How to Run Sound

at your local music show

by Davy Goldsmith

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Every local music show needs someone to run sound for it. It's a crucial role that needs to be filled when booking and organizing. If you're able to do it yourself (DIY!), you're able to fill an important need for your local music scene! It could even become something you do on a professional or creative level!

There are lots of ways to run sound, but I will be explaining how I choose to run sound for the events I help out with. I prefer to use straight forward methods that give me the best results for my time and effort. There are a lot of cool and creative methods that I won't cover (such as parallel compression and double micing) that are really interesting but don't audibly change the sound for the better in small or mid-size venues, at least for me and the general audience. So, I don't use these methods regularly and they may be more than you need to know when starting off.

Contents:

Inputs and Outputs

When you're getting ready to run sound for a show, you should first decide what you're going to be running through your sound system. This could be any and all instruments in the band. You should also think about what speakers you'll use and if you'll provide monitors. These are called your "Ins and Outs." The signals from audio sources are your "inputs" and the speakers and devices for projecting or capturing sound are your "outputs."

A common question I have for bands I run sound for are, "What's your instrumentation?" This question helps me decide what my "inputs" are. In other words, it helps me decide what I will be micing up or providing inputs for.

Another consideration is the size of the venue. Is the venue small enough so that some instruments make enough noise where they don't need to be amplified by my sound system? If they do make enough noise, I won't bother micing them or if I do, I end up muting the channel and they won't be considered an "in."

Small venues (up to 150-person capacity.)

- Inputs: vocals, backtracks, and quiet instruments that don't have their own amps. Backtracks usually come from a computer or DJ mixer and quiet instruments may be violins, upright bass, or a guitarist.
- Outputs: Your sound system can be one or two speakers and you may just have a monitor for the lead vocalist and/or drummer if any at all.
- You might not have much control over the sound of the performance and find the sound system struggling to bring up the loudness of the vocals to match that of the drums and other instruments with amps.

Mid-size venues (150 to 1000-person capacity.)

- Inputs: Each instrument and amp is typically miced and the drums may have a kick mic and one or two overhead mics.
- Outputs: Each musician may have a monitor and you'll have main speakers capable of more sound.
    - Recording out: When you're micing each instrument, you have an opportunity to record the show. This can be a stereo or mono output from a tape out or monitor output, or if your mixer has direct outputs on each channel, you can record to a DAW on your computer and mix after the show. You can use a cassette deck to record by using an aux output and mixing separately on that aux channel and monitoring through a headphone jack on your cassette deck or mixer.
- You'll have more control over the sound of the performance but will likely still find the vocals are the loudest in because they are still being pushed the most to be brought up to the other instruments and inputs.

Large venues (beyond 1000-person capacity)

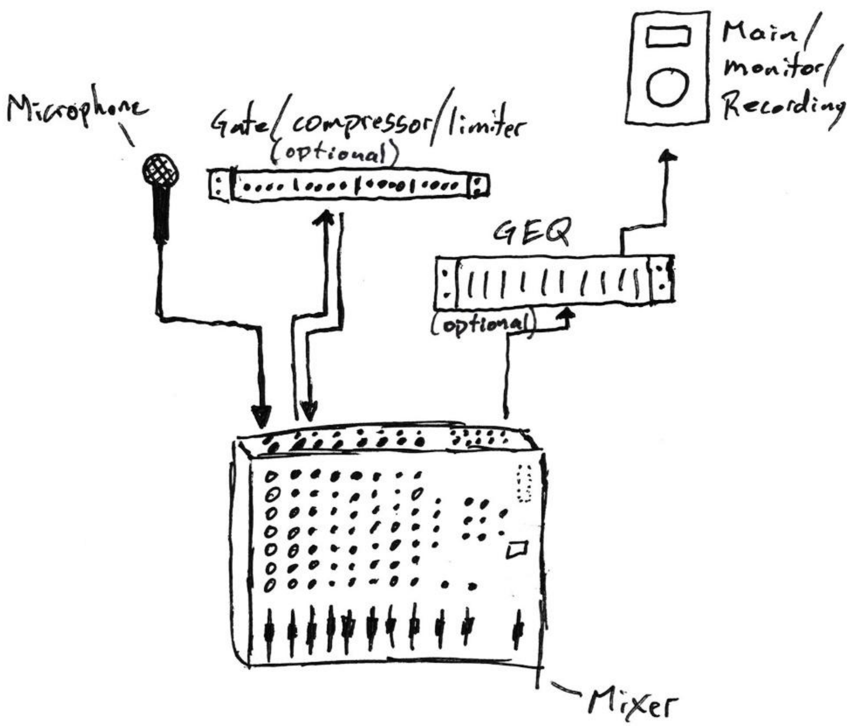
I won't talk much about the methods of shows of this size because a show with over 1000 people in attendance can hardly be considered a local show. However, the techniques, equipment, and methods are still theoretically similar to mid-size and small shows.

- Inputs: Each instrument and component of the drum set is miced. You may double mic some performers and using advanced techniques to capture sound.
- Outputs: Each musician will have a monitor and you may provide the lead vocalist with two monitors and the drummer with a sub under their standard monitor.
- You'll have nearly complete control over the sound of the performance because the sound generated on stage becomes insignificant to the audience when compared to the sound levels of the sound system.

Signal Flow

A crucial concept for anyone who runs sound is Signal Flow. This is how and through what pathways the audio signal travels from its "in" to its "out," or more typically, how sound gets from a vocalist to the speakers.

Generally, audio travels from an audio source such as a vocalist, instrument, or device into a microphone or DI (direct input) box, though an xlr cable, into an audio mixer where it is processed, filtered, and mixed by compressors, gates, limiters, graphic equalizers, effects processors, (amongst others) and the mixer itself, and is sent to the amplifiers and speakers and/or recording devices.



I'll explain audio equipment in the order through which the signal travels through the system.

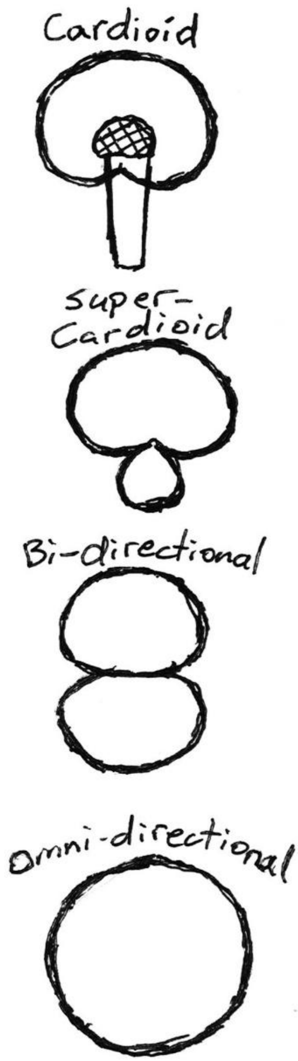
Microphones

There are a lot of different microphones that are designed and used for picking up different audio sources, but there are three types that are defined by their design.

1. Dynamic – These are the most commonly used and work like a speaker driver, except in reverse. They're great for live use because they pick up audio close to the mic and not a lot of the background noises. They're also super durable and affordable.
2. Condenser – These are the next most commonly used and use a capacitor in the diaphragm to capture audio. They pick up a wider area of audio sources and are good for picking up a general sound at a further distance from the source. They require **Phantom Power,** which should only be used when condenser mics are connected to the mixer. They are more expensive and fragile than dynamic microphones. There are also two types of condenser mics:
    a. Large Diaphragm: These are physically larger and are used when low frequencies are present.
    b. Small Diaphragm: These are a little smaller and skinnier. They only pick up mid and high frequencies well.
3. Ribbon – Don't even with this type. They're super old technology and sound bad and really canny, are expensive, and super fragile. If you ever see one on stage, you're allowed to roll your eyes. And even if you do, they're probably just a dynamic mic in disguise and someone wants to be like Elvis or the Beatles. Gross.

Polar Pattern

All microphones also have a Polar Pattern. This describes what areas they pick up and how well. There are a lot of different patterns, but I always recommend using a cardioid. There's not much multiple cardioid mics can't do better and cheaper than other polar pattern mics, but you do you and we'll go over them all anyway.



- Cardioid: This is your most common pattern. It picks up most of the audio signal directly in front of the microphone and none from the direct back. This is great for reducing feedback from a monitor and is perfect for your typical audio source which would come directly in front of the mic. It does allow for some audio on the sides, but the bleed from other audio sources can be managed.
- Super-Cardioid and Hyper-Cardioid: These are similar to the cardioid, but they allow less audio to come from the sides while allowing some audio to come in from the rear. This does help reduce bleed a bit but is more prone to feedback from a monitor.
- Bi-Directional: Audio signal from the front and rear is registered at equal strengths.
- Omni-Directional: Audio signal from all directions is registered at equal strengths.

What mics for what inputs?

Vocals: an SM58 is a classic choice but if you want a more affordable option or if you don't trust the vocalist to be gentle with your mic, a Behringer XM8500 is a fine mic as well. These are dynamic mics with cardioid patterns. The closer the vocalist can bring the mic to their mouth, the better. If they're holding the mic, make sure they grip it below the grill and don't cup or cover it.

Guitar amps/cabs: an SM57 or an Audix i5. These are dynamic mics with cardioid patterns. Quick note: an SM58 with its grill unscrewed is almost identical to an SM57, in case you need to exchange one for the other. A SM57 also works totally fine for vocals to be honest. When micing an amp, use a flashlight to locate where the drivers in the amp are located and aim the mic just an inch or two off center. The mic should be within an inch from the grill of the amp.

Acoustic guitar with no pickups: an SM57 or an Audix i5. Point the mic around 6 inches away to allow the guitarist movement and place it just above the sound hole at the base of the neck.

Bass guitar: Most bass amp heads have an xlr direct out that you should run an xlr cable directly from and into your mixer. If not, you can mic the bass cab just like a guitar amp with an SM57, an Audix i5, or kick drum mic. If they don't use an amp, you can connect their quarter-inch instrument cable to a DI (direct input) box and run that to your mixer through an xlr cable.

Snare and tom drums: an SM57, Audix i5, Audix D2, Audix D4, or Sennheiser e604. Mount the mic so the front is just over the rim, a few inches above it, and pointed right at the center of the drum. For snare drums, you can also or instead mic the bottom of the drum either at the rim or directly below and pointed at the snares to capture the sound of the snares more directly.

Kick drum: SM52 or Audix D6. If the kick drum has a hole cut out of the front head, point the mic directly into the hole so that the entire grill is just inside the kick drum but not so much that the entire mic is inside. If the kick drum head does not have a cut out, you can place the kick mic just off center and within an inch from the drum head. You could also remove the drum head if you have the time and energy and place the mic just off center and several inches into the kick drum.

Overheads for drums: Audix ADX-51 or MXL 991. Use one or two of these small diaphragm condenser mics to capture the drum set. Don't forget to turn on Phantom Power! They mostly pick up cymbals but also capture snare and toms very well. They don't pick up much low end or kick drum, so when I use overheads, I also mic the kick and sometimes nothing else. If you use one overhead, place it a couple feet above the kick drum, pointed down. If you use two, place one a foot or two above the high hats and another above the ride or cymbals. Use a cable to measure out the distance of each mic to the center of the snare drum and make sure they are the same distance. Also flip the polarity by using the phase flip toggle on your mixer on one of the mics if it sounds off and if your mixer has that option.

Choir or orchestra: MXL 990 or MXL 991. Preferably use large diaphragm mics but small diaphragm mics are fine if that's all you have. Don't forget to turn on Phantom Power! Place several feet away from the audio source and point directly at it. One neat trick, if you have two mics is to place one directly above the other and rotated 90 degrees from each other. It will make a forked, V shape pointed toward the audio source. Flip the polarity on one if your mixer has that option.

Anything with an xlr output. This can include bass amp heads, DJ mixers, or some instruments: Run an xlr cable from the output into your board.

Anything with a quarter-inch or headphone output. This can include instruments with pickups, pedal boards, keyboards, laptops, or electronic instruments like omnichords and pocket operators: Connect to a DI box and run the xlr output to your mixer.

DI (Direct Input) Boxes

DI boxes convert audio signal from a quarter-inch cable to an xlr cable, which makes it much easier to run to your mixer. I like the DB-01 because it's dependable and cheap. DI boxes will usually have two switches or buttons on it. One is the ground/lift button. If your signal has a buzz or hum to it, press this button and if it gets worse, switch it back. The second button or switch will be 0/-20/-40dB. If the signal is really loud, press the button to remove 20 or 40 dBs.

Analogue or Digital

All audio signals begin as analogue except for USB and Bluetooth. For digital mixing and processing, the analogue signal is converted into ones and zeros which represent a close approximation to the original signal. After being mixed and processed by the computer in your mixer, the signal is converted back to analogue so it can be understood by the amps and speakers. While digital mixing and processing does output an analogue signal that is pixelated and simplified, it is very difficult to tell the difference just by listening, even for experts. So, while it is true that digital mixing fundamentally alters the nature of the signal and sound, I wouldn't let that be the reason you choose analogue over digital. Digital is fine and makes sense in a lot of situations. The pursuit of purity isn't punk, it's puritanical and that sucks. I prefer analogue mixing over digital for other personal reasons like affordability, charm, warmth, nostalgia, and how the knobs and faders feel to me compared to a cold touch screen, motorized faders, and free spinning knobs.

Analogue mixing is the real thing, but signals do pick up noise and buzz over long distances and generally require more equipment and more time to set up. Analogue equipment is much more dependable and cheaper to purchase. They don't require software and firmware updates but can eventually begin to wear down physically. A well-taken care of analogue device can last decades though.

Cables

Digital audio signals are generally sent through CAT 6 or ethernet cables. Their range varies somewhat based on the equipment being used at both ends but is typically at least 300 feet.

Analogue audio cables have many more variations and applications. One important distinction is balanced vs. unbalanced cables.

Unbalanced cables have 2 wires within them. One to send the signal, and the other to return to the beginning of the circuit. These two cables are often called, “hot” and “cold” or “positive” and “negative” and they are often colored white and black under the insulation or sleeve of the cable. Unbalanced signals quickly loose quality and pick up a buzz and other noise. This can be noticeable to me at 25 feet and gets worse the longer the cable.

Balanced cables have 3 wires within them. For these cables, the “hot” signal is split and the polarity is reversed for one of the signals. When the two “hot” signals are recombined, noise and buzz picked up along the way is nearly eliminated and can be good for hundreds of feet. I don’t understand this process completely so don’t ask me more about it and I won’t have to make up something using words I barely understand. The third wire is for the return or “cold” signal.

How to tell these cables apart: All xlr cables are balanced. If you look at the ends of xlrs, you’ll see 3 pins. That’s the 3 wires within the cable. For quarter-inch cables, look to see how many black bands are on the tip. If there is one band, it’s unbalanced because the black band separates terminals for the two wires in it. You’ll use unbalanced quarter-inch cables for instrument cables because they are usually under 15 feet to get to an amp or DI and won’t suffer much signal degradation. If there are two black bands, that’s a balanced cable because the black bands separate three terminals for wires.

There are also stereo and mono cables.

Stereo cables carry two different audio signals in the same cable, normally for a left and right speaker. Each has its own signal independent from the other. Stereo signals are typically carried only by eighth-inch cables and have 3 wires in the cable. Two wires are “hot” and carry the two independent signals and the third wire returns the combined “cold” signal. Both signals are unbalanced. When stereo is sent out, two xlr cables are used to go out from your mixer to your left and right speakers.

Mono cables carry only one audio signal and have two or three wires inside of them. One or two are “hot” and the remaining wire is “cold.”

Types of cables:

You can tell what a cable is and how to refer to them based off of what the end looks like.

An **xlr** cable has a shell shaped end with three pins inside the sleeve. You could look up what xlr stands for but no one will ever call it that and you could spend your time better by touching grass or watching clouds or something. These cables are great for carrying unpowered signal to your board and in-between devices and processors.

A **quarter-inch** cable is more pointy on both ends and is a quarter-inch thick. They can be just as good as xlr cables if balanced and can be used for many of the same things.

An **eighth-inch or 3.5mm** looks like a smaller quarter-inch cable and is sometimes referred to as an aux cord or headphone jack. It can be either stereo or mono and functions similarly to a quarter-inch cable.

RCA cables have a short tip and sleeve. They are mono only. They are often used for Audio-Visual or HiFi home equipment.

Converter cables have different ends and are often referred by one end to the other such as an eighth-inch to RCA or an eighth-inch to quarter. These specific cables separate a stereo signal into two mono cables so they can be plugged into a mixer from an audio source or music player. There are also converters that attach to the ends of cables that change their connection type.

An **insert** cable is a cable that is only used on the insert jacks on an analogue mixer. This cable splits a balanced quarter-inch to two unbalanced quarter-inches so that it can be sent to an external processor like a gate, compressor, and/or limiter.

A **speakon** cable is used to carry powered signal from an amp to a passive speaker.

One of the first skills you learn should be how to wrap cables using the over-under technique, which can be found on youtube.

Mixers and Processors

I talked a little about this earlier, but there are analogue and digital mixers. I recommend analogue over digital especially for small venues or shows mainly for their durability, affordability, and reliability. However, I'll somewhat cover both because it's impossible not to come across digital mixers especially as you begin running sound at mid-size venues.

There are a lot of good analogue mixers out there and it's fine to buy something used as long as all the knobs and faders are there and there's no obvious damage. I personally really like the Mackie sr24-4 vlz or if you're looking for a smaller, all-in-one style, the Behringer Xenyx X1832 is a good choice.

There are also some analogue, powered mixers that have amps built into them, which allows you to send signal straight from the mixer to the passive speaker.

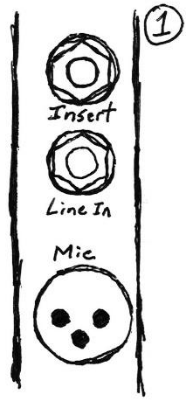
Digital mixers look very different from each other based on brand, make, and model. I like the Midas M32 or the Behringer X32, which are made by the same company and are actually very similar to each other. The M32 is becoming the standard digital mixer you'll see out there. I also like the Soundcraft Si Expression, but they have really fallen out of favor with audio engineers and technicians and you won't see them nearly as much.

Digital mixers have compressors, gates, limiter, graphic equalizers, and other processors built in that are usually separate components when mixing analogue.

When looking at a mixer, you'll see numbered columns with lots of knobs and faders. Each one of these is called a channel and is used for a specific input. The good news is what each knob and setting does for one channel, it does the exact same for the rest of the channels.

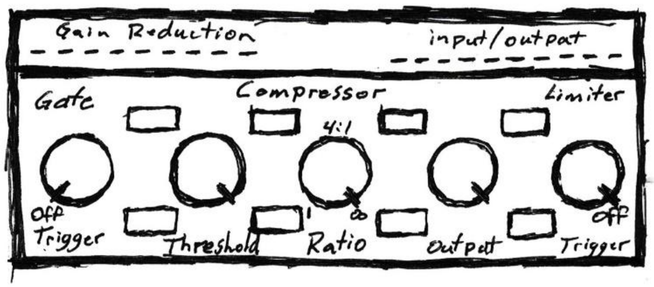
When describing the position of which knobs are turned, the time on a clock is referenced. 12:00 or midnight is straight up, where there is no adjustment is being made to the EQ. 3:00 is straight to the right for EQ, where this is a boost or addition of usually 7.5 dB, and 9:00 is straight to the left for EQ where there is usually a cut or reduction of 7.5dB.

Let's go over some typical sections of analogue mixers.



1. **Patchbay**: Here's where you'll connect your cables. At the top of each channel, you'll have your inputs. It's often three jacks. One for xlr input, one for quarter-inch input, and one for an insert cable. The insert jack allows you to connect external devices and processors such as gates, compressors, and limiters. Use either the xlr or quarter-inch input. Never both. The output jacks vary somewhat between mixers, but they are commonly labeled "out" or "send."
2. **Gain** knob: Also known as trim. This knob controls the strength of the input signal for the channel and for any processors connected by the insert jack. This knob should be rarely adjusted and should only be when the strength of the input signal changes. This is very uncommon but can happen when, for instance, a different vocalist comes to the mic for a song or if the guitarist adjusts the volume on their amp. This knob is critical for building your "gain structure" which is important for how you mix.
3. **Insert Processors:** Technically you can connect any processor to a channel on a mixer, but generally you'd only want to connect gates, compressors, and/or limiters. You don't have to insert any processors at all and the signal will flow through to the next points. I usually don't use any processors for small venues or shows. For med-size venues, I'd recommend the Behringer MDX 4600 because it has 4 channels and includes three processors: gate, compression, and limiting in one rack unit.

Gate: Only allows signal to pass through to the mixer once the signal strength crosses a settable threshold, which is set by the "threshold" knob. This helps reduce noticeable bleed from other instruments on stage.



Compression: This begins to reduce the gain or strength of a signal once it crosses a settable threshold. This makes loud parts sound more similar to quieter parts and is used to make inputs of various strengths on an input sound more similar. There are a few settings controlled by knobs for this processor.

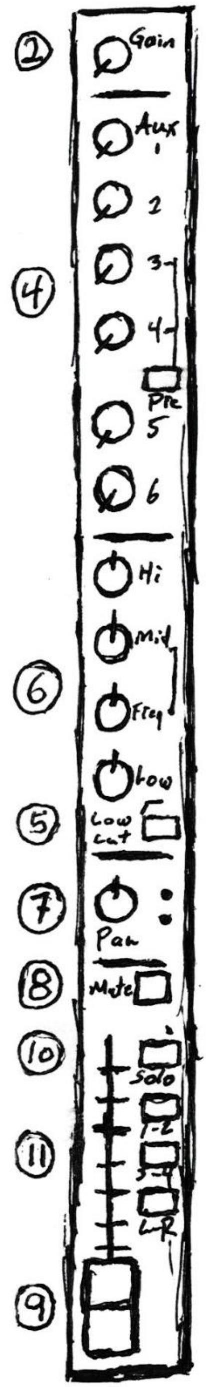
- **Threshold**: This sets the signal strength at which gain begins to be reduced and output signal is lowered.
- **Ratio**: This is the ratio at which gain is reduced as the input signal rises. For instance, if a ratio is at 4:1, an incoming signal 4dB above the threshold has an output signal only 1dB above the threshold, with a gain reduction of 3dB.
- **Attack**: This knob controls how quickly the compressor kicks in once the signal crosses the threshold. I generally like to use a quick attack setting.
- **Release**: This knob controls for how long the compressor stays active once the signal crosses back below the threshold. I generally like to use a quick release setting. For the Behringer MDX 4600, the attack and release settings are automated and these knobs are not present.
- **Output**: This knob allows you to adjust the gain and signal strength after compression. This allows you to boost the

gain back up and should be around the typical gain reduction for your compressor.

- **Other**: There may be some more buttons or toggles on your compressor such as "soft knee" and "couple." I recommend reading the manual to figure out what all of these these may be used for, but I'll explain these two since I brought them up.
    - **Soft Knee**: This button begins to compress the signal slightly leading up to the threshold setting to make the gain reduction a little less noticeable or harsh.
    - **Couple**: This button copies the settings of the compression channel onto the following channel. This is used for stereo signals so they are always the same.

Limiter: A limiter is actually a compressor with such a high ratio so that a signal strength cannot pass above the threshold. This is sometimes denoted as ∞:1. This is useful for ensuring that a signal strength never gets too high for your speakers to handle without getting damaged. Be careful because analogue limiters especially are not perfect and very high increases in input signals can still cause some increase in the output signal and this can cause distortion and clipping.

4. **Aux**: These knobs let you build a separate mix for outputs separate from your main speakers. These can be used for a monitor, recording device, livestreaming device, effects processor, and more! Using this knob controls the strength of the signal going to your aux outputs. There are “pre” and “post” aux outputs. “Pre” aux signals are “pre-fader” and are not effected by the position of the fader, while “post” aux signals are “post-fader” which means the strength of the signal is adjusted by moving the fader.
5. **Low Cut or High Pass Filter**: This button removes low-end frequencies. Typically 80Hz and under, but this number should be labeled next to the button. This is useful for inputs that do not carry low-end sound such as vocals and guitars.
6. **EQ**: These knobs adjust the EQ or equalizer. This adjusts the strength of the frequencies of each channel. People hear sound from 20Hz to 20,000Hz (20kHz) and the EQ gives you the ability to reduce or increase the signal strength of select frequencies. You will typically have a “high,” “mid,” and “low” knob which adjusts the frequencies as labeled. The “high” knob usually boosts or cuts frequencies of 12k and higher. The “mid” knob is usually paired with a “sweeper” knob which allows you to select the



frequency you'd like to boost or cut. The "low" knob boosts or cuts frequencies of around 80Hz and lower.

7. **Pan**: This knob allows you to send more or less signal to the left or right speakers. This can be used for stereo signals with the right input panned all the way to the right and the left input all the way to the left.
8. **Mute**: This button mutes the channel entirely when engaged. Sometimes it's an "on" button instead.
9. **Fader**: This adds or removes signal strength at the very last stage of the channel. Look closely at the numbers along side of the fader and find the 0. When the fader is at this mark, the strength of the signal is not affected by the fader. When it is above this mark, the fader is adding that number of decibels to that channel's signal. When the fader is below the 0, it is removing that number of decibels from the channel's signal. The very bottom number is an infinity symbol (∞), meaning when the fader is all the way down, all signal is theoretically being removed before leaving the channel.
10. **Solo**: This button sends the signal to the headphone jack so that the channel can be isolated and listened to through the headphones.
11. **Sends**: These buttons select to where the signal from the channel is being sent. They can be sent to a group fader to directly to the main speakers. One of these buttons must be selected if present for the signal to be sent from the channel.
12. **Aux Send Masters:** These knobs control the overall signal strength sent to the aux outputs. If they're all the way down, no signal will be sent out.
13. **Aux Return:** These knobs control the strength of the signal that's meant to be the return signal from processor devices such as effect processors back into the mixer. I don't actually use these and prefer to return signal into a channel with a fader. This gives me more control and allows me to eq the signal. Just be sure not to

send this channel to the aux it was sent from or you'll create a weird feedback loop.

14. **Tape in:** There are two RCA jacks on your mixer labeled, "Tape In" and this knob controls the strength of the signal from these jacks. There may also be a button that says, "send to main." Press it to send this signal to the main speakers.
15. **Group Faders:** If you selected a group send for any of your channels, the signal was sent here. Turn up these faders to hear the channels. These can be used if you want to have a fader for all of your drum channels, or all of your guitar channels, or all of your vocal channels for instance.
16. **Main or Master Fader:** This fader is the final signal strength adjustment for your main speakers on the mixer and can be either one fader to control both left and right or two separate faders to control left and right separately.
17. **Effect Processors:** Abbreviated Fx. This processor adds an effect to a signal. These effects can include reverb, delay, chorus, flanger, amongst many others. When it's a device separate from the mixer, you can send signal to it from a post-fader aux send and into the input of the effects processor. Run a cable from the output of the effects processor and into a channel on the board or into the aux return. This will return a parallel "wet" signal that has the effect incorporated into the signals from the channels being sent to the processor through the aux sends.
18. **Graphic Equalizer:** Abbreviated GEQ. This device allows you boost or cut signals at specific frequencies, which are labeled and are not able to be selected. This can be used for feedback suppression and/or pitch correction. GEQs are described by how many "bands" they have. They typically come in 7, 15, or 31 band GEQs. The higher the band, the greater the number of frequencies you can adjust and how specific the adjustment is to that frequency. You typically connect GEQs into your signal chain after

being sent from your mixer and just before making it to your speakers, both your mains and monitors.

That's it for what analogue mixers and processors can do for now. There's definitely more to it and more ways to connect them, but this is a good starting point.

Digital Mixers have all of these same functions and more. Where these functions are in the software and how to reach them depend greatly on the make and model of the mixer. Search and pull up a youtube video for the specific mixer for a tutorial. There's a lot of information and demonstrations out there.

Typically, digital mixers have a "select" button at the top of each channel which allows you to adjust the processor settings for each channel, be careful to have the correct channel selected when adjusting these settings. Faders are also motorized and knobs are free spinning, meaning that as you select through input and output pages, the faders' and knobs' position is saved as you toggle through these pages.

Speakers

There are two important aspects to speakers. Powered and unpowered speakers. Powered speakers require unpowered signal, and unpowered speakers require powered signal to work. Signal is powered by an amp, and a powered speaker has the amp built into it.

Powered speakers require AC power, usually delivered by a universal power cable (UPC,) which is the same cable that supplies power to the mixer and processors and also equipment like desktop computers. Signal is usually sent over an xlr cable but can be sent by other cables if those input jacks are on the speaker. Powered speakers should be stored in a climate-controlled room. They are also becoming more standard over unpowered speakers.

When setting up your powered speakers, put the gain knob at "zero." This will set up your speaker to clip at the same signal strength as your mixer. This helps you know when the signal clips the speaker by monitoring the mixer. If you have EQ knobs on your speaker, set them to "midnight" so that they are making no adjustments.

Unpowered speakers do not require AC power and have usually have their signal sent to them over a speakon cable from an external amp. When pairing an amp to a speaker, make sure that the power (measured in Watts) delivered by the amp is not more than the speaker is rated for. Check the specs and the manual of the amp and the speaker for this information. Unpowered speakers have the benefit of not needing to be stored in a climate-controlled environment and generally weigh less.

Tops and Subs. Tops are generally two-way speakers, meaning they have two drivers or cones built in. One delivers mid-range sound and the other delivers high-end sound. Tops should be mounted at least head-level of your audience and be generally pointed towards them. Subs deliver low-end sound, usually under 200Hz. Subs are usually bigger and heavier than tops. And they don't deliver a wide range of sound. So, sometimes when I know I won't be sending signal 200Hz or lower through my mixer, I don't bother bringing subs or setting them up.

Selecting a good speaker can be pretty difficult, but you should always look up the specs! If you find a speaker that looks good online or at a store, you should look up the manufacturer spec sheets. To me, the most important figures are the peak SPL, which gives you best information on loudness, the frequency range, which tells you how much of the frequency range it covers, and the dimensions and weight to know how difficult it will be to store and move.

For small venue shows, I recommend the EV-ZLX powered speakers. I also recommend the Gemini AS-15P, which sound suspiciously decent and loud for the price. These were my first speakers and have somehow lasted me nearly 10 years at this point.

For mid and large shows, I recommend the JBL SRX series powered array speakers and subs. Their frequency range could be better but it's good for the price.

Powering Off and On Sequence

When you get to a venue or once you set everything up, you should power on your mixer before powering on your amps or speakers. At the end of the show, you should power off your amps or speakers before powering off your mixer. In other words, you shouldn't turn on or off your mixer while your amps or speakers are on. If you do, you will hear a loud pop that could damage or blow your speakers. Honestly, this is not as much as a problem as it used to be in the 90s with older equipment, but doing so is still considered irresponsible or an indicator that you're under-experienced. If for some reason you have to or accidentally do it, it's probably fine but people might judge you but ehh what can you do?

Testing the Speakers and Troubleshooting

Connect your phone or music player to your mixer and send a signal out. If you have everything powered on and connected correctly, you'll hear the music through the speakers. Also test out your monitors and processors by sending signal through and to them.

If you don't hear music through each of your speakers, let's go over connections and signal flow.

Your input should be going into your mixer. The main LED indicators should be bouncing and moving. If they are not, the signal is not being sent out of the mixer. If this is the case, bring up the fader on the channel the music player is connected to towards "zero" and make sure the main left and right button send is selected. Make sure the main fader is at "zero" also. Bring up the gain on the channel until you can see the LEDs light up and move.

Check that the power is turned on at the amps or powered speakers. Double check your signal connections too.

If none of this is working, make sure all your connections are correct and none are loose. The signal should be going from your music player into the mixer. The channel may use an insert. Make sure the inserted processors are turned on and gain knobs are turned up. The outputs from the mixer may be going to a GEQ and/or a compressor or other processor. Check these connections and make sure power is turned on for them. The output signal from these devices runs to an amp and/or speaker.

If everything looks good, keep checking until you start to hear something. There's a lot of things going on here so it's ok if it takes you a while to figure it all out.

Sound Check

Let's run through how to sound check different types of inputs and instruments and some typical settings for your average band on stage. Just prior to sound check, make sure your gain knobs and aux sends are turned all the way down. Channels using mics can have their faders brought to zero, but channels with line inputs should have their faders kept down as well. Thresholds on gain, compression, and limiters should not be engaged, and output knobs should not be boosting or cutting gain.

Talkback: This mic is used by you at the mixer to communicate with the band on stage and not the audience. Take a vocal mic and connect it to the board, usually one of the last channels on the board. Bring the fader up to zero and starting saying, “check” repeatedly as you begin to bring up the gain. When your voice reaches a reasonable loudness in the room, bring the fader all the way down to take it out of the main mix. Now bring up the aux sends to where the band on stage can hear you through the monitors. Ask them if they can hear you and raise the aux send in each monitor until each member signals that they can hear you. If you’re close enough to the band for them to hear you without the talkback, you don’t have to set it up.

Kick: Bring the fader for this channel up to zero. Say to the drummer, “give me some kick.” They should begin kicking the drum steadily. Make sure the desired “send” button is pressed.

- Gain: Bring up the gain on that channel until the loudness reaches a good level in the room.
- Low Cut: I don’t use a low cut for Kick or other inputs that have low-end in their signal.
- EQ: Make a decent cut around 200Hz. If it sounds too “boomey” make a slight cut on the lows. You can also cut the highs but you may not notice a difference in sound.
- Aux Sends: Ask the band over the talkback, “who needs it in their monitor?” Those band members who want it should raise their hands. Start raising the signal slowly while the drummer is still kicking and tell the band members to “put your hand down once you’re good in the monitor.”
- Gate: Bring the threshold of the gate up until you start to cut out the signal, then bring it down slightly so that the signal just makes it over the threshold. You should see the gate close on the LED when the kick is not producing audio.
- Compression: Set the ratio around 4:1. Bring the threshold down until you have a gain reduction of 3-6dB. Boost the output gain by 3-6dB.

- Limiter: Make sure this threshold is set well above the strength of the signal so it only engages if the mic is dropped or hit with a drumstick.

Snare: Bring the fader for this channel up to zero and say, "alright, snare." The drummer will start steadily hitting the snare drum. The rest of the process is just like the usual with kick except for EQ, where you could just start with a low cut. For the aux sends, you don't have to and can skip the process of sending it to monitors.

High Tom: Bring the fader up to zero and say, "alright, tom 1." Checking this channel is just like snare.

Low Tom: Bring the fader up to zero and say, "alright, floor tom." Checking this channel is just like the high tom except the EQ shouldn't need a low cut.

Overheads: Bring the fader up to zero and say, "alright, full kit." Checking this channel is just like the high tom and snare. Make sure again that phantom power is turned on. Just make sure the whole kit fills in as you raise the gain.

Bass: This is likely a line input so keep your fader all the way down and say, "alright, bass."

- Gain: Begin bringing up the fader towards zero. Once you reach zero, if you still need more gain, bring up the gain knob until you reach the desired loudness through the mains.
- EQ: I normally don't make any eq adjustments to bass. If a part of the pitch sounds too loud or too quiet, you can cut or boost that area of the pitch.
- Aux Sends: Ask the band over the talkback, "who needs it in their monitor?" Those band members who want it should raise their hands. Start raising the signal slowly while the bassist is playing and tell the band members to "put your hand down once you're good in the monitor."

- Gate: I only use this for drums and not other instruments unless there is a noticeable or distracting buzz coming from their amps. In this case, I set the gate threshold just above the signal generated by the buzz on these inputs.
- Compression and Limiter: follow the same steps as previous channels. I use a ratio of 3 or 4:1.

Guitar: If this is a microphone input, you can bring the fader up to zero and say, "alright, guitar." Follow the normal steps from here like you did with bass and kick. I usually do a low cut but don't mess with the EQ much if at all. I often don't use compression, but if I do, I use a small ratio of 2 or 3:1.

Keyboard: This is a line input so start with your fader down and slowly bring up as your keyboardist starts playing. You can check this in like you did with bass and I don't make any eq adjustments or use gate, compression, or limiter.

Vocals: These inputs are where you'll spend most of your time checking. Vocals are what your audience tends to notice the most, even across all genres of music. These inputs are generally the strongest signals in your mix and will be the most prone to feedback. If you encounter feedback, lower the gain slightly for now. I'll go over methods of feedback suppression in the next section. You can check vocals just like other inputs, but let's go over it again from the top.

First, bring the fader for this channel up to zero. Say to the vocalist, "check the mic for me." They should begin checking the mic or singing or rambling. Make sure the desired "send" button is pressed.

- Gain: Bring up the gain on that channel until the loudness reaches a good level in the room.
- Low Cut: Engage the low cut and if you can select the frequency, set it around 200-300Hz.
- EQ: Make a decent cut around 1kHz. This cut makes the vocals sound a little more clear and helps decrease the potential for

feedback. Make additional cuts at frequencies where feedback occurs.

- Aux Sends: Ask the band over the talkback, “who needs it in their monitor?” Those band members who want it should raise their hands. Start raising the signal slowly while the vocalist is singing and tell the band members to “put your hand down once you’re good in the monitor.”
- Gate: Don’t use it for vocals. Vocalists tend to have a more dynamic level, meaning they sometimes dip to a softer loudness than you’d expect. If they dip below the threshold of the gate, their signal will be dropped and it would seem as if the mic is “cutting out” and people will notice.
- Compression: Set the ratio around 2 or 3:1. Bring the threshold down until you have a gain reduction of 3-6dB. Boost the output gain by 3-6dB.
- Limiter: Make sure this threshold is set well above the strength of the signal so it only engages if the mic is dropped.

Feedback Suppression

Feedback occurs when a microphone picks up a specific frequency or set of frequencies from a speaker, either a main or a monitor and sends this frequency back to the speaker, which is again amplified and a loop is created. Some frequencies will always be more prone to feedback than others depending on unpredictable aspects such as the room shape and size, the number of people in the audience, the type and number of microphones being used, and their positions on stage.

Suppressing feedback is essential for providing a strong gain and should happen during mic check but can also be done as needed. During mic check, the person checking the mic should simulate where the vocalists could be on stage and at a volume similar to the performance.

There are a few methods which can be used, some of which should only be used during soundcheck. All of these methods involve altering the EQ, which we'll go over now.

- The ole boost and cut: Use this to locate and suppress the frequencies on the EQ that are feeding back. On the mid-range EQ knobs, make a slight boost of 3-5dB and then sweep across the frequency range until you find an area that feeds back. At this frequency, make a slight cut of 3-5dB. This will theoretically allow you to boost the gain of the channel by 3-5dB before it feeds back again at that frequency.
- Listening and cutting: You can train your ear to recognize frequencies just by listening! It's easier for some people more than others, but you can practice by searching for a website or app that specializes in this. Try searching for "audio frequency trainer." On digital mixers, such as the M32, the EQ graph is displayed along with a frequency analyzer which will show you on the display which frequency is feeding back. You can then make a cut there.
- GEQ: Before making changes on your GEQ, I would start with adjusting the EQ on individual channels. This is because GEQ is usually set up to adjust the EQ of the mix to the mains or monitor and not to the input channel, meaning it will change the EQ of all inputs for that output. Your adjustments on your GEQ should be minimal for this reason. You don't want, for instance, to set EQ for a vocal mic that will also adjust the EQ on the keyboard or bass.
  To find feedbacking frequencies on a GEQ, boost each frequency and listen for feedback. Make a cut on the frequencies that feedback that correspond to how quickly and loudly feedback occurs. When a frequency feeds back a lot, make a large cut, but if a frequency only slightly feeds back, make a smaller cut.

- Feedback destroyer: You can get a processor like the Behringer FBQ2496 Feedback Destroyer Pro, which will automatically remove frequencies that it detects feeding back. These aren't great but they're fine to use if you feel like you need them. They can't distinguish between desired feedback from guitars for instance and I believe they're unnecessary once you understand how to suppress feedback yourself.
- When you've done all you can: If you made a lot of EQ adjustments and find yourself boosting the gain to get back some of the signal strength taken out from adjusting the EQ, you'll start to hear multiple frequencies feeding back at once. At this point, you've done everything you can and the mic's gain just isn't able to be boosted any further. Attempting to suppress feedback by cutting frequencies more will be like chasing your own tail at this point.

Note: Adjusting EQ also changes the pitch of your input. You'll notice that making low-end cuts will make your input sound higher and brighter. If you've adjusted the EQ to the point that it's made the input sound out of character, try dialing some of those frequencies back in and bring the gain down to avoid feedback if you have to.

I usually only make cuts in the EQ section but if boosting the EQ settings for some frequencies helps adjust the sound quality for the better, go ahead! I do this occasionally myself.

Also take as long as you have to sound check. You can save a lot of embarrassment, stress, and in-show adjustments by taking a 10, 20, or even 30 minute sound check if you have to and have the time.

Mixing

When people talk about a "mix," they mostly are talking about the strengths of inputs in comparison to each other. They can also be talking about how compressed the overall mix is and what the general EQ tone is like.

To me, a "good" mix is one that works with the genre of music being performed. The audience's expectation of what the mix will sound like depends on how this genre of music is represented on their recorded tracks. You can also look up the band before you mix for them and try to mimic the mixes of their recordings. The bands and their fans are usually very happy when you do this.

- Pop: This mix should be very vocal centered. The vocals should be very loud and clear so you can hear and understand each word. The instruments should be very background in level and do not need to be heard clearly and the only overall vibe of the instruments' sound is needed. Generous use of compression is preferred and the sound should be very bright but with plenty of low-end if supplied by their instrumentation. You can put some reverb and delay on the vocals. Honestly, I put more reverb on a vocalist that sounds bad and less on a well-trained or talented vocalist.
- Hip-Hop: This mix is also very vocal centered, but more balanced with the instrumentation. Make sure the vocals are again very clear and you can understand each word. Make the drums very present and that the low-end is boosted well. You can use a good bit of compression. No effects on vocals.
- Indie: This is a much more balanced mix but still make sure the vocals are nice and present. Make sure each instrument can be heard well but not so much that any standout too much. Don't compress so much and have some nice, warm EQ settings. You can put some reverb on the vocals.
- Punk/Harcore-punk: This is a very balanced mix where each instrument is loud and present. The vocals should be present enough to be heard but not so much where each word has to be understood. Try to make each instrument stand out as

much as possible but the mix will often end up being somewhat muddy and that's ok. Don't be surprised if they are dismissive of the use of monitors and prioritize speed of set up over quality of sound. Don't use much compression if any at all and use warm EQ settings. Don't use any vocal effects unless specifically requested.

- Commercial Hardcore: Mix it like pop, except no vocal effects.
- Metal: Mix it like punk but try to have more present vocals and cleaner instrumentation.

From time to time, you may receive criticism on your mixer's settings or how they look during a show. If something sounds bad, that's a different story, but do whatever you have to on the mixer to make a good sound. That's the most important thing and you don't need to prove yourself or take anyone's bad advice.

Starting Off on Running Sound

I was really lucky to find someone who was willing to let me shadow them and explain this all to me while doing it themselves. So, I would recommend finding and talking to the sound person at one of your favorite music venues or scenes and asking them if you could shadow them. Ask questions but not too many when they look stressed and try not to get in the way, talk too much, or flex your knowledge.

I'll apologize now for the general volatile and impatient nature of most people who run sound. Honestly, the more difficult a sound person is to work with, the less they know, the worse they are at their job, and the more misinformation they swear by. It's uncommon to find someone who is patient and regulates their emotions, but if you do, ask them if you can shadow them during gigs. You might not get paid, but you don't get paid to go to school for this either.

If you get lucky, and when they think you're ready, they may ask you to stand in for them for an easy show when they can't make it. In the meantime, get yourself some audio gear (used or cheap is fine!), just enough for a small venue or show and start running sound! It's ok if you mess up at first or if it doesn't sound great. It's not easy and you're still learning.

You can also start to look for a job like audio assistant or technician. Learn how to wrap cables using the over-under technique, which can be found on youtube. Knowing how to wrap cables is honestly a good enough start for a beginner level.

Final Words

This is about everything I think you'd need to know to get started mixing! A lot of things come by experience and if you have any questions along the way, you can reach me at dwgoldsmith3@gmail.com. Be patient with yourself while you're learning because this is a really complicated and technical process! I didn't even set out for this zine to be this long honestly.

There are still things you can learn about that I didn't touch on such as waveforms, history, a lot about digital mixers, other equipment, and festival-style shows. So, I hope you continue learning!

Glossary

AC – Alternating Current. This supply of electricity comes out of electrical outlets and actually increases and decreases at a consistent rate of 50 to 60 times each second in the states. This is what you'll use to supply all your equipment with power.

Bleed – This is the background noise of other instruments or audio sources being picked up in a mic that is designated for one specific source. It's like how you can kind of hear the drums through your vocal mic. With live sound, it's impossible to eliminate completely but you can minimize it through cleaver microphone placement and/or using a gate. Generally, you have to accept some bleed and work with it.

Bright – Used to describe sound. The EQ has boosted mid-highs and highs.

Clip – The signal has reached the maximum strength that the mixer, amps, and processors can handle. This carries a risk of damaging your equipment, especially your risk of "blowing" or destroying speakers. However, this risk has dramatically decreased with modern audio equipment and built in limiters. The sound of the signal will also be distorted, which can be a dead giveaway that your signal is clipping.

Dark – Used to describe sound. The EQ is mostly low-mid and lows.

Decibel (dB)– This how signal strength is measured. Inside of electrical equipment, it corresponds to voltage and depends on the strength. In the air and on your eardrums, it corresponds to air pressure and how much is being exerted.

Distortion – The sound of a signal being clipped. When this is intentional, it is done by an effects processor. A distorted signal is gained up so high that it's being "crushed" so that your signal no longer varies in strength and otherwise weak signals are as strong as the strongest signals.

Dry – Used to describe sound. An unprocessed signal with no effects.

Gain Structure – How your gain knobs are set based on how strong the input signals are. A good gain structure is marked by how close to "zero" your faders are set. Within 10dBs is reasonable but do what you have to get a good mix. This will also help have reasonable settings on your insert processors.

Hot – A strong signal or gain setting on a mic or other signal.

Impedance – This is electrical resistance for alternating current. It corresponds inversely to how much electrical current is present in a circuit.

Loudness – This is how our ears interpret signal strength. Loudness increases with signal strength but not exactly on a one to one scale across all frequencies equally. For instance, we tend to interpret high frequencies louder when overall signal strength is low. This is why whispering sounds like a bunch of “psss” and “thhh” sounds.

Muddy – The inputs sound like they’re overlapping and can’t be easily distinguished from each other. This could be caused by bleed or instruments with overlapping frequency ranges.

Muffled – Used to describe sound. The highs and mid-highs are too low or the mid-lows are too dominate.

Ohm: How electrical resistance is measured. Lower ohm speakers process signal from amps more efficiently and are louder than higher ohm speakers.

Parallel – This has to do with how audio processors are connected to each other. In parallel, there is a “dry” and “wet” signal or an affected and unaffected channel, where you can mix or blend the two together to taste.

Phantom Power – This is a somewhat misleading term as it’s neither spooky nor power. It is a supply of 48 volts to the microphone from the xlr jack in your mixer and is sometimes labeled as “+48.” They should only be used for condenser mics and are necessary for them to work and pick up sound. When phantom power is sent to a dynamic mic, it does not affect it at all, but other devices especially music players that use the xlr jack can be damaged if phantom power is used on them. It can be activated for all channels on your mixer by a switch or button on the mixer that can be located near the power switch or main fader. Some mixers, especially digital mixers, have phantom power toggles for each individual channel. This toggle will be in the settings of each channel or just below the gain knob.

Phase Flip – This reverses the polarity of a signal and is denoted (Φ). This has a lot to do with waveforms, but this action helps the waves of the signal not cancel each other out and sound more desirable.

Series - This has to do with how audio processors are connected to each other. In series, a signal is processed before entering your mixer and the amount of processing can't be adjusted or blended on the mixer.

Shadow – Like an assistant, but you are more there to observe, learn, and not get in the way, like a shadow!

Signal Strength – This describes how present a signal is in an electronic device or in the air by measuring it in decibels either by voltage or air pressure.

Voltage – This is sometimes referred to as electrical potential or difference. It corresponds to a higher signal strength, but the amount of electrical current is also inversely affected by the resistance and impedance in the current.

Warm – Used to describe sound. The mids or high-mids are somewhat boosted. It's normally a good thing and is used to describe analogue equipment and recordings.

Watt – This is a unit of electrical power, or rate of which energy is being used.

Wet – Used to describe sound. Processed signal typically with heavy effects.