A BASIC INTRODUCTION TO CONCERT SOUND ENGINEERING

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Coverage:

This Guide is my attempt at a summary to aid the training of volunteers in the basics of live sound mixing with an emphasis on non-technical details, and oriented towards acoustic music settings. It is based upon my experiences with sound at the Laurel Theatre and a few other venues over the past 20 years and includes details on the social aspects of being a good sound engineer along with a fairly quick overview of technical aspects. It is designed to go along with a two-session workshop covering basics, and though I give some technical suggestions, these are not designed to provide anything other than complementary material to what you would learn by reading a good guide to sound or the manuals for whatever sound system you are using. Even more important is hands-on experience, and watching over the shoulders of experts. Some references are listed at the end of the Guide along with some Web sites for more info.

The below includes essentially no coverage of many highly important components of live sound. Notably excluded are details of effects (e.g. reverb, delay, and harmony units) which are used to mix in an external signal along with the original audio, signal processors (e.g. compressors, limiters, gates, etc.) which are used to modify the audio signal, and speaker details such as crossovers.

Table of Contents:

I. Goals of Live Sound Mixing

Your main responsibilities are:

1. Do what the artist wants.

2. Get the best sound in the hall you can with the equipment available.

As a subsidiary to these, you are typically an advisor to the venue (or the artist if you are traveling with them) regarding the appropriate equipment necessary for providing a quality aural experience for the audience. This means, if you are engineer for a venue, carefully reading the sound riders for a visiting artist, being certain that you will have the equipment requested available or, if not, being certain that the artist has approved whatever modifications you suggest. If you are traveling sound engineer for an artist, this means making certain that a complete, detailed sound rider is made available to all venues well before the performance, and making yourself available to discuss these with a representative of the venue.

Sometimes 1 and 2 above are not totally compatible, in which case you must trade-off some of the artists desires with what you feel is best for the audience. The most difficult part of this typically involves the potential conflict between the artists desire for monitor sound and the potential for this to cause difficulties with sound in the hall. In general though, most artists would agree that you have responsibility for sound in the hall, and they won't try to control it but rather give you suggestions.

II. Following the Signal

In the below we will proceed with following the audio signal from the artist through the various wires and equipment until it reaches the speakers. This is the usual procedure to apply whenever there is some problem with the system that you don't know the cause for - follow the signal, carefully checking each lead, plug and piece of equipment until you isolate the problem. A bit of background definitions and equations are given in the first section, but these really gloss over alot of the details, so see the references for more on this.

A. Background definitions

Measuring audio signals: The key unit in audio is the decibel (dB) where deci is from the Latin for one tenth and bel is from Alexander Graham. A Bel is a logarithmicly scaled measure defined as the logarithm (base 10) of the ratio of two numbers. Since 1 Bel has 10 decibels, the formula is

$$\text{decibels} = 10 \log(A/R)$$

which measures the relative relationship between A and a reference R. The reason for using logarithmic scales here is twofold: the human ear responds to sounds in much more of a relative manner than an additive manner, and the range of measurements of various audio signals is so large that on a linear scale sufficiently large to cover the entire range, low signal levels would be indistinguishable from zero. To calculate dB therefore, you need a reference level as well as a signal to compare to this reference level. For power levels (measured in watts say) a doubling of a signal corresponds to an increase of 3dB since comparing a signal 2A to a reference A gives

$$\text{dB} = 10 \log (2A/A) = 10 \log(2) = 10\ (.301) = 3 \text{ (approx.)}$$

and similarly halving the power corresponds to a decrease of 3dB. Note that a 10 fold increase in a signal corresponds to a 10dB increase. For dB to be useful, it's important to know the reference level and there are several different dB measures depending on what you are measuring and what the reference is. The above formula is for power ratios, while for voltage ratios to be measured in decibels, it is necessary to remember that power is proportional to the square of voltage (from Ohm's law $V = IR$ and $P = I^2 R$)

$$P = V^2 / R$$

where P is power, V is voltage, I is amperage, and R is resistance. Due to this, calculating dB differences between two voltages (or two sound pressure levels - SPL) is

$$\text{decibels} = 20 \log (A/R)$$

You will see lots of different dB measures including
dBV for the case of voltage in which the reference R is 1 volt rms

dBm for the case of power output with reference R 1 milliwatt
dBu for the case of voltage with reference R 0.775 volts

In the above rms stands for root mean square and is useful in describing the average level of a varying signal such as a complex waveform. Note that for measuring sound pressure level (SPL) which is a measure of the force of air pressure provided by a sound system at a location, a doubling of SPL corresponds to a 6dB increase (here 0dB for SPL corresponds to the threshold of hearing in the ears most sensitive frequency range - about 1kHz). Another rule to keep in mind is the inverse square law - for a fixed sound source, for each doubling of distance from the source, the SPL will drop by 6dB since the power produced by the source is spread over approximately four times the area.

B. Microphones and other inputs

There are several different types of microphones, with hosts of manufacturers for each. The basic types are:

Dynamic: here the mic is like a speaker in reverse, since there is a diaphragm which vibrates according to the sound applied, causing a coil of wire to move in a magnetic field producing a very small electrical signal.

Condensor: here the sound is picked up by a capacitor, which must be provided power either from a battery, or from phantom power provided along the mic line (this is a DC current provided by either a mixer or by a separate phantom power unit)

There are also a wide variety of other types (e.g. ribbon mics, radio mics, electret condensors, etc.), but the vast majority of live sound work is with the above two types.

Key factors determining differences between various microphones are their frequency response and the pickup pattern. The frequency response is quite different for mics designed for use by vocalists than for those designed for various instruments, so different mics are typically used for these purposes (e.g. a Shure SM 57 for instruments and a Shure SM 58 for vocals). The pickup pattern for the majority of live sound mics is either cardioid, with a heart shaped pickup pattern around the central mic axis, or supercardioid, which is more directional than a cardioid, particularly designed for cases in which you want to reject some acoustic signal from the sides that a cardioid mic will generally pick up. Exactly what mic to use where depends upon what you have available, the artists preferences, and your experience with the particular vocalist or instrument.

Other inputs aside from mics are direct line signals (line level is -10 to +30 dBu and is much higher signal level that mic level which is typically -40 dBu or lower) which are typically obtained from an on-stage amp, or from a DI box (DI stands for Direct Injection, though these are typically just called direct boxes). A DI simply converts an unbalanced, high impedance signal from an instrument pickup or amp to a balanced low-impedance signal. There are two kinds of direct boxes - passive, which is essentially just a transformer inside a shielded box for converting a high impedance to a low impedance signal, and active, which require a battery or phantom power to operate.

C: Lines - balanced and unbalanced

There are two basic types of lines used in audio:

Unbalanced lines have a single lead running down the middle, with a wire braid shielding around it. Here the hot signal (e.g. in-phase or +) is in the center wire and the braid serves as both ground and the cold side (e.g. out of phase or -) of the signal is carried by the braid. The end of the line typically has a quarter-inch jack plug with just a tip and sleeve. Unbalanced lines are used typically only for the relatively short leads from an instrument to an amp or DI.

Balanced lines have two center wires carrying the in-phase (hot) and out-of-phase (cold) signal, with a wire braid around them both which is the ground. The typically end of the line is a Cannon or XLR type plug with the male end sending the signal and the female end receiving the signal.

What you want to be sure is that all connections to the mixing console (and any snake going to the mixer from the stage) is with balanced lines. Otherwise noise would be picked up in an unbalanced line and dumped right into the mixer. A balanced line greatly reduces noise problems (due to spurious electrical transients produced along the length of the line) since the shielding dumps this to ground in the mixer.

The other caution is to be sure not to use any speaker lines for connecting the audio components prior to the amplifier stage. Speaker wires have two wires to carry signal, but have no wire braid shielding around these. This shielding is essential to reject radio frequency and other interference that would greatly compromise the low level signals being sent to the mixer.

D: The mixer

(i) Gain setting

A very important factor in making a clean, even mix possible is an appropriate gain structure for all inputs. What this means is that all signals coming into the mixers internal circuitry are at roughly equivalent levels. This is necessary to ensure that no one input controls the amount of headroom (how many dB increase is possible above nominal operating levels) available from the mixer. Setting the gain (e.g. how much amplification goes on in the pre-amplifier stage of the mixer) for each input channel appropriately not only ensures that no one input overwhelms the mixer, but also ensures that the lowest possible noise level is achieved from each input.

Achieving appropriate gain structure is relatively easy, but requires carefully going through each input channel to set the gain (or trim as it's often called) for the preamp stage so that only the appropriate amount of signal is sent into the mixer. Exactly how to do this depends somewhat on the mixer being used. A standard approach is to sent the channel slider at center (0dB), and adjust the input gain on each channel while that channel is being used at the level it will be during the performance (by having the artist sing or play into it) so as to have the VU or LED meters on the mixer show 0 dB. It is often best to roughly adjust the channel EQ at this time as well, since this affects the level from that input going to the mix.

(ii) Channel levels and EQ

Once the gain level is set for each channel, there are two other main controls of the input signal - one is simply the slider (or fader as it often is called) which controls how much of that input is sent to the output of the mixer. The level here should typically be close to the center location if you have set the gain correctly, but will certainly be modified from this as the entire set of inputs are mixed together, and should be taken out of the mix completely when the input is not used (a mute button does when you don't want to have to remember or write down the slider position).

The channel EQ (equalization) allows adjustment of particular fixed frequency ranges for each input separately. This allows you to boost or reduce certain frequencies depending upon the needs for a particular input. The exact frequencies ranges used (there are typically Hi, Mid and Low EQ adjustments) vary considerably from mixer to mixer, as well as the structure within these ranges that is affected by the EQ. Some mixers allow you to adjust the frequency affected (particularly in the midrange).

E. House EQ

This is a graphic EQ that allows you to boost or cut (up to a certain dB) a variety of frequency ranges. The frequency ranges are set up logarithmicly, from typically 20 Hz to 20,000 Hz (10 octaves), so that each slider on the EQ affects an equivalent ratio of frequencies, though the bands covered by any two sliders will be quite different numerically (e.g. the first slider might cover only 5 Hz while the last one might cover 4000 Hz).

You typically set the House graphic based upon the room acoustics, and your expectation for how the room will sound when the audience arrives. Note that the audience can make a considerable difference in how a room sounds, so it is not a good idea to "over EQ" a room during sound check (e.g. cut out alot of frequencies) unless you know from experience that it is needed. There are a variety of methods to "ring out" a room to find harmonic frequencies that might make the sound harsh or indistinct. One method is to pass white noise (e.g. noise with equal power at all frequencies) through the house system and use a frequency analyzer in the house to pick out what frequencies are enhanced, and then reduce them using the graphic EQ. Another method is to simply place a microphone (preferably of the same type you are using on stage) in the center of the hall facing the stage, and slowly bring up the mic level until you get feedback squeals, cutting out the main frequencies of those squeals. You don't want to overdo this though, because you can greatly deaden a room.

F. Amps and speakers

The output from the mixer goes typically first to a house graphic EQ and then to an amplifier. The amplifier boosts the relatively low signal coming from the mixer to a power level sufficient to drive the speakers that you are using. Amplifiers are heavy and produce alot of heat. It is very important that they have plenty of air flow around them. An amp needs the most power for low frequencies, less for midrange frequencies and the least for high frequencies. It is very important to match the power produced by an amp with the sound requirements of the type of music and the venue, as well as the power that the speakers can handle. It is typical to run an amplifier wide open (e.g. at the maximum output level) so that all variation in output level is completely controlled by the input level to the amp from the mixer. Troubles arise when the input level is too high for a particular amp - this leads the amp to try to reproduce the signal at the appropriate power level, causing clipping. This essentially chops off part of the amplitude of a waveform signal, and causes the speakers to try to reproduce a much higher amplitude waveform than the amp is providing power for. This leads first to distortion, and then, if it continues, the speaker fries (e.g. the cones rip or the coils burn up).

Speakers are of several types, with the majority consisting of coils of wire in a magnetic field driven by the amplified signal causing a cone of material to vibrate and produce a sound wave of the appropriate waveform. Horns are a means to focus the sound in particular directions. Speakers are horrendously inefficient, in the sense that a very small fraction of the power supplied to them actually gets transmitted into sound. Much of the power is lost as heat from the coils. Speaker systems can include separate speakers for different frequency ranges, with different amps for each speaker (two speakers here would be called a bi-amped system) and active crossovers controlling what frequency ranges are sent to each speaker. The single cabinet speakers typical of home systems and smaller PA's have more than one speaker in each with a passive crossover which splits the frequencies between the speakers. Here passive means that you have no control over how the split occurs - it is hard-wired into the speaker.

G. Equipment provided by the artist

As prices of out-board gear have declined, more artists are carrying with them a variety of equipment that they wish to use. Typically, this involves microphones, effects processors, in-ear monitor units and recording devices. As you will generally not have a great deal of time to setup and deal with this equipment before a performance (unless you have worked with the specific equipment previously), it is essential that you go through in a step-by-step procedure every change you might need to make to the equipment during the performance. This includes how to pause a recording device, how to mute and un-mute a microphone including wireless receivers, how to adjust an effects unit, etc. The objective is to avoid at all costs the possibility that a performance will have to be stopped so that the artist can show you how to do something. Some units have timed settings which can power them down after some period of non-use and you need to be able to bring them back to life quickly if they are needed and reset them as appropriate.

In general, you should assume that the artist is well aware of the appropriate application of the equipment they are carrying. However, do not be bashful about making suggestions for issues such as mic placement of large-diaphrahm condensors, clip-on mics for fiddles, internal mics on guitars. etc. If something doesn't sound right during sound check, it isn't going to magically fix itself during the show. Take the time necessary to try out various changes, particularly if the equipment has been obtained recently by the artist, or has been borrowed.

III. Mixing

A. Setting up the stage

The stage arrangement is critical for several reasons:

(i) Mic placement can be very important in some venues and for some artists. Generally the artist will have a very good idea as to how to best set up the mics for their instrumentation - follow their suggestions. If the artist is inexperienced, inform them as to the best way to use the mics you have available, and offer suggestions about both singing into them as well as placement for their instruments. Be aware that any mic which is out in front of the stage offers the potential for feedback problems in the house, so keep them back far enough that this doesn't occur. If the artist is going to walk around with a mic in the hall, this requires a specialized mic (note that it's not a problem with most direct pickups inside instruments, but can be if it's an internal mic).

(ii) I will say a bit about speaker placement in the hall but for much of this guide I've assumed the venue of concern to you has a fixed speaker and sound system. If it does not, you should be guided by past experience in the hall, and if you don't have any, you soon will! So much of speaker placement depends upon the specifics of the hall acoustics and the speakers properties, that there's little general advice I can give. One is to not be afraid to move the speakers around a bit if the current arrangement doesn't sound as you'd like. Small changes in just the vertical or horizontal angle of speakers can make big differences in clarity in the hall as well as turning a major feedback problem into a minor one.

(iii) Monitor placement is critical not only to ensure the artist can hear what they want to hear, but also because a large number of monitor feedback problems can be reduced by appropriately placing the monitors relative to the mic placement. The general rule is that no mic should be pointed towards a monitor, and preferably they should all be aligned perpendicular to the plane of the monitor speaker. Note that very loud monitors, particularly if any are angled towards the audience, may well interfere with sound in the hall. Loud monitor systems typical of highly amplified bands automatically require higher sound levels in the hall than might be preferable otherwise. Thus it is preferable to maintain a monitor level high enough that the artist hears what they want, but not so high that it causes problems with sound in the hall. How much interference occurs with the hall sound is greatly affected by the on-stage acoustics, and whether any stage monitor sound is bounced back to the hall from loud monitors. Any onstage instrument amplifiers (typically for electric bass or guitar) also have the same potential interference with sound in the hall. Again, don't be afraid to make changes in positioning of monitors if you are having difficulties, particularly feedback problems.

(iv) You don't want to unnecessarily block the audiences view of the artist, but this often takes secondary consideration to mic and monitor placement. So be aware of the way the stage looks, particularly with any stage lighting you have. I often find it best to have the stage lights set up as they would be for the show during the sound check to allow the artist to get a feel for the light level on stage.

(v) When the stage is set, and you are done with the sound check (thus you are certain all cables and lines are functioning correctly), carefully dress all cables on stage and in the hall so that audience members and artists won't trip on them. Typically this means you coil excess lines in locations that are out of the way, and tape down with gaffers tape any lines that people could trip over. I typically coil excess mic lines at the base of each mic stand. Dressing the cables appropriately adds to the perceived neatness of the stage for both audience and artist, and is an additional measure to both as to how important the concert producer views the performance. A messy stage detracts from the overall ambience of the performance.

B. The House Mix

First, pay attention to what the artists instruments sound like acoustically, if they're acoustic, or what is coming out of any on-stage amps, if they're electric. Generally, you want to make the instrument sound like that in the hall. Secondly, if the artist has recordings available, listen to them prior to the performance to get some idea as to how they might prefer a mix. At the same time remember that artists may not at all want a live performance to sound just the same as a recording. It is typical that an artist will give you very little guidance as to how to make the house sound. If possible, after a basic mix is down, I suggest that you request a band member, manager, or other person who is travelling with the artist give you some feedback on how the mix sounds. Artists with pickups in their instruments may well walk into the hall and request changes based upon how they prefer the sound.

There are three basic components to the house mix: (i) the overall level, (ii) the relative levels of various instruments and vocals and their channel EQs and (iii) the graphic EQ and any other effects in the mix (e.g. reverb units). The overall level is mostly determined by the size of the hall, the type of music, and how rowdy the audience is. It can also be affected in part by how loud a monitor mix there is. In general, the level set during a sound check will be changed when the audience is in the hall - people in the hall tend to dampen out alot of the sound you will hear during the sound check. If you have experience in the hall you will probably automatically accentuate certain components of the mix during the check because you are taking this into account. It is important to walk around the hall during the sound check to listen for any hot spots, as well as to hear how the mix sounds in different parts of the hall. This is particularly important if you are doing a stereo mix in the hall. If possible, I suggest you walk around the hall a bit during the performance as well - don't just stay at the board and assume the sound everywhere in the hall will be the same as it is there.

In general, in the house mix you ought to be able to pick out each instrument clearly, and all vocals should be distinct. If the mix sounds "muddy", a basic start to getting it fixed is to turn down the overall level in the house, and adjust the EQ and level on each channel so that each instrument becomes clearly defined. This is easier to do if the overall level is reduced, but is also made easier if you can "solo" each channel and hear it in the headset as you adjust the EQ. Keep in mind that the headset sound will be quite different from how the hall sounds for that instrument, and the channel EQ should be adjusted for the hall. You can also solo an instrument to the house, but I have found that most artists do not particularly like you to spend any extensive time running just a single instrument through the house while a whole group is playing. Therefore, before the entire band does a piece, I request a run through of each channel for just a brief time to get a basic level (e.g. adjust the gain pot on each channel), and a very rough channel EQ.

A house graphic EQ is used to accentuate certain response frequencies in the hall that might be absorbed due to the hall acoustics as well as to reduce particular frequencies that arise due to the hall harmonics or due to feedback. Generally you tend to reduce rather than boost particular frequencies, but it is not at all atypical to start out with a W-looking setup - boosting the lows, reducing slightly the mid lows, raising the mids, lowering the hi mids, and boosting the hi's. This is only a starting point however, and you will have to adjust any graphic to suit the particular room. During a mix, the graphic can be used to remove "harshness" in particular frequency ranges, as well as boost the clarity of some. However, since the graphic is typically used to affect the entire house mix, if the problem is with a particular channel it is preferable to first try to fix the problem by adjusting the EQ on that channel. Having a graphic EQ on a particular channel allows a great deal of flexibility as to how that channel sounds, but this is rare in my experience except when an artist has a small number of instruments and wants alot of control over these. Of course, many artists have small graphic EQs with a few frequency bands on stage with their instruments, particularly if they are using pickups. You can ask them to change these if you feel it appropriate.

C. The Monitor Mix

The purpose of monitors is to allow the artist to hear what they want to hear, and should complement whatever on-stage sound there is from the house system. The onstage mix - or mixes if there's more than one - is whatever the artist wants. The basic two choices are having the monitors the same as the house mix or having a mix that accentuates particular instrumentation or vocals. There are few general rules of thumb here, as this is very much artist dependent. Typically it's not necessary to have any instrument which onstage is very loud to be in the monitor mix - such as drums and bass - but this depends on the size and arrangement of the stage.

As monitor speakers are typically quite different in sound from the hall speakers (generally the hall speakers will be of higher overall sound quality), it's important to keep in mind that what the artist hears will not be the same as what is heard in the hall, even if you are using the same mix on stage and in the hall. For this reason, and because it is the monitors that often give any feedback problems, it is preferable to have a good graphic EQ available for the monitor mix, even if this means you can't use it for the hall or have to run the hall in mono so you can use one side of a stereo graphic for the monitor mix. A good graphic can solve lots of onstage problems with overall "feel" as well, since you may well not have channel EQ controls (e.g. Hi, Mid and Lo) for the monitor mix separate from that for the house mix.

If the monitor mix is the same as the hall, you typically have two options: pre-fader and channel EQ or post-fader and channel EQ. Most mixing consoles "Monitor" send will be pre-fader and EQ, which means that the monitor send is not affected by changes you make to either that channels level in the house or its channel EQ. This is typically what artists want, since the channel EQ setting you are using for the hall will not in general be the same as what you'd want in the monitors. Additionally, artists would get quite confused onstage if the monitor level for the instruments kept changing, as they would if you used a post-fader monitor send and you modified the house mix during the show. Only high-end mixing consoles (or having a separate console on stage as is used for large venues), typically allow you to do a separate EQ for each channel for the monitor mix. Thus, making the monitor mix the same as the house is really a misnomer - you don't want to do this. What you want to do is set the monitor level for each channel approximately the same as you have for the house mix, and then modify this as requested by the artist.

If possible to set up, it is very useful to have a way for you to hear the monitor mix, using a headset, at the mixing console. Some mixing boards make this easy - you just switch amongst various inputs for the headset. For other boards you may have to route the monitor mix to a particular channel and monitor that channel in the headset. Whatever way you do this, it makes it much easier during the sound check to make the changes the artist requests, and provides the opportunity during the performance to make modifications (if a signal from the artist tells you to do so) in the monitors which don't go beyond what the artist may want. If the monitor mix is appropriately complex (e.g. if there are several band members, or several separate monitor mixes), it's a good idea to check with the artist before hand about any typical signals they might give you during the performance about changes they'd like in the monitor mix. Most artists don't want to interrupt the flow of the performance to give you instructions for monitor changes, so typical signals are look at you, point to an instrument and give a thumbs up or thumbs down. If there is an intermission, be sure to check with the artist about how the monitor mix is, and whether they'd like any changes.

IV. Social skills

A. Artist Relations

You may be the one individual in the venue whom the artist deals with in any extensive manner, so you should remember that you are representing the hall, and the artist may well remember the experience there based in part upon how you dealt with them. Therefore, be friendly, courteous, treat the artist with respect, and be sure to listen attentively to their requests and suggestions. Start by introducing yourself, asking them if they have any particular suggestions, and then get into details such as:

(a) What kind of mix do they want in the monitors?

(b) Are there any particular suggestions they have for the house mix (e.g. one particular vocal or instrument out front, an even mix, EQ settings for any instruments, etc.)?

(c) How do they want the stage arranged (if they haven't provided a stage plot - if they have you should have already had the stage arranged before the sound check)?

(d) Do they prefer to hear the monitor mix first, then bring in the front of house mix, or the reverse (this is a matter of artist preference and also depends upon the size of the hall)?

B. Audience Relations

Again, you are often the most readily visible person in the hall who "looks official". Thus it's expected that you be courteous to audience members, and be able to direct them to rest rooms, water fountains, refreshments, etc. It is not unusual at all to get requests from audience members for changes in the sound during a performance (if you don't get many of these, you either have a very laid back audience, or you are doing a great job). Be as polite as possible to these individuals, and do listen to their comments, particularly if they are sitting in a part of the hall that you are not able to get to, and which could require some changes. Then do as you feel best, but remember that the audience are paying customers and a regular stream of requests from them probably means you should make some changes.

C. At the End of the Show

Thank the artist and ask them if they have any suggestions about the sound arrangements. If necessary keep out of the artists way during breakdown, and let them remove their instruments and equipment first before unplugging all mics, etc. Clean up your mess, and check with the hall managers about any problems before leaving. You may find it useful to maintain a list of artists you have run sound for, and the basics of the setup and channel EQs you used, in case you have to run sound for them again.

V. References

A few books that you may find useful:

Davis, Gary and Ralph Jones. 1989. The Sound Reinforcement Handbook (Yamaha). (2nd ed.) Hal Leonard Publishing. Milwaukee, WI. (Classic, very complete guide, including many technical details))

Fry, Duncan R. 1992. Live Sound Mixing (2nd ed.) Roztralia Publications, Victoria, Australia. Available from: ShedWorks, 4411 Brookford Ave., Woodland Hills, CA 91364 (818)225-1809. (Easily followed, non-technical guide for aspiring concert sound engineers covering all the basics with useful hints)

Moscal, Tony. 1994. Soundcheck: The Basics of Sound and Sound Systems. Hal Leonard Publishing. Milwaukee, WI. (Concise breezy guide to sound system components, with few technical details).

Rumsey, Francis and Tim McCormick. 1994. Sound and Recording: an Introduction (2nd ed.) Focal Press, Oxford, UK (Solid, basic audio components descriptions, with focus on recording rather than live sound)

Web Sites - there are many of these for various sound companies. My favorites are

http://www.prosoundweb.com/live/   live audio portion of ProSoundWeb

This includes a message board for pros interested in live sound and features hosts of useful info on equipment and techniques, and is generally tolerant of questions from beginners.

http://www.synaudcon.com/   The Synergetic Audio Concepts Home page

A premier organization of sound professionsal - this site has great FAQ's on many aspects of live sound.

http://www.mackie.com/   Mackie's Home Page

Has a great glossary of sound terms as well as details on their products.

http://www.northwindvt.com/contradance/sound/ The Contra Dance sound Home page.

This includes Bob Mills' excellent starting guide for mixing sound for live dances - All Mixed Up.

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