1 wedge now, the problem may be the mix 1 speaker cable. Try the cable from the mix 1 wedge in the mix 2 monitor cabinet. (It really helps if all your speaker cables are labeled).

Is there signal now? If not, your mix 1 cable is bad. If there IS signal, we know the mix 1 cable and the mix 1 wedge are both good, so next we look at the power amp.

Is there a signal light on the amp channel indicated for mix 1? If not, there is an issue in the patch cables between the mixer aux send and the amp. If there is good signal indicated on the amp and the cable is good and the monitor is good, it would be a real mystery…

Perhaps the amp needs repair. At that point, you will use the same channel-switching method on the patch cables from the return snake to the amp to figure out if the amp has a bad channel. The channel is receiving signal but not sending it out. Try a different channel or a different amp.

DUST TO DUST

Nine times out of ten, the reason that mixers and amps need repair is just to clean out dirty connections. Dust and moisture are the mortal enemies of all sound gear. Equipment installed in clubs becomes extremely dusty very quickly. Very often, you will find that a mysterious signal loss can be fixed by wiggling a dirty gain pot.

Always cover the mixer when not in use, and dust off the console and power amps every few days with a paintbrush and/or a can of compressed air.

GLOSSARY EIGHT

power conditioner
processors
signal phase
speaker phase
canceling
delay speakers
acoustics
scientific method

FURTHER RESEARCH

For those with extra sensitive ears who are intrigued by the complex relationship between frequency tones and musical notes, you might choose to explore the vast world of "**harmonic theory**," and potentially pursue such challenging careers as piano tuning, instrument manufacturing, or advanced system design.

PhD dissertations and thick books have been written in an attempt to answer such mysterious universal questions as:

"The 440hz vs. 432hz western music standard conspiracy,"

"Why instruments sound strange when tuned to mathematically accurate frequency ratios,"

"What if middle C was a lower octave of 1k instead of nearly 1.05k?"

"Are Robert Johnson and Chuck Berry's master recordings sped up?"

Google "equal temperament" for a fascinating rabbit hole of various harmonic hypotheses involving cultural evolution, the golden ratio, complex logarithms, and human physiology.

A PARTING TALE

Years ago, a golden disc went out into far outer space aboard the Voyager satellite containing recordings of human languages, nature sounds, classical music, and popular tunes.

It was reported that many years later the first alien transmission was received in response:

"*SEND MORE CHUCK BERRY!...*"

The signal has finally completed its long path from source to listener. We are now at the end of the line, and the beginning.

This phase of your training is complete, and it's time to get valuable experience out in the field and learn these skills the best way possible: by trial and error, loud music, and hard knocks.

Above all, have fun! Good luck, and we're all counting on you...

ONE PAGE QUICK GUIDE

Print this page and take it for helpful reference on your first sound jobs.

SHOW UP BEFORE THE BAND

TURN ON THE MIXER FIRST

SET UP MIC STANDS

RUN ALL MIC CABLES WITH COIL AT MIC END

SET UP STANDARD STAGE PATCH

LINE CHECK ALL INPUTS

CHECK ALL VOCAL MICS

ADJUST MONITOR SEND EQs

HIGH PASS ON SELECT CHs.

ASSIGN SUB GROUPS

INSERT AND SET UP COMPS

SET UP EFFECTS

SET UP GAIN STRUCTURE

SOUND CHECK

TRAVELING ENGINEER KIT:
mic package
headphones
flashlight
music device
phone charger
device chargers
1/8 to RCA adapter
sharpies
masking tape
gaffer tape
iPad with RTA, digital console software, clock

ABOUT THE AUTHOR

Paul Minor is a 30 year veteran of the Austin music scene and a graduate of St. Edward's University Masters in Human Services program. He has toured with major artists including Fastball, Joe Ely, and the Texas Tornados, and has installed, maintained and operated sound systems in numerous venues throughout Texas. Paul has been production manager, booking consultant and head engineer at the Austin Beer Garden Brewery since 2013.

This textbook is a work in progress.
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