BASICS of GOOD SOUND in the CHURCH

MICHAEL E. LUITHLE
Director of ITS
Church of God of Prophecy
International Offices
I. INTRODUCTION TO AUDIO

A. How Sound Works
   1. Sound is actually a much more complicated topic than one might think. To the average individual sound is simply what we hear. But a more accurate definition is more like what is all around us no matter if we can hear it or not. In fact what we cannot hear is just as important to us as what we do hear. Sound actually for us is sound pressure. We use the term SPL to really measure what we hear. SPL is going to be really important when we start talking about volume or loudness. Let’s start though with frequency and wavelength.

B. Frequency, Wavelength & Volume
   1. Sound is created when vibration results in a wave of air pushed across a distance at the standard temperature/pressure of air. Sound moves at approximately 343 meters per seconds (m/s) or 767 mph. Looking at a picture of a sound wave, there are peaks (highest point above the baseline), and troughs (the lowest point below the baseline). The wider the distance from peak to peak, the lower the pitch of the sound. Therefore, a sound wave that moves up and down over a short wavelength produces a higher pitch sound. Frequency is the distance of the wavelength of a sound over a particular distance.
   2. The higher the number of peaks, the higher the frequency of the sound. Amplitude is the height that the wave moves away from the baseline. It would appear that amplitude would equal volume. However, to the human ear, certain frequencies appear to be louder in volume even though their amplitude is the same.
   3. The human ear does not detect every frequency at the same volume. Frequencies in the speech region appear louder than other sounds at the same amplitude. For example, a sound at 100Hz (bass) will sound quieter than a sound at 1000Hz (mid-range) even when the amplitude of the sound is the same. Amplitude is measured in watts. When purchasing amplifiers, consideration of how many decibels is needed to increase the amplitude of the sound.
4. The adjacent table demonstrates that a decibel is the perceivable measure of sound. A sound engineer needs to understand the following from this:
   The decibel is a logarithmic scale. Changing decibels results in large changes of watts, especially at high volumes.
   A decibel is a good scale to use when measuring sounds because, for most individuals, 0 is silent and 100 is painful.
   Audio at 85 decibels may seem normal, but to others it may sound like a jackhammer.
   Volume, while measured in decibels, is not a scale of perceived sound quality.
   1) When a sound system is set to 80 decibels, a hearing impaired person may complain that the audio is too loud. However, the actual issue could be that they cannot distinguish voices from instruments.
   Decibels of some frequencies may result in complaints of “too loud” levels. Adjusting bass frequency appears to lower the volume, resulting in a more pleasant audio experience.
   Sound engineers must become master mixers, considering all facets of audio. This results in the best audio experience, regardless of volume, stage talent or amount of instruments or voices.

C. Basic Components of Sound System

1. There are basic components of sound that translate into the pieces of equipment necessary to produce quality audio.
   The source or input. This is the device or even the person that creates a sound wave. In order to capture this we will need a microphone. When we talk about microphones we will come back to the topic of frequencies.
   The output or replication device, which is a speaker. Obviously, the goal is to replicate the input sound wave and, with minor changes, reproduce it.
   The Sound Desk, Sound Board, the amps, and all of the other electronic devices are utilized between the input and the output.

<table>
<thead>
<tr>
<th>Source</th>
<th>Power (watt/m²)</th>
<th>SPL (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Threshold of pain</td>
<td>.1</td>
<td>100</td>
</tr>
<tr>
<td>Jet noise from 500 feet</td>
<td>1</td>
<td>120</td>
</tr>
<tr>
<td>Medium-loud rock concert</td>
<td>.1</td>
<td>110</td>
</tr>
<tr>
<td>Circular saw</td>
<td>.01</td>
<td>100</td>
</tr>
<tr>
<td>New York subway</td>
<td>.001</td>
<td>90</td>
</tr>
<tr>
<td>Jackhammer from 50 ft</td>
<td>.0001</td>
<td>80</td>
</tr>
<tr>
<td>Vacuum cleaner from 10 ft</td>
<td>.00001</td>
<td>70</td>
</tr>
<tr>
<td>Normal conversation</td>
<td>.000001</td>
<td>60</td>
</tr>
<tr>
<td>Light traffic from 100 ft</td>
<td>.0000001</td>
<td>50</td>
</tr>
<tr>
<td>Soft conversation</td>
<td>.0000001</td>
<td>40</td>
</tr>
<tr>
<td>Whisper from 5 ft</td>
<td>.00000001</td>
<td>30</td>
</tr>
<tr>
<td>Average household silence</td>
<td>.000000001</td>
<td>20</td>
</tr>
<tr>
<td>Breathing</td>
<td>.00000000001</td>
<td>10</td>
</tr>
<tr>
<td>Threshold of hearing in young</td>
<td>.00000000001</td>
<td>0</td>
</tr>
</tbody>
</table>
II. SOUND DESK

A. Introduction to Mixing Board
1. When we understand sound it takes some of the mystery about the scales and reference numbers on the sound board face. The basic sound board is divided into sections to control both the input and the output. The cost of the mixing board generally is proportional to the number of channels, the capabilities of the sound board and the quality of the components used in the board.

B. Channels
Channels are the vertical columns on the sound board. There may seem to be a huge amount of buttons and dials available, however, each column represents one channel and each channel consists of the same functions from ten (10) dials, the learning curve is drastically reduced. Almost every sound board consists of the same basic items on every channel.

The trim/gain control is on the top of each channel on most boards. This control regulates the incoming sound. Adjusting the control too high results in clipping or distortion. Adjusting the control too low produces unnecessary noise in the system, i.e., a hiss or buzz.

Equalization and auxiliary assignments are beneath the trim/gain control on the channel. controls allow selection of either the left or right channel for output of the signal. A fader is the slider used to set the output level of each channel measured in decibels. There may also be buttons for output assignments (the port that will be used on the sound board for output), solo controls (these allow us to select one channel to listen to), and mute controls (where we can completely shut off a channel).

Channels are generally either mono or stereo. Mono channels are more prominent. They represent singular instances of sound. These include vocals, single instruments or mono playback devices. Stereo channels are reserved for stereo playback equipment or instrument like keyboards. These ports are used with pan controls to produce a sound design.

1. Stereo channels are more prominent on recording studio sound boards.
2. Channels are generally labeled numerically; mono channels are denoted with one number, stereo channels with two numbers, e.g. 14/15.
3. Channels may contain multiple port types including XLR, quarter inch (¼”), or RCA.
4. Number of channels vary and more expensive boards usually have more channels.
C. Equalization

1. Two topics regarding equalization are channel equalization and full range room equalization. Channel equalization is done at the sound mixing board and has limited ranges. Most channel equalization is geared more for the vocal range.
   1) Room equalization is accomplished through a full range graphic equalizer.
   2) Graphic equalizers breakdown the hearable range of audio into bands.
   3) Graphic equalizers vary in the number of bands.
   4) Graphic equalizers are referred to by number of bands or by octave spacing, e.g. ⅓ or ⅔.
   5) The bands are split by ⅓ of an octave, e.g., 3 bands for every octave. A ⅔ octave has only 2 bands for every octave.

2. Channel equalization is generally broken down into 3 ranges:
   **Low Range** is sometimes referred to as bass.
   **Mid-Range** embraces the heart of sound, particularly of voices.
   **High Range** represents the upper treble sound.
   Low, mid, and high affect sources differently. E.g. a bass guitar does not have much high range and will not be affected much by high EQ.

3. Channel equalization is said to be **parametric** if utilized in addition to a dial for lowering or heightening (measured in decibels). There is also a dial or knob for setting frequency. Sometimes mid-range is split again into upper mid-range and lower mid-range. These both can be parametric.
   Parametric equalization actuates by moving the slider across the frequency range to find the spot that affects the input source the most. Think of it as centering on the middle of a sound.
   Most vocal ranges live in the 1 kHz – 4 kHz range.

4. The last type of channel equalization is not a dial or knob, but may not be available on all mixers. It is the high and low pass filter. These are preset specific to the mixer. It is set to allow for dropping undesirable frequencies.
   **The Low Pass Filter:**
   1) Cuts out thumping or thudding sounds;
   2) Extremely useful for vocals;
   3) Examples of these sounds: breathing into mic, spitting in mic, bumping mic against an instrument, mouth or music stand;
   4) Not applicable to certain instruments, i.e., bass guitars with a distinct lower bass sound.
   **The High Pass Filter:**
   5) Cuts out noises such as whine, hiss, metallic ping;
   6) Not applicable to certain instruments or sound sources which are high in treble;
### SOME CRUCIAL EQ BANDS & WHAT THEY SOUND LIKE

<table>
<thead>
<tr>
<th>Frequency Range</th>
<th>Description</th>
</tr>
</thead>
</table>
| **50-60 Hz**    | - Thump in a kick drum  
|                 | - Boom in a bassline  
|                 | - Essential in dubstep & reggae!  
|                 | - Too much: flapping speakers & flabby mix  
|                 | - Too little: weight or depth  |
| **100-200 Hz**  | - Adds punch in a snare  
|                 | - Gives richness to anything  
|                 | - Too much: boomy/woolly  
|                 | - Too little: thin & cold  |
| **200-500 Hz**  | - Crucial for warmth & weight in guitars, piano & vocals  
|                 | - Too much: muddy/congested  
|                 | - Too little: thin & weak audio  |
| **500-1000 Hz** | - One of the trickiest areas  
|                 | - Provides body to instruments  
|                 | - Too much: hollowness  
|                 | - Too little: thin & harsh  |
| **2 kHz**       | - Gives edge/bite to guitars/vocals  
|                 | - Adds aggression & clarity  
|                 | - Too much: painful!  
|                 | - Too little: soft or muted  |
| **5-10 kHz**    | - Adds clarity, openness & life  
|                 | - Important for top end of drums  
|                 | - Too much: gritty or scratchy  
|                 | - Too little: presence & energy  |
| **16 kHz**      | - Can add air, space or sparkle  
|                 | - Almost too high to hear  
|                 | - Too much: artificial or fizzy  
|                 | - Too little: dull & stifled  |

### D. Auxiliaries

1. Auxiliaries allow creation of separate channel outputs for different purposes. Common uses of auxiliaries are: producing mixes for monitor reinforcement of vocals, producing recording mixes, outputs for control rooms, or as an output for other areas of facility. Auxiliaries are assigned to output ports on the sound board and bypass the main mix. Any channel combination can be used and each channel can set the gain strength. (This is extremely important when creating monitor mixes because the lead vocal or the main instrument may need to become louder than other audio sources.) Auxiliaries provide a button for post-fader or pre-fader. Post-fader affects the auxiliary if the slider is positioned at the bottom of the channel. This is useful when lowering of gain should also result in lowering gain for auxiliary output. Alternately, pre-fader should be positioned to prohibit any effect on the mix when an audio source is lowered.

### E. Assignment

Most sound boards, especially those with groups, provide buttons for assignments.
1. These assignment buttons will have labels such as: L, R, LR, 1, 2, 3, 4, etc. This allows each channel to be associated with an output port.

2. Assign L & R to main house mix. Assign numbers to a group.

3. Assign the L, R or LR buttons to the Main/House output. Assign the channel to the group and the group to the Main/House.

4. The Main/House and group should not be assigned to the same channel.

5. This is not to be confused with Pan Control which signals the board to send audio only on the right or left speakers.

F. Pre & Post Fader

Pre & Post Fader refer to whether the control is affected by the slider/fader when it is repositioned up or down. Pre or Post fader can refer to auxiliaries or other effects used on the board.

G. Mixing

Mixing is an art that requires both understanding and experience. Frequencies vary in strength therefore, channels should be programmed to output at varying gain level. The output for a strong soprano vocalist may overwhelm a group of altos if mixed together. Using the meters to blend at similar levels is helpful for vocalists with similar styles. The sound engineer must understand how to mix various instruments and vocals to produce a clear, quality representation of the final project.

1. Mix vocals separate from instruments. It is easier to blend vocals with each other without adding band and instruments.

2. Mix instruments together without vocals using the solo or PFL to single out individual channels.

3. Try to identify the lead vocalist, speaker, or instrument. This should be a more prominent section of the total audio output.

4. For supplement, backup, or complementary audio sources, ensure the output is set to the appropriate levels. Oftentimes, instruments with heavy sound drowns out these audio sources.

5. Grouping assists the mixing process by allowing gain to be raised or lowered quickly on a group of similar instruments or vocals.

6. Separate stage or monitor output from the Main/House mix.

7. Compare the final mix output on the monitors and the Main/House.
   If the monitor/stage audio is too loud, it may result in muddied Main/House output.
   In-ear monitors eliminate floor/hotspot speakers from interfering with main/house audio.
   Try not to repurpose retired main/house speakers for floor monitors. If they were unfit to use as Main/House speakers, they are probably not suitable as stage monitors either. The final audio mix is only as good as its weakest link.

H. Putting It All Together

The planning stages of creating a mix is imperative for successful audio practices.

1. A layout of utilized instruments and vocals creates a dynamically blended mix.
   Running all sounds at equal levels through both the left and right side speakers may produce a monotone mix. Instead, send some output to the area where they are produced. E.g. if the trumpet
player stands right stage, set the pan control to 70%/30% toward the right. This does not eliminate others from hearing the trumpet, but does give a truer representation of the actual sound. Different patterns of sound mix reveals little nuances that make audio seem more dynamic. Instead of sending all sounds through two (2) speakers, utilize multiple speakers located throughout the room and send different mixes to them. It will produce a unique sound. Utilize sound reinforcement. Place smaller speakers, toward the back of the room and output to them at lower levels. This helps level volume across the room. Room preparation/consideration is important when planning audio mixes.

1) Room size has a great impact on audio output.
   a) Does the room need sound baffling?
   b) Are hard surface elements, i.e., cement, present?
   c) Does it create an echo?
   d) Do certain corners of the room trap the audio and produce standing waves?

All of these things need to be taken into consideration to produce a quality sound mix and a successful audio presentation.
III. MICROPHONES

A. Wired vs Wireless

Wired microphones have been the mainstay of audio systems for many years. The thought that wireless microphones are newer, therefore must be better, isn’t necessarily correct. Newer is not always better.

1. Wireless

Decisions to purchase wireless microphones are usually based on convenience not quality. Wireless microphones interpret a sound, convert it into a digital signal, transmit through the air, and reinterpret as an analog sound wave. Because of this intricate process, there are more points of failure with wireless systems. Cheap wireless equipment usually does not accomplish a successful analysis of the sound source. Because there is a limit to how much digital signal can be sent from a wireless device to its receiver, a subrange of frequencies and certain amount of compression is used. This signal, while good, may not have the same breadth of signal a wired microphone produces. Wireless microphones vary in price because of the quality of the transmitter and receiver. Wireless microphones can transmit at different frequency bands. A lower cost VHF wireless mic uses VHF frequency, but may be prone to interference. UHF wireless equipment may overcome some problems, but are generally more expensive. High end wireless equipment may use proprietary frequencies, use better quality components and have fewer issues; however, the cost is significantly more. Compare frequency specs of mics to determine the right mic for a particular audio source.

2. Wired

A wired microphone contains the capability of carrying a clear broad signal over a distance. However, they are susceptible to bad cables, frayed ends or bad solder connections. Because the wires move as vocalists move, a constant amount of wear and tear is definite. Consider the needs of the instrument or source when purchasing equipment. Getting the right mic that will best represent and carry the audio signal of the source is more important than wired vs wireless, unless the audio source must be able to roam the room. No matter wired or wireless, quality of the microphone is of the utmost importance. Comprehensive research is necessary to determine what type and style is best for the audio source.

3. Condenser Microphones

These are usually smaller microphones with wide patterns of input or larger specialty microphones used for studio recording. A condenser microphone may have 180° (degrees) of acceptable placement of audio sources compared to the microphone placement. This is especially important for podium mics, choir mics or stage mics used for plays or performances. They are usually very sensitive mics and require either a battery or phantom power in order to work.
1) Phantom power is supplied from the sound board and can be activated per channel.

4. Always check the pattern of allowable inputs as well as the frequency range of condenser mics to determine if they are appropriate for use. An associate at a good audio store could assist with the quality of microphones before a purchase is made.

B. Types and Uses

There are many types of microphones, and even subtypes that are designed for specific audio needs. Below are a few of the most common.

1. Handheld
   These are used primarily for vocalists or situations where multiple people use a mic. Some subtypes of microphones provide ranges more suitable to a female voice than to a male voice, e.g., the Shure SM58 for males and the Shure SM57 for females.

2. Lapel Microphones
   Often wireless, lapel mics are designed for mobility and general speaker needs. Placement on the speaker or vocalist are very important because these mics tend to catch small noises in the near proximity. While sensitive, lapels sometimes struggle producing a clear signal without being muffled by clothing, hands, or placement. Due to indirect input, this type of mic suffers from volume increases and decreases depending on the direction or angle it is held.

3. Boundary Microphones
   These are usually placed, or adhered to, a solid surface such as a podium or the floor. They pick up a wide range of sound sources without being completely visible to the audience. Sounds generated close to them are much louder than sounds from a distance. They are also sensitive to vibrations and should be placed on firm surfaces.

4. Condenser Microphones
   Condenser microphones are specialty microphones for specific instruments or choir mics.

5. Headset or In-Ear Microphones
   By placing this microphone close to the speaker or vocalist, these mics overcome issues that lapels encounter. These are used when visibility of the microphone is a concern. They are usually wireless and cost more than lapels. Cheap headsets can often be found and are convenient for plays or programs.

Each of these types will also have some basic characteristics through the microphone element (the part of the microphone that picks up the sound goes into).
6. Unidirectional or Cardioid
These pickup patterns are most sensitive to sound produced on the front side of the microphone capsule. Super-cardioid pickup patterns have a greater sensitivity than cardioid pickup patterns.

7. Bidirectional
Bidirectional pickup patterns are sensitive to signals emanating from the front and back sides of the microphone capsule while rejecting sounds from the left and right sides of the mic capsule.

8. Omnidirectional or Boundary
These pickup patterns are sensitive to sound from all directions of the microphone capsule.

9. Switchable Microphones
Switchable pickup pattern mics are hybrids that can be switched from one pickup pattern to another for all-in-one flexibility in different environments.
IV. GAIN STRUCTURE

Gain structure is one of the most critical and overlooked areas of sound mixing. Gain is not volume but translates directly to the ability to set volume. It affects the quality of the sound setup tremendously. There are several places the gain structure is set up which makes it one of the misunderstood areas of sound mixing.

A. Proper Gain Structure

Gain structure captures the input source at an adequate level to eliminate clipping, yet provides a strong signal with which to work. Capturing to strong produces clipping while too low produces unwanted noise in the signal. Channel gain structure should allow us to raise the gain in using the heart of the fader or slider. Grouping gain structure if used should allow for proper positioning of groups of channels into the full mix. Main or house gain structure should provide good range of volume for the entire mix while fully utilizing the abilities of the sound board. Room gain structure is accomplished with amplifiers or powered speakers to appropriately give us the most flexibility for the room. Failure on any one part of gain structure ultimately flaws all other areas resulting in low quality sound mixing.

1. Source gain structure is dependent upon several factors:
   Source type
   Microphone used to capture source;
   Input level for direct attached instruments.

2. Proper Source Gain Structure Process

The process for setting proper source gain structure is:
   1) Set the gain level of each channel.
   2) Slide all other channel faders down or mute them.
   3) Place the fader and the group/main level at -3 dB (decibels).
   4) Start the audio source playback or live talking/singing.
   5) Increase the gain control until the meter on the audio level bounces just above the 0 dB level. The goal is to capture the sound at a processing level close to the actual gain of the audio source without raising or lowering it.

Repeat this process for each channel. Be sure to mute the other channels as gain is added to each audio source.

B. Troubleshooting Gain Structure

Gain structure has the potential to present various problems. The following are simple steps to troubleshooting some of those issues.

1. Audio source is too low.
   Ensure closeness of the audio source to the microphone.
   If utilizing a direct input, turn the sending level up.

2. Audio source is too high. (This is sometimes referred to as hot.)
   Move microphone away from the audio source.
   Lower the input level.
   Padding may also be implemented to assist in the adjustment.
3. Audio source is picking up multiple audio inputs.
   Move microphones away from each other.
   Lower input levels of audio source.
   Move audio sources away from microphones.
   Implement audio containment booths or sound baffling.
      1) Drums are notorious for overpowering audio and being captured by multiple microphones. Drum houses made of sound baffling or Plexiglas isolates the instrument producing a more controlled environment for output.
      2) Strategically placed sound mats, booths or sound proofing reduces the potential of unwanted audio conditions.
4. Audio source produces a buzz.
   Generally this is caused by direct input devices. The buzz is actually an electrical interference sometimes referred to as a 60 Hz buzz.
   Run input cables through a direct input (DI) box, hum eliminator or other small audio box to eliminate unwanted noise. As a rule, guitars and keyboards should use these when connected to a sound system.
5. Feedback.
   Feedback appears when the output, generally from speakers, sends through the microphone and produces a loop of sound that escalates quickly into a squeal. Ensure that speakers face away from microphones or are placed at a distance that is not in range of the microphones.

C. Fader/Slider Gain Control
   This control is usually located toward the bottom of the channel controls and uses a graduated scale of values. The values around the 0 dB are farther apart than the values at the lower end. This provides greater fine tuning adjustments in the most used area of the fader. When the input gain control is set incorrectly, it becomes difficult to control the gain for the channel. Gain input set too low creates a difficult situation in which to raise the fader, therefore faders should always be set all the way up. Gain input set too high results in really low faders. Little movement of faders produces huge swings in gain.

D. Group Gain Control
   This convenient feature is built in to some boards and allows grouping of like channels, providing ability to raise one or the other resulting in consistency of levels per group. Examples include:
      1. Placement of all vocals or band instruments into one group;
      2. A selection of mics, such as choir mics or a set of drum mics;

   Group gains tend to replace main mix gain control because groups are usually output through a separate set of ports. Groups can sometimes be assigned back into the main mix on some boards to serve as an additional gain control.

E. Main/House Gain Control
   This control produces the final mix before the sound is output to the amplifier/speaker system. Ultimately, the amplifiers and speakers produce the audio and volume. The goal is to ensure the amplifiers receive quality mixes of varying gain audio sources. Improper setup leads to underutilization of capabilities of the sound board, potentially causing degraded the sound quality of the input.
F. Amplifier Gain Structure

This is measured in watts of power. The larger the watts, the greater amount of decibel gain we can give to the signal.

1. Proper purchasing of amplifiers can be accomplished by buying amps large enough to power the speakers (which also have an acceptable range of watts) while not over powering the speaker, indicated as the peak level.

2. Amplifiers should not be set so that the maximum output exceeds the peak levels of the speakers. Example: A 2400 watt amplifier with 2 outputs each rated at 1200 watts should not be set much over ¾ volume (measured in decibels of gain) for a 900 watt peak level speaker. Permanent speaker damage can occur to overdriving speakers. It is important to note that trying to daisy chain varying speakers, with different peak levels, can be difficult. The lowest peak level should be observed to not damage the speaker but may result in not enough signal being given to large speakers.
   1) Multiple amps or outputs should be used to separate different groups of speakers. E.g. an amp for monitors and an amp for house speakers.

G. Room Gain Structure

Room gain structure must be taken into account when positioning and setting amplifiers. Try not to encase speakers or move into tight corners. This is often a mistake by some to try and hide speakers, thereby making the sound system ineffective.
V. SPEAKERS

Speakers are an integral part of the sound system. They recreate the original input source and the quality (good or bad) of everything processed through the other components. Ironically, they are sometimes overlooked, damaged, or partially working. Because of their size and appearance, they are sometimes hidden or placed in improper locations for quality audio settings.

A. Speakers are constructed of the following:
1. The case or box;
2. The crossover
   The circuit board on a speaker.
3. The driver
   The magnet on the back end of the speaker.
4. The basket
   The front end of the speaker that flares out and is made of various materials including some plastics and papers.

B. Powered vs Unpowered Speakers

Speakers are either powered or unpowered and it’s easy to determine the speaker-type. Is there a power cord plugged into the speaker? If so, they are powered speakers. Alternatively, unpowered speakers have only an audio cord attached to them and are connected to an amplifier. The amplifier is placed between the sound board and the speaker providing power to drive the speaker. Powered speakers do not need amplifiers, which is a huge benefit. The soundboard can be attached to the speaker directly or, in some cases, a microphone can be attached to the speaker as a PA system.

The disadvantage of powered speakers is the need to connect a power supply to them. This could result in an abundance of cords on the floor and a need for multiple outlets. Unpowered speakers do not need power and separate the amplifier from the speaker. This allows each component to be replaced separately. Unpowered speakers can be less expensive if amplifiers are already in place.

C. Full Range Speakers

These produce the full range of frequencies in one speaker. While useful in general practice, they do not represent the high or low tones as well. A graphical band equalizer balance this by brightening or raising the desired tones.

Some speakers are known as full range when the box actually contains more than one speaker. A crossover circuit board, located in the back of the box, splits the signal. It sends the highs to the horn, which looks similar to the bell of a trumpet. It sends the lows to a mid-range or sub style speaker, which is a round-style speaker located toward the bottom of the box.

D. Subwoofers and Box Speakers

These speakers move a tremendous amount of air but are not considered very loud. Also included in the subwoofer category are box speakers that are paired with a subwoofer speaker and tuned to low frequencies.
An SPL sound meter detects the amount of decibels that a subwoofer exudes. Overpowering subwoofers produces low sound waves that affect our hearing and also vibrate strongly through skin and body. Headaches, nausea and blurred vision are symptoms of standing too close to subwoofers, especially the types utilized at concerts.

Low sounds are difficult to determine directionally, therefore, they do not need to be centered in the room. They can even be concealed due to the fact that the long wave tends to be less resistant to dampening.

E. Monitors

Monitors are designed to support musicians, vocalists or presenters. They are placed on stage or in the vicinity of them to assist in hearing the surrounding audio above themselves.

F. In-Ear Monitor

In an effort to reduce the amount of audio output on stage due to floor monitors, etc., earbuds are used for sound reinforcement. Utilization of in-ear monitors allows a cleaner look to the staging area. It lessens the amount of cabling, etc. that runs along the floor.

Utilization of in-ear monitors lessens the chance of feedback caused by mics being positioned too closely to monitors.

Different systems include wired and wireless. They often provide the ability to utilize independent mix control and volume.

G. Hotspot Speakers

These speakers are usually placed on stands that can be positioned in close proximity to the vocalist or musician. These help reduce the amount of sound on the stage by bringing the speaker closer.

H. Floor Monitors

While these are common, they tend to give the most opportunity for feedback. An advantage of floor speakers is that they can cover sound reinforcement for many audio sources. However, the ability to give individual mixes is gone as each floor monitor at most can have only one mix.

I. Monitor Placement

Ensuring great sound in every room does not come with a magic formula. However, understanding and identifying basic acoustical principles will strengthen the ability to create a successful audio experience. Proper placement of the loudspeaker system drastically improves the quality of the output. Upgrading speakers improves output range as much as 1 to 5 decibels and simply changing a speaker position can make a difference in excess of 15dB in response!

1. Tools of the Trade

Utilization of some basic equipment can reveal crucial information regarding the audio set.

A 20’ measuring tape;
a test CD with a variety of test tones;
an inexpensive analog SPL (sound pressure level) meter;
a calculator;

J. Placement and Room Peripherals

1. Frequency Response

Similar to a speaker, every room has its own frequency response. Adding to this complexity, the response varies according to:
the listener location;
the room dimensions;
room construction;
room furnishings;

2. Room Dimensions

Room dimensions determine standing wave frequencies. In general, rooms with dimensions divisible by a common factor, i.e., 10' x 20' x 30', tend to compound standing waves at one frequency. Room dimensions with non-equal or divisible dimensions produces a higher quality of audio output. Vaulted ceilings, non-parallel walls and irregular surfaces help reduce slap echoes. However, they have little effect on low frequency standing waves. Room construction affects bass reinforcement, floor noise, and adjacent room noise.

A wall constructed with average drywall material resonates around 70Hz. Doors rattle, windows sing, and air vents “whoosh”. Use a test CD or tone generator and play a sweep tone. The difference heard during the sweep is almost entirely due to room coloration.

Signals that have bounced off walls, ceiling and floors, mix with the original direct signal and are called early or “first” reflections.

Listening tests have shown that when multiple reflections are received within 20 milliseconds of the direct sound, they are perceived as part of the original. This alters the tonal balance and confuses vocals and dialog.

Sound panels are available from various acoustical material suppliers. They may be obtained in a variety of fabric and finish options in order to blend with or complement most interior schemes.

If purchasing professional sound panels is cost-prohibitive, consider a DIY sound panel. Using compressed fiberglass (Owens-Corning #703) covered with fabric creates an attractive homemade alternative to a store purchase. For improved low frequency effectiveness, use 2” thick panels and stand them away from the wall a bit, or use thicker material.

Slap echoes are reflections that bounce back and forth between bare parallel walls. They are easily identified by clapping hands and listening for a ringing tone. Continue clapping and move from the middle of the room towards one end. Notice that the slap echo pitch and ring duration will change. This relates to the different round trip distances the sound travels as the wave leaves the clapping hands, heads off in different directions, bounces off the front and back walls, and returns back by its inception.

Some common methods of treating slap echoes are the “live end, dead end” scheme and the “dead end, live end” scheme.
Both methods involve treating one end of the listening room, leaving the other end “live” for a natural room ambience.

It is preferable to diffuse slap echoes with diffuser panels. Slap echoes may also be absorbed with carpets, fiberglass panels or drapes. When treating sidewall slap echoes near the sides of the
loudspeakers and/or listeners, it is desirable to treat both walls evenly, left and right, to provide a balanced sound field.

When treating reflections and echoes, best results are obtained from a proper mix of direct and diffused sound—a balance of diffusive and absorptive materials strategically placed throughout the room. While trying all types of room treatments, the key to finding what works best is to utilize test equipment which is designed to measure the time, energy and frequency relationship within the room.

Standing waves are high and low pressure energy buildup, which are determined by frequency and room dimension. They are appropriately named because they do not travel or propagate. Instead, they anchor at various spots in a room determined by boundary conditions.

Although standing waves occur at all audible frequencies in a contained space, the widely spaced, low frequency waves cause severe peaks and dips in the system’s in-room bass response. This creates the dreaded “one-note bass” while obscuring a truly deep bass. All rooms (except very large rooms whose wavelengths are so low in frequency that they can be ignored) have low frequency standing waves of consequence.

There is no quick fix to standing waves. Play a continuous low tone and walk throughout the room. Notice the soft and loud spots. Sound baffling will not repair this problem because low frequencies tend to pass through instead of becoming absorbed.

Speaker placement is one of the few options to attempt to overcome this problem. Speakers placed in a corner will create the greatest number of room standing waves. Speakers placed in locations away from walls or hard surfaces will create the least number of standing waves. Listeners will experience the smoothest response when speakers and the listening location are placed away from standing waves.

Another weapon in the standing wave knowledge arsenal, is the conservative use of equalizers to help flatten system response. Equalizers are not a cure-all. They are yet another potential tool when used properly. Always exercise caution, because equalizing a particular frequency from one listening position can cause a severe dip or boost in another. This will severely alter the overall frequency balance. The potential also exists to blow up amplifiers and woofers if attempting to apply excessive bass boost to compensate for a low cancellation point.

Before applying any equalization, invest time in determining the best physical placement. A qualified “sound consultant” will have the necessary test equipment to collect level measurements from several listening positions and average the results by frequency before making adjustments. In short, this may be the time to call a professional for assistance.

K. Amplifiers

Amps are often overlooked but are an important part of the sound system. Many audio enthusiasts simply go for the largest amp they can afford. The thought behind this rationale is that there will be no boundaries
or limits to the setup. In part, amplifiers and speakers work together and therefore, must be considered when purchasing one or the other.

1. What do I need?

This is a somewhat complicated question. There are many variables to consider before making a purchase and/or setting up the sound system.

The **ROOM SIZE** is probably one of the more clear-cut variables in selecting amps. Obviously, it takes more power to send a sound wave across a larger distance. Please note: distinguishable volume is different for listeners in closer proximity to the speakers. For this reason speakers are sometimes directed toward the center of the room or are raised above the height of listeners located in the front.

**SPEAKER WATTAGE** is specified by 2 ratings: RMS and Peak watts. It is important to note that watts do not necessarily mean volume. The RMS rating refers to the endurance of the speaker when given a continuous feed. Pink noise (the sum of all frequencies) is sometimes used to test speakers. This is considered a torture test and caution should be used to prevent damage to the equipment. Pink noise at an RMS rating (in watts) should be sustainable by the speaker, especially understanding that live performance is not as brutal as continuous pink noise. It stands to reason that speakers handle more watts than the RMS rating. The Peak rating, therefore, is a higher rating stating that beyond this point damage can occur to the speaker. While it is possible to produce a signal at higher watts that does no damage this is the manufacturers recommended threshold.

The **SENSITIVITY RATING** of a speaker is measured in decibels. A very basic formula for this measurement is 1 watt at 1 meter from the speaker. The higher the number the less watts the speaker needs to produce a given decibel rating. A low number means the speaker needs for wattage. Different types of speakers have different ranges and a high sensitivity does not mean the speaker is better quality. The one true statement is that a low sensitivity speaker will need a lot more watts for higher decibel levels.

**DESIRED DECIBEL LEVEL** is actually a measure that leadership and the audio team need to agree upon. While not the norm, it is good practice to agree upon a decibel level for music performance and general speaking. Coupling that with a desired range and peak limit gives a baseline for consistency in all events.

**SPEAKER HEADROOM** is the measurement of flexibility available to raise the volume with the amplifier without peaking the speakers. A common amount of headroom is 3 decibels because it represents the volume of sound doubled. This does not mean that the amplifier can’t go past a 3 decibel increase. It is rather a measure of the speaker.
Crown, a maker of audio system amplifiers, supplies a calculating tool to determine watts of amplifier needed according to specifications entered.

The illustration demonstrates that the decibel level of 83 at a distance of 20 meters (65') with standard headroom and sensitivity would require approximately 1000 watts per speaker. Therefore, a 2500 watt amplifier is needed to operate 2 speakers for main/house sound. Keep in mind, a second amplifier is needed for monitor mixes. In this case, the speaker chosen would probably have the following specifications: 600 Watts RMS (continuous) / 1000 Watts Program / 1200 Watts Peak. If the above specifications for the speakers were only half of the numbers, 4 speakers are necessary instead of 2 to accomplish the goal.

L. **Cabling**

This is an important part of the sound system. Clean, quality connections ensure that all the hard work and expensive equipment is experienced at the optimal level. Minimizing splices and connectors should always be a goal as well as purchasing quality cables.

1. **SNAKES** are used to bundle wires together into a convenient package and to protect them from interference from outside signals. Snakes are usually created with shielding and built in reinforcement to ensure that the cable itself is not strained nor compromised. They come in many shapes, sizes, and lengths. The longer the snake, the heavier the gauge of wire is needed. This also translates into a thicker and heavier weight cable.

   They can be purchased in increments of the correct number of channels or cords bundled together, i.e., a 16 channel snake would have 16 cords bundled together to connect 16 channels of a sound board to 16 audio devices. Remember: stereo devices (LR) may need 2 channels from the snake. Snakes also may have sends or returns bundled with the number of channels. A snake that has 16 channels and 4 returns/sends would be 16 channels of input and 4 cords for sending audio to the other end of the snake. A typical setup for audio is to place the mixing board in the back of the room and the stage is positioned in the front. If 16 microphones are on the stage, one end of the snake contains connections to the mics. The snake would lead back to the mixing board and plug into it. This setup allows for the 4 returns/sends to connect to the left and right of the main/house output, plus the left and right output of an auxiliary back to the stage. The stage end of the snake uses the 4 returns and connects into the amplifiers or directly into the speakers if they are powered.
2. There are 2 types of ¼ INCH CABLES (TRS), sometimes called 6.5mm; mono and stereo. They may both be the same length and shape but the barrel of the connectors are slightly different. The tip of red connector into 3 parts: tip, ring, and sleeve. The ends of XLR cables snap together and are very sturdy even when moved and used often. For this reason they have been the standard for microphones for many years.

3. XLR CABLES are commonly used for microphones. They are called balanced because they have 3 pins that provide a positive (+) wire, a negative (−) wire, and a ground or shield. Most sound boards provide inputs for ¼" and XLR cables. The ends of XLR cables are color coded to identify left or right audio channels. Generally, the color coding is white/black or red/black. Occasionally, it will provide a yellow end which signifies that a video cable is bundled with audio RCA cables. Regardless of color, they are identical and have no difference in specifications. They are also provided in different wire gauges. Lower gauge wires are thicker than their higher gauge counterparts. Each cable has a center pin and outer shell, therefore, they are not balanced cables.

4. RCA CABLES are common with consumer audio equipment as well as playback audio devices. The ends of RCA cables are color coded to identify left or right audio channels. Generally, the color coding is white/black or red/black. Occasionally, it will provide a yellow end which signifies that a video cable is bundled with audio RCA cables. Regardless of color, they are identical and have no difference in specifications. They are also provided in different wire gauges. Lower gauge wires are thicker than their higher gauge counterparts. Each cable has a center pin and outer shell, therefore, they are not balanced cables.

M. Other Equipment

1. Equalizers

Equalization, or EQ for short, means boosting or reducing (attenuating) the levels of different frequencies in a signal.

The most basic type of equalization familiar to most people is the treble/bass control on home audio equipment. The treble control adjusts high frequencies, the bass control adjusts low frequencies. This is adequate for very rudimentary adjustments — it only provides two controls for the entire frequency spectrum, so each control adjusts a fairly wide range of frequencies. Advanced equalization systems provide a fine level of frequency control. The key is to adjust a narrower range of frequencies without affecting neighboring frequencies. Equalization is most commonly used to correct signals which sound unnatural. For example, if a sound was recorded in a room which accentuates high frequencies, an equalizer reduces those frequencies to a more normal level. Equalization can also be used for applications such as making sounds more intelligible and reducing feedback. There are several common types of equalization.

Shelving EQ

All frequencies above or below a certain point are boosted or attenuated to the same amount. This creates a “shelf” in the frequency spectrum.

Bell EQ
This boosts or attenuates a range of frequencies centered on a certain point. The specified point is affected the most, frequencies further from the point are affected less.

Graphic EQ
Graphic equalizers provide a very intuitive way to work — separate slider controls for different frequencies are laid out in a way which represents the frequency spectrum. Each slider adjusts one frequency band. Therefore, the more available sliders, the more extensive the control availability.

Parametric EQ
Parametric equalizers use bell equalization, usually with knobs for different frequencies. They have the significant advantage of the ability to select which frequency is being adjusted. Sound mixing consoles and some amplifier units such as guitar amps, small PA amps, etc., contain Parametric EQ’s.

2. Sound/Noise Gates
These are used on audio sources that have a set range of frequencies. Because of this, a compressor is a better choice to use on vocal audio sources. A Sound/Noise Gate will generally have 3 settings:

THRESHOLD setting determines what is allowed into a gate. When a drum set utilizes multiple mics to pick up each drum, the threshold desired is allowing each mic to only pick up one drum. To accomplish this, dial the threshold down until only the desired drum is heard. Caution must be taken to control the aggressiveness of the threshold or the audio will sound “choked” or flat.

The RATIO selects how aggressively the gate will cut off the extra sound or noise. Setting an aggressive ratio can cause a “choppy” output. To prohibit an obtrusive shut down of the frequency, the ratio decides how quickly the excess sound tapers off.

The OUTPUT GAIN allows the output to compensate for any loss of strength from the audio source due to the lack of full signal input.

Sound gates help reduce feedback because it causes the mic to appear less sensitive to extraneous sound. Ultimately, the concept is to receive only the desired sound to process through the sound board while discarding the rest.

3. Compressors
Compressors allow usage of the heart of the desired frequencies, while compressing the rest into a smaller portion of the signal. Consider a male vocalist where the bulk of vocal frequencies are between 1500 hz and 2500 hz. The desired sound is in the middle (heart of the sound), and is the range needed for successful audio. Instead of pulling it out, rather it compresses the frequencies outside of the range.

Setting the threshold indicates where the compression will begin.

The ratio sets how much compression should occur. A ratio of 2:1 will reduce every 2 decibels to 1 decibel. Setting a very high ratio is effectively creating a limiter, where nearly all sound is compressed into very little signal.
The attack dial (optional) allows for setting how aggressively the frequencies outside the threshold are compressed. The release dial (optional) allows for setting how aggressively the compression is released. Because live sound is rarely monotone, compressors are constantly evaluating, based on the settings, how much and how fast to compress the sound. Overly aggressive compression reduces the fullness of the sound while little compression remits little improvement. Recorded audio is highly compressed. Compression improves the quality of sound but should be approached with caution. Improper settings can alter the audio source and make it feel less “bright”.

4. Limiters and Normalizers

These tools help control the signal strength. Utilize these to bring up low gain signals and reduce high gain bursts. These units contain the signal within the preferred gain range.

5. Reverb Units

By adding effects to the input signal, reverb units alter the audio signal. Common effects from reverb units may include:

- Echo – a hard repeat with sustain
- Delay – a replication of frequencies at lower strength, delayed a set period of time usually measure in milliseconds.
- Chorus multiple replications of frequencies with delay and sustain added;
- Reverb (high, medium, low) - the sustained effect of sound appearing to bounce in a large room;

This is just a sample of reverb effects. They may be used alone or in combination with each other.

6. Recording Units

Utilize recording units to capture processed sound. These units generally attach to the sound board through a port that is not amplified. Recording units can vary in media format. Cassettes, reel magnetic tape, CDs, Media cards, these are but a few of the different types. Most recording units are digital which means within the media type there may be multiple encoding styles for the files created. Common file types are MP3, WMA, AU, m4a, aac, among others.
7. Playback Units

This equipment allows for adding non-live audio sources such as CD’s, MP3’s, or other media formats into the audio mix. Common devices connected to audio systems are iPods, Computers, CD and Cassette players.
How do I conduct a sound check?

Mute all channels and turn the Main/House all the way down. Check each audio source, using headphones, with any stage monitors set at their operating/program level. This gives the truest sense of the audio input. This is also the time to adjust the amount of volume going to each audio source in the appropriate monitors.

Separate the band/instruments from the vocals. After all audio sources are done, allow the vocalists to sing together to ensure the blend of voices is in correct proportions. Then have all instruments play together and check the blend.

Now, mixing all audio sources together, determine if vocalists are at the correct level in comparison to the instruments. When that seems balanced, add the main/house support. If there seems to be very little difference, the stage mix may be too loud. Try bringing it down as a whole. There should be an obvious difference when adding in the main/house speakers.

Walk the room and check for evenness of the vocalists with the instruments. Also, walk on the stage during practice to check for loud areas or monitors that may cause feedback. After completing this, conduct a sound check with presenters or performers and mark any changes that need to occur during the program for the particular speakers. (See the section on gain controls for further information.)

How do I stop feedback?

The simple answer is prohibit the output sound from speakers from reaching audio source inputs. Move microphones away from speakers. Use isolation products to deaden noise, i.e., sound baffling, drum cages, floor deadening mats. Use sound/noise gates for instruments. Lower gain on problematic audio sources and raise the levels though channel faders/sliders. Move audio sources closer to mics.

Which Microphone is best?

Wireless devices seem convenient but are often over used. Quality microphones are much cheaper in the wired variety then in the wireless variety. Generally, the smaller the microphone the less dynamic range of frequencies are picked up. Using a lapel microphone may not provide the same fullness of sound as would a wireless handheld. Informed decision-making is key when purchasing microphones. A $20.00 microphone is probably not going to perform as well as a $150.00 microphone. Evaluate whether the microphone will be used by one person or by a variety of people. A microphone intended to receive audio from multiple vocalists at the same time will affect the microphone selection. The following are general guidelines:

Lapel mics are good for general presenters who need the flexibility to have hands-free mobility. These are generally not preferred by vocalists.

Headset or ear-set microphones work well for presenters who use the same headset every time. These can be customized to each individual and are more sensitive to damage.

Handhelds, wired or wireless, are the general purpose workhorses of audio setups. They work well in all general utilities and are somewhat durable. This type represents the lowest cost microphones, therefore, providing quality at the lowest price.
Boundary mics used for podiums or stages are a logical choice when trying to mic an area of sound. They are flexible for various voice types and are capable of picking up sounds at greater distances. However, they are also prone to feedback issues.
Cardioid/Choir mics successfully capture full frequency ranges from a distance and filter out unwanted frequencies. They come in varieties and are more costly for specialty microphones.

In addition to all these choices, each of them, in turn, have the options of unidirectional, bidirectional or omnidirectional.

**What is needed or not needed to set up an audio system?**

When it comes to sound systems, there is no limit to gadgets and effects that can be added. Before buying special effects or high end compressors/gates, invest in the basic parts of the sound system. Start with good quality microphones. Output is only as good as the quality of the input device. In turn, good speakers are equally as important because the sound quality will only be as good as the quality of the output appliance.

Cabling can destroy the sound package with loud snaps, crackles and pops. Use quality connectors that eliminate hums and buzzes. Potentially, this could also help protect the system from major damage. Finally, look at some of the larger components and determine what really is affordable. While reverb units, effects, or digital processors are nice, much can be accomplished with a good basic sound system that is correctly operated. Start with the basics and add the rest later, kind of like icing on the cake.

**Digital or Analog sound system?**

Within the last decade, digital sound systems have made great impacts into the sound industry. Because of its smaller size and lighter footprint, many sound companies, especially in the professional market, are switching. More capabilities, along with ease of use, are options that professional sound engineers appreciate. For lower end systems—churches, schools, and auditoriums—there are a few cost effective digital systems available. At a higher price, digital systems may be a bit overwhelming for some sound engineers. This may also be an opportunity to pick up quality analog sound boards at lower prices and professionals are upgrading. Ultimately, it usually comes down to budget constraints. While digital will encompass more and more of the market, there is still a large group of analog sound systems still used and purchased.

**How are amplifiers set?**

The key to amplifiers is to allow them to be turned up as much as possible to give the greatest flexibility of volume. However, amplifiers that are considerably larger than needed for speakers, run the risk of overdriving or damaging speakers. In this case, amplifiers should not be turned up more than the peak ratings of the speakers.

**Which is best to use, stereo or mono?**

Although mono is very popular in small venues, it’s really a question of whether there is enough equipment to run stereo. Stereo setup requires left and right outputs for each set of speakers. This may require more amplifiers. For temporary setups like outdoor events, mono may be an option for a PA system. If theaters offered a choice of Surround Sound or Mono, which would most likely be chosen? Why? This is the same answer needed to assist in giving the listeners the best experience possible.

**What is the process to check for blown speakers?**

Inspect speakers visually. Look for tears or holes in the basket. Does the speaker vibrate when music is being played? Does it make any sound at all? Speakers are relatively simple in construction and can be dismantled
easily. Unplug the speaker and open it. Examine the circuitry which is usually found in the back of speaker. Check the fuses. Replace them when necessary. Check for burnt marks, blackened colors or swelled electronics. These are often signs of damage. Many professional audio stores also offer their services to verify a broken speaker. They may be able to suggest if the speaker can be repaired or if it needs to be replaced. Also, keep in mind the age of the speaker. Old speakers produce lower and lower quality audio as their materials become more brittle and worn.