Audio Basics

A sound reinforcement system is the combination of microphones, signal processors, amplifiers, and loudspeakers; its purpose is to amplify live or pre-recorded sounds in order to distribute them to a larger or more distant audience. In most situations, a sound reinforcement system is also used to enhance the sound of the sources on the stage, as opposed to simply amplifying the sources unaltered.

Signal Flow is a very important thing to understand before mixing for your first time. A simplified common signal flow found in the Performance Hall would be from a microphone ➝ XLR cable ➝ stage input ➝ patch bay ➝ Venue Console ➝ digital processor ➝ amplifiers ➝ main speakers. A lot of these steps have many more steps of internal signal flow, but this the general order of processes.

Basic Audio Definitions

<u>A/D converter:</u> Analog-to-digital converter, a device that transforms incoming analog signals into digital form.
<u>AFL:</u> An acronym for After Fade Listen, another way of saying post-fader solo.
<u>Aux Send:</u> A mixer bus output designed to send a signal to an auxiliary processor or monitor system.
<u>Aux Return:</u> A mixer input with limited control capabilities, intended for bringing the output of an auxiliary processor or other line-level source into the main mix bus.
<u>Balanced Input:</u> An input consists of two leads, neither of which is common to the circuit ground. This is a “differential pair”, where the signal consists of the difference in voltage between the two leads. Balanced input circuits can offer excellent rejection of common-mode noise induced into the line.
<u>Balanced Output:</u> In a classic balanced audio circuit, the output is carried on two leads (high/+ and low/-) which are isolated from the circuit ground by exactly the same impedance. A symmetrical balanced output carries the same signal at exactly the same level but of opposite polarity with respect to ground. A special case of a balanced output carries the signal on only one lead, with the other lead being at zero voltage with respect to ground, but at the same impedance as the signal-carrying lead.
<u>Bandwidth:</u> The band of frequencies that pass through a device with a loss of less than 3dB, expressed in Hertz or in musical octaves.
<u>Cardioid:</u> Heart-shaped. In sound work, cardioid refers to the shape of the sensitivity vs. direction plot for a particular style of directional microphone. A cardioid mic rejects sound arriving from the rear.
<u>Channel:</u> A functional path in an audio circuit: an input channel, an output channel, a recording channel, the left channel and so on.
<u>Channel strip:</u> The physical realization of an audio channel on the front panel of a mixer.
<u>Clipping:</u> A form of severe audio distortion that results from peaks of the audio signal attempting to rise above the capabilities of the amplifier circuit. To avoid clipping, reduce the system gain in or before the gain stage in which the clipping occurs.
<u>Compressor:</u> This is a dynamic processor used to smooth out any large transient peaks in an audio signal that might otherwise overload your system or cause distortion. The amplitude threshold and other parameters such as attack time, release time, and ratio are adjustable.
<u>Condenser:</u> Condenser microphones require electrical power to run internal amplifiers and maintain an electrical charge on the capacitor. They are typically powered by internal batteries or “phantom power” supplied by an external source, such as a mixing console.
<u>Cueing:</u> In broadcast, stage and post-production work, to “cue up” a sound source (a record, a sound effect on a CD, a song on a tape) means to get it ready for playback by making sure you are in the right position on the “cue,” making sure the level and EQ are all set properly. This requires a special monitoring circuit that only the mixing engineer hears. It does not go out on the air or to the main mixing buses.
<u>D/A converter:</u> Digital-to-analog converter, a device that transforms incoming digital signal into analog form.

Decibel (dB): A unit used to measure the intensity of a sound or the power level of an electrical signal by comparing it with a given level on a logarithmic scale.
Delay: In sound work, delay usually refers to an electronic circuit or effects unit whose purpose it is to delay the audio signal for some short period of time. Delay can refer to one short repeat, a series of repeats or the complex interactions of delay used in chorusing or reverb. When delayed signals are mixed back with the original sound, a great number of audio effects can be generated, including phasing and flanging, doubling, Haas precedence-effect panning, slap or slapback, echo, regenerative echo, chorusing and hall-like reverberation. Signal time delay is central to many audio effects units.
Dry: Usually means without reverberation, or without some other applied effect like delay or chorus.
Dynamic microphone: The class of microphones that generate electrical signals by the movement of a coil in a magnetic field. Dynamic microphones are rugged, relatively inexpensive, capable of very good performance and do not require external power.
Equalization (EQ): Refers to purposefully changing the frequency response of a circuit, sometimes to correct for previous unequal response (hence the term, equalization), and more often to boost or cut the level at certain frequencies for sound enhancement, to remove extraneous sounds, or to create completely new and different sounds.
Frequency: The number of times an event repeats itself in a given period of time. The audio frequency range is generally considered to be 20 Hz to 20, 000 Hz. This covers the fundamental pitch and most overtones of musical instruments.
Gain: The measure of how much a circuit amplifies a signal. Gain may be stated as a ratio of input to output voltage, current or power, such as a voltage gain of 4, or a power gain of 1.5, or it can be expressed in decibels, such as a line amplifier with a gain of 10dB.
Gate: A dynamics processor that automatically turns off an input signal when it drops below a certain level. This can reduce the overall noise level of your mix by turning off inputs when they are not in use. Threshold, attack time, hold, and release time are some of the adjustable gate parameters.
Graphic EQ: A graphic equalizer uses slide pots for its boost/cut controls, with its operating frequencies evenly spaced through the audio spectrum. In a perfect world, a line drawn through the centers of the control shafts would form a graph of the frequency response curve.
Ground: Also called earth. Ground is defined as the point of zero voltage in a circuit or system, the reference point from which all other voltages are measured.

- In electrical power systems, ground connections are used for safety purposes, to keep equipment chassis and controls at zero voltage and to provide a safe path for errant currents. This is called a *safety ground*. Maintaining a good safety ground is essential to prevent electrical shock.
- In sensitive electronic equipment, tiny currents and voltages riding on the ground (so it's not truly zero volts) can cause noise in the circuits and hamper operation. Often a ground separate from the power ground is used as the reference point for the electronics, isolating the sensitive electronics from the dirty power ground. This is called a *technical ground*.

Hertz: The unit of frequency, equal to 1 cycle per second. Abbreviated Hz. kHz = 1000 Hz, and is usually pronounced "kay" (with "Hertz" implied) by sound professionals who ask for "a little more two and a half K" when they want you to boost 2.5 kHz (2500Hz).
Impedance: A measure of the AC resistance, capacitance, and inductance in an electrical circuit, measured in ohms; all of these three properties impede the flow of current. AKA: the higher the impedance, the lower the current. In audio circuits (and other AC circuits) the impedance in ohms can often be much different from the circuit resistance as measured by a DC ohmmeter. Maintaining proper circuit impedance relationships is important to avoid distortion and minimize added noise.
Level: Another word for signal voltage, power, strength or volume. Audio signals are sometimes classified according to their level. Commonly used levels are: microphone level (-40 dBu or lower), instrument level (-20 to -10 dBu), and line level (-10 to +30 dBu).

Line level: A signal whose level falls between -10 dBu and +30 dBu.
Main (house) speakers: The main loudspeakers for a sound reinforcement system. These are usually the largest and loudest loudspeakers, and are usually positioned so that their sound seems to come from the area of the main stage.
Master: A control affecting the final output of a bus on which one or more signals are mixed. A mixer may have several master controls, which may be slide faders or rotary controls.
Mic level: The typical level of a signal from a microphone. A mic level signal (usually but not always coming from a microphone) is generally lower than -30 dBu. With a very quiet source (a pin dropping?) the signal can be -70 dBu or lower.
Mic preamp: Short for microphone preamplifier. An amplifier whose job is to bring the very low microphone level signal up to line level, or in the case of a mic preamp built into a mixer, the mixer's internal operating level (approximately 0 dBu). Mic preamps often have their own volume control, called a trim control, to properly set the gain for a particular source. Setting the mic preamp gain correctly with the trim control is an essential step in establishing good signal-to-noise ratio and sufficient headroom for your mix.
Monitor: In sound reinforcement, monitor speakers (or monitor headphones or in-the-ear monitors) are those speakers used by the performers to hear themselves. In recording, the monitor speakers are those used by the engineer and production staff to listen to the recording as it progresses.
Noise: Whatever you don't want to hear. Could be hum, buzz or hiss; could be crosstalk or digital hash or your neighbor's stereo; could be white noise or pink noise or brown noise.
Normal: A wiring method which electrically ties together two jacks or two poles of one jack so that in *normal* operation, there is signal flow between them. Inserting a plug breaks this connection, allowing the signal path to be modified. *Normal* wiring is common in patchbays and *insert jacks*.
PA: Acronym for Public Address. An electronic amplification system used as a communication system in public areas. Today, people who work with PA systems like to say they're working in "sound reinforcement".
Pan, pan pot: Short for panoramic potentiometer. A pan pot is used to position (or even dynamically move) a monaural sound source in a stereo mixing field by adjusting the source's volume between the left and right channels.
Patch bay: A collection of usually a large number of jacks allowing convenient access to various points in a system's interconnect wiring. A patch bay can make re-routing signals very convenient without having to fish around with cables in the back of racks or consoles.
Parametric EQ: A "fully" parametric EQ is an extremely powerful equalizer that allows smooth, continuous, and independent control of each of the three primary EQ parameters: frequency, gain, and bandwidth.
PFL: An acronym for Pre Fade Listen. Sound folks call it being able to solo a channel with the fader down.
Phantom Power: A system of providing electrical power for condenser microphones (and some electronic pickup devices) from the microphone input jack; it provides +48V across two pins in a 3-pin XLR. The system is called phantom because the power is carried on standard microphone audio wiring in a way that is "invisible" to ordinary dynamic microphones. Generally, phantom power is safe to use with non-condenser microphones as well, especially dynamic microphones. However, unbalanced microphones, some electronic equipment (such as some wireless microphone receivers) and some ribbon microphones can short out the phantom power and be severely damaged. Be careful!
Phase: The time relationship between two signals, expressed in degrees around a circle. 0 and 360 degrees represents an in-phase relationship – both signals change in the same way at the same time. Anything else is out of phase. 180 degrees out of phase is a special case which, for a continuous waveform, means that at any given time the two signals have the same amplitude but are opposite in polarity. The *phase reverse* switch found on some mixers or mic preamps actually reverses the signal polarity. When out-of-phase signals are mixed, there will be some cancellation at certain frequencies, the frequencies and the degree of cancellation being a function of the amount of phase shift and the relative amplitude of the signals. Attention to mic placement and careful listening will allow you to use this effect creatively.

Post-fader: A term used to describe an aux send (or other output) that is connected so that it is affected by the setting of the associated channel fader. Sends connected this way are typically (but not always) used for effects.
Pre-fader: A term used to describe an aux send (or other output) that is connected so that it is not affected by the setting of the associated channel fader. Sends connected this way are typically (but not always) used for monitors.
Return: A return is a mixer line input dedicated to the task of returning processed or added sound from reverb, echo and other effects devices. Depending on the internal routing of your mixer and your own inclination, you could use returns as additional line inputs, or you could route your reverb outputs to ordinary line inputs rather than the returns.
Reverb: The sound remaining in a room after the source of sound is stopped. Reverberation and echo are terms that are often used interchangeably, but in audio parlance a distinction is usually made: reverberation is considered to be a diffuse, continuously smooth decay of sound, whereas echo is one or more distinct, recognizable repetitions of a word, note, phrase or sound which decreases in amplitude with every repeat. Highly reverberant rooms are called live; rooms with very little reverberation are called dead. A sound source without added reverb is dry; one with reverb or echo added is wet.
Signal-to-noise ratio: This is a specification that describes how much noise an audio component has compared to the signal. It is usually expressed in dB below a given output level.
Surround sound: Multi-channel audio playback systems in 6, 8, or 10-channel formats. Surround sound is typically found in movie theaters and home theater systems.
TRS: Acronym for Tip-Ring-Sleeve, the three parts of a two-conductor (plus shield) phone plug. Since the plug or jack can carry two signals and a common ground, TRS connectors are often referred to as stereo or balanced plugs or jacks.
TS: Acronym for Tip-Sleeve, the two parts of a single conductor (plus shield) phone plug. TS connectors are sometimes called mono or unbalanced plugs or jacks.
Unbalanced: An electrical circuit in which the two legs of the circuit do not have the identical impedance to ground. Often one leg is also at ground potential. Unbalanced circuit connections require only two conductors (signal “hot” and ground). Unbalanced audio circuitry is less expensive to build, but under certain circumstances is more susceptible to noise pickup.

Performance Hall Microphones and Their Uses

- SM 58: Dynamic (Cardioid)
  - Common uses: Live vocals
  - Rugged, sound source should be very close to the grill
- Beta 58: Dynamic (Supercardioid)
  - Common uses: Live vocals, Voice-over
  - Sounds brighter than the SM58
- SM 57: Dynamic (Cardioid)
  - Common uses: Guitar & Bass amp, Brass, Percussion
  - Rugged, very similar to the SM58
- MD 421: Dynamic (Cardioid)
  - Common uses: Toms, Bass amp, Guitar
- e609: Dynamic (Supercardioid)
  - Common uses: Guitar amp, Brass
- KSM 137: Small-Diaphragm Condenser (Cardioid)
  - Common uses: Acoustic Bass & Guitar, Cymbals, Percussion, Strings, Piano, Woodwinds
  - Delicate mic, Requires phantom power

- AT U853R: Small-Diaphragm Condenser (Cardioid)
    - Use: Hanging microphone found in the Penthouse for distance miking
    - Delicate mic, requires phantom power
- SM 27: Large-Diaphragm Condenser (Cardioid)
    - Common uses: Acoustic Bass & Guitar, Brass, Strings, Percussion, Piano, Woodwinds
    - Phantom power required
- SM 87A: Large-Diaphragm Condenser (Supercardioid)
    - Common uses: Live vocals
    - Phantom power required
- PCC-160: Pressure Zone Microphone (Cardioid)
    - Common uses: Floor or wall mic for distance miking
    - Phantom power required, will pick up room sounds
- ULX2-58: Dynamic Wireless (Cardioid)
    - SM58 equivalent wireless
- WL185: Condenser Lavalier Wireless (Cardioid)
    - Basic wireless clip on mic
    - Phantom power required
- ULX2-87A: Condenser Wireless (Cardioid)
    - SM87A equivalent wireless
- DPA 4090: Condenser (Omnidirectional)
    - Hanging mics found in the Performance Hall for distance miking
    - Delicate mic, requires phantom power
- MX418S: Condenser (multiple pattern cartridges included)
    - Lectern Gooseneck Microphone

Stands & Safety

The hall has two basic types of stands: boom stands and straight stands.

Straight stands: Given that they are less versatile than boom stands, straight stands are generally used to hold microphones for vocal performance or speaking. We usually use straight stands for wireless mics because they are easier to move around the stage, but they can take any standard mic clip. Our straight stands have a heavy round base and an adjustable neck. The connector at the neck twists right to tighten and left to loosen; adjust the stand to the appropriate height and tighten the neck.

Boom stands: Given that they can be contorted into good positions for picking up sounds from many different instruments (including drums, strings, brass, woodwinds, etc), boom stands are more adept for miking instruments. When using a wired mic on a boom stand, the cable should be wrapped a couple of times across the length of the boom, over the pin, and down the length of the stand. Do not wrap the cord too many times; just enough to keep it from flailing everywhere.

Safety: Each time you want to make a change to a stand, loosen one joint, adjust the length/direction, tighten and move on to the next joint until the stand acts how you want it to and will hold the weight of the mic you're about to put on it. Pay attention to how the legs are aligned when you position the stand. If the stand seems unstable, position it so that one of it's legs is directly under the boom arm. Make sure every joint is tight before walking away from it.

When disassembling, also remember to work with one joint at a time. It is very easy to get your fingers pinched between the moving parts, so be very careful when adjusting and always tighten! The goal when disassembling, is to minimize the space it takes up, so shorten all lengths and align all arms and legs.

Cables

The Performance Hall uses XLR, TRS, Mini TRS, SpeakOn, Cross-wired SpeakOn, Ethernet and Longframe patchbay cables.

<u>XLR:</u> The most frequently used cord in the hall for passing audio signal. Three-pin XLR cords are able to pass a balanced signal that travels farther than unbalanced signal. Generally XLR cords run from a wired microphone and into Floorboxes, AV panels off-stage (AVPs), or to a snake.

The hall has 10’, 25’, and 50’ XLR, stored on the bottom shelf of the backstage cabinet. When running cable from a mic to a floorbox input, you should look at the length from the stand to the input and choose a cable length that will give you a few feet of slack, as performers may move the stands during soundcheck and occasionally midshow. Leave the slack at the base of the mic stand and not near the input location or somewhere in between.

If the stands will be in the same place during the whole show, you should tape down all the cables in order to create a safer walking space for the performers and a cleaner look for the audience. When taping on stage use the brown gaff tape; it will blend into the wood and make the stage look cleaner. Any backstage cable taping should be in black gaff as it is less expensive.

<u>¼” TRS Instrument Cable:</u> Also known as ¼” cable, “tip, ring, sleeve” or instrument cable. We don’t stock a variety of lengths in the hall because musicians typically bring their own. We do own a few shorter patch length ¼” cables that could run from an amp, keyboard or an instrument with a pick-up to a DI box. These are kept in the storage closet.

<u>Mini TRS:</u> Also known as ⅛”, 3.5mm or aux cable, are commonly used audio cables for connecting phones, ipods and computers to an audio system. While these devices normally have an ⅛” input, you often need to convert it to an XLR or ¼” output when connecting to a mixer. In the hall we have a TRS ⅛” to stereo XLR cable connected to the mixer at FOH for device connection, but we keep a spare in the storage closet.

<u>SpeakOn:</u> Used for running sound to floor wedges/monitors, usually from Floorboxes or AVPs. SpeakOn cables vary in length, so choose the appropriate length for each monitor. Floor Monitors that use a SpeakOn input usually will not require any external power; all the monitors we have are “passive” and therefore do not require any external AC power adaptor. SpeakOn is stored on the audio shelf next to the backstage cabinet.

<u>Crosswired SpeakOn:</u> ONLY used in the patchbay for connecting Venue outputs to the appropriate floorbox or AVP connections. These cords are about 2.5’ and stay in the amp room at all times; they are marked CROSS on the neck of the cable.

<u>Long-frame/Military cords [Patchbay cords]:</u> Green, red, and black cable used in the patchbay to connect mic inputs to the Venue or Recording Booth. Also used to split signal when necessary.

Coiling Cables

There are two methods for coiling an XLR cable and there is much debate as to which is better. In the hall, we use the “over/under” method. This same method can be used to coil almost all other audio cables.

First, hold the female end of the cable in your left hand with the connector pointed toward you. Pull your right hand out along the cable until you reach about 18” and bring that length up into your left hand while guiding it into becoming your first loop. Pay attention to the direction your right hand rotates as you guide the cable into making this first loop. You are about move your right hand in the exact opposite direction when making the second loop.

After grabbing this first loop in your left hand, pull another 18” with your right hand. This time you’ll loop it up under itself, by twisting in the opposite direction, and again place it in your left hand. The cable should still be moving in the same direction, away from you, it’ll just be coming in from underneath the current coil.

Continue repeating these two steps “over and under) until you hit the end of your cable. Inspect the cable for a proper coil and then use the velcro tie attached to secure. It should look like a series of same sized loops, no figure 8’s. If yours has a figure 8, you must start over. If you have both large and small loops within one coil, start over. If one of your tails is way longer than the other, fix it by adjusting the position of the velcro.

If you have a hard time getting good coils, use this video to help while practicing:
http://www.lifehacker.com.au/2012/04/use-these-alternating-coil-methods-for-knot-free-cables/

Speakers

<u>Main speakers:</u> Also known as the “mains”. In the Performance Hall our mains are in a L-C-R (left, center, right) configuration. They are Electro-Voice, or “EV” brand speakers, model: Xi2123A/106, and are flown to heights controllable from the automated rigging system “Scene Control”. The projection system has its own surround sound system with a L-C-R EAW KF740 Line Array.

<u>Monitors:</u> Also known as “wedges” or “floor wedges”. We have 3 different monitor speakers; 3 of the smaller 12” EV’s (TX 1122 FM), 2 of the 15” EV’s (TX 1152 FM), and 2 of the 12” Community (M12’s). Monitors should be placed in front of a musician who needs to hear himself, or his fellow musicians, more clearly. While the EV’s need to be set further away from the artist because of how the speakers are pointed within the wedge, the Communities have a wider range and can be placed much closer.

<u>Subs:</u> We own 2 EV X-SUB’s that live in the work lofts to the left and right of downstage. The subs supply the low end frequencies to both the main and projection systems.

<u>Sidefills:</u> The 2 side fills we have, EV Zx1’s, are built into the downstage left and right walls, just where the plank design ends. The side fills are also tied to both the main speakers and the projection audio system.

Sending a mic through the PA

<u>Setting up a mic</u>

1. Find out who or what you are miking and how versatile they need to be during the show. Choose a stand and mic accordingly.
2. Find out where the mic needs to be placed on stage. Setup the stand, attach the clip, and slip the mic into the clip. Make sure everything within the stand is tight.
3. If the mic is wired and you are using a boom stand,
    - Take your XLR cable and first plug the male end into the floorbox or AVP.
    - Carry the coil to the stand (when possible, follow along with other cables going in that same direction) and drop the coil under the center of the stand.

- Run the female end up the stand, wrapping it around once.
- As you hit the boom, make sure the cord runs over the tightening pin.
- Wrap the cord once more around the boom in the same direction.
- Plug the female end into the mic, making sure the cord has enough slack at the end so that the mic can be easily adjusted, but not so much that the cord is hanging all over.

4. Always leave the cable slack at the base of the mic stand, never at the input or somewhere in between.
5. Try to run the cables in right angles when possible and along with other cables nearby. However, avoid running microphone cables alongside AC power cables or SpeakOn unless absolutely necessary; this will generate noise in the signal.
6. Any loose cables onstage or backstage should be taped down after soundcheck is complete, and prior to doors opening. Remember to not tape down any mics that will need to move around during the show.
7. If the mic is wireless, check to make sure the battery has at least two bars for a rehearsal setting and three for a public performance. Batteries with less than two bars should be recycled in the "dead battery" bin.

<u>Patchbay</u>

8. This is where you will be connecting the signal directly to the SC48. Check the number on the floorbox or AVP input you plug into; it should be in the format of the letter M then a sequence of ascending numbers, example: M12. Take a long-frame cable and run it from that input to whichever channel you intend to use on the mixing board. Use different colored cables to show different groups of mics (*wired vs wireless* **or** *vocals vs instruments* **or** *input to split to SC48 vs booth).*
9. While patching, make sure the sound system is on by checking that the sequencer key is turned to "on" and that all its associated amps have turned on. While you're doing this, check the subwoofer amp and turn on manually if needed.

<u>Venue SC48 console</u>

10. Turn on the Venue if it hasn't been turned on yet. Once it is on, make sure all channels have been muted or have their faders turned down. Then turn the master fader (at the far right) up to "unity".
11. Select each channel you have patched mics to and label them by double clicking on the "Channel #" in the top left corner of the onscreen interface and naming what instrument or type of mic it is ("SL Vox" for stage left vocals, or "W1" for wireless mic #1). This makes it easier to keep track of your mics and keeps you from forgetting which channels are which.
12. If you are using a condenser mic, remember to mute the channel before turning on phantom power because the 48 volts will cause the mike to pop momentarily. If the mic is not muted, the entire house may hear an very loud and unpleasant sound. To engage phantom power:
    - Select the channel that needs the phantom.
    - Again, make sure the channel is MUTED and the fader is down.
    - Press the 48V button near the top of the Venue.
    - Unmute, gain and then fader up.
13. Find someone assigned as or willing to temporarily be your audio assistant to test the mics. Have them first go to each mic and test that each one is working by gently tapping the grill of the mic. You are checking to see that you are getting signal to the channel; if you see nothing, turn up the gain until you do. Remember to keep the channel muted and fader down while line checking.
14. Once you have completed the line check, you know all your patches are correct, and you have extra time and want to do things right, you can now do your pre-soundcheck soundcheck! Ask your assistant to go back to each mic and spend more time with each one. They should speak through vocal mic and play approved instruments, one at a time, to help you set gains and test monitors.
15. During line check, confirm that each channel is going to the appropriate main speaker (LR or C). To

do this, decide whether the sound from each independent channel should be coming out of the left and right speakers, the center speaker, or all three. *As a rule, vocalists should be routed to the center speaker only, instruments to the left and right and pre-recorded music again to just left and right (although left, center, and right is also an option if you need more level). On the board, select the channel, then choose the L/R or Center assigns from the master mains routing section.

16. Adjust the panning, limiter, comps, gate and effects (FX) as you see fit. These should all be adjusted pre-show; turning any of these on mid-show would have to be done very gradually and strategically. When done improperly, you could lose sound completely and totally throw off a performance.
17. Double check that the speakers are at the the right height for the show. For a lecture or panel, the center speaker can be the only one down; instrumental performances can have the left and right speakers down. For use with the projection screen, left and right should be set below the break in the screen. For a film screening from the booth, fly the line array below the break in the screen.

At this point the basic loop of mic to patchbay to Venue to speakers is complete. There are a bunch of technical processes in there, but basic mic amplification is taking place, which can be enough for some musical performances and most lecture settings.

Using a DI-Box

A DI-Box (Direct Input Box) allows you to connect high-impedance sources to a mixer without adverse effects. The most common uses are for connecting electric keyboards, guitars, bass guitars and most forms of instrument pickups directly to the mixer without using an amp or microphone. Plug the ¼" TS cable from the instrument's pick up into the DI (using the high-Z input) then plug an XLR into it's output and run that to a system input (floorbox, AVP, snake, etc).

If an amp is involved, but they still want a direct line as well, you'll have to split the source. Usually the musician will have his/her instrument plugged directly into the amp, so you'll have to break this chain. Pull that plug from the amp and inset it into either input of your DI box. Grab a shortie instrument cable and plug it both into the second input of the DI and then back into the input of the amp.

Setting up monitors for a musician

The basic idea here is that you're sending a certain mix of sounds back to the musicians so they can hear themselves better or other instruments that might be too quiet or far away for them to hear otherwise. This mix is, more often than not, different from the house mix that the audience hears. You are sending audio from a channel to an auxiliary output on the Venue (a separate output from the mains), which goes to the patchbay, then to a floorbox or AVP, then to a monitor.

Grab a monitor

1. First, figure out how many monitors will be needed by talking to the band or band leader. Most musicians will only need one, but sometimes a drummer or lead singer will ask for two.
2. Bring the monitor to stage and set it downstage of the musician who will be using it. Take a look at the direction the speakers inside are facing and try to point them toward the musician's face. If you are using a Community M12 stage monitor, the square speaker (behind the grill) should be on the side closest to the musician.
3. Find a SpeakOn cable with enough length to reach between the monitor and the nearest floorbox.
4. Connect either end of the cable to the floorbox (or AVP) Speakon output that you intend to use (just pick one, they're all the same) and remember that number for patching.
5. Attach the other end of the speakon to the monitor itself. If there are two identical ports on the back