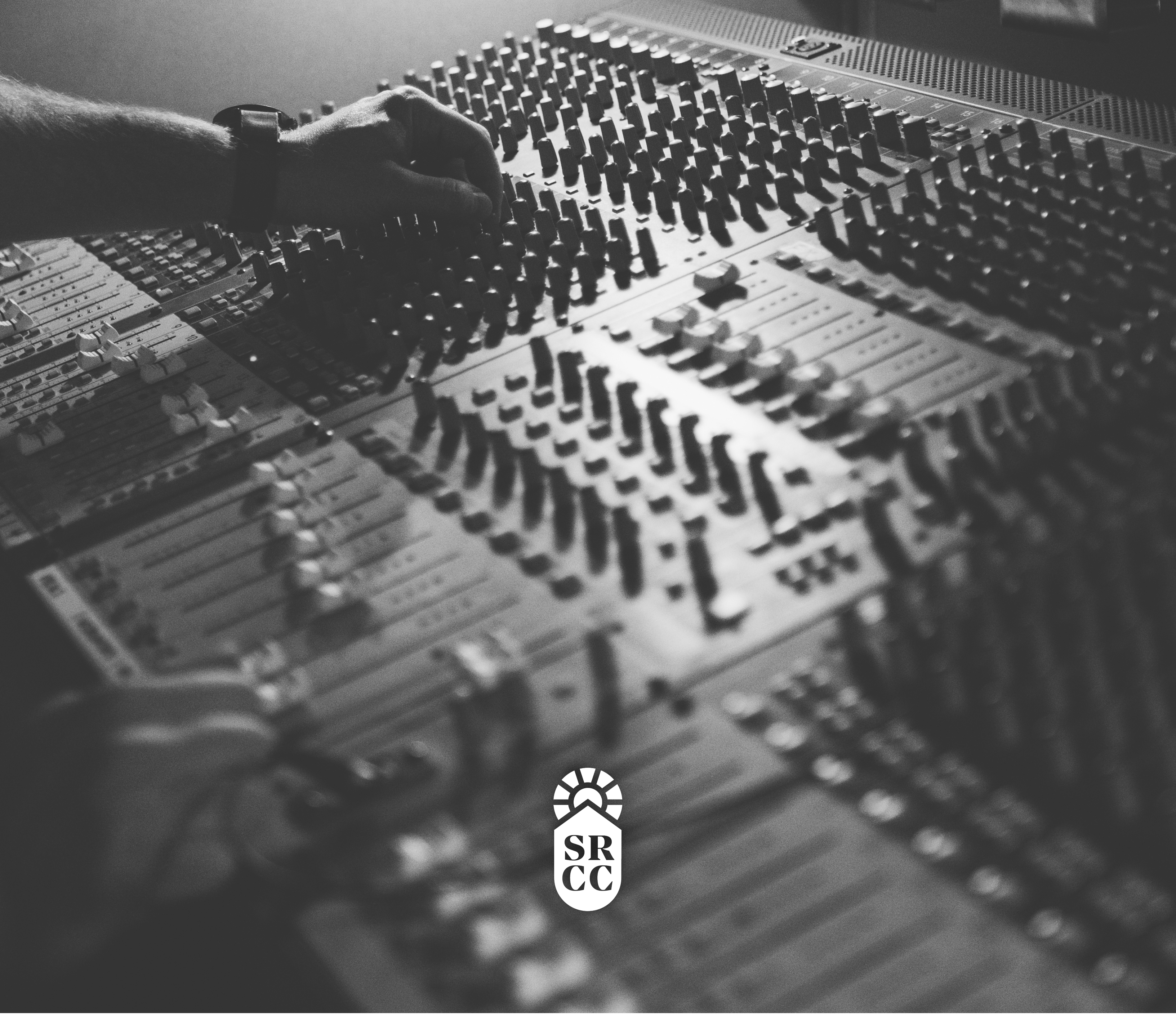

Sound Training Manual

January 04, 2023

SunRidge Community Church
Sound Training Manual

Section 1 ….. Introduction

At SunRidge Community Church we believe that Excellence Honors God and also Inspires People. We take the passage in Malachi 1 very importantly; we want to give our very best to God so as to Honor Him. God is worthy of our Best.

This is why we believe that it takes effort on our part to be our best, we need to develop the skill set so that we can be the best sound people that we can possibly be. The information behind this manual is so that this document could become a training manual for members of SunRidge Community Church Sound Team.

It is intended to provide an overview of basic concepts (with a fair amount of detail) concerning the operation of sound equipment. To help us be the best that we can be. Another important objective of this manual is to describe the goals of our Church Sound Team and to outline the scope of the Sound Team's work; both physical and spiritual.

If you are, or want to be, part of SunRidge Church Sound Team, you should think about your motivation:

- Who are you doing this for - yourself or God?
- Are you doing it to get something - money, recognition, attention, or praise?
- Are you doing it to give something - your time, talent, service, sacrifice, and love?
- Are you willing to spend the time it takes to learn to do the job right?

Running Church sound, even if you do it for 50 years, will not get you into Heaven. Only by faith in Jesus will you enter Heaven. But, if you did run Church sound (and did it well), Jesus may let you play with thunder once in awhile!

Sound System Overview

(Sound starts at the top, progresses through equipment, ends at bottom)

<table>
<tr><td align="center"><u>ELECTRICAL SOUND SOURCE</u>
(Tape, CD, Record, or Electronic Instruments)</td><td align="center"><u>ACOUSTIC SOUND SOURCE</u>
(Voices or Acoustic Instruments)</td></tr>
<tr><td align="center"><u>DIRECT</u>
(Direct Electrical Interface to Sound System)</td><td align="center"><u>MICROPHONE</u>
(Conversion of Acoustical Energy to Electrical Energy)</td></tr>
<tr><td colspan="2" align="center"><u>MIXER</u>
(Electrical device used to blend the sound from several sources)</td></tr>
<tr><td colspan="2" align="center"><u>EFFECTS</u>
(Electrical Manipulation of the Audio Signal)</td></tr>
<tr><td colspan="2" align="center"><u>EQUALIZER</u>
(Tonal control of the sound)</td></tr>
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(Electrical device to amplify or multiply the power of the sound)</td></tr>
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(Conversion of Electrical Energy back into Acoustical Energy)</td></tr>
<tr><td colspan="2" align="center">PROPAGATION
(Affect of Building, Air, and Audience on Sound)</td></tr>
<tr><td colspan="2" align="center">AUDIENCE (ears)
(Appreciation of the Program)</td></tr>
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Section 2 ….. Definition of Terms

Sound Source: Person speaking, singer, musical instrument, or a recording (these will be referred to as the "performer")

Microphone: Converts audible sound energy (moving air) into electrical energy (voltage)

Mixer: Blends the electrical signals from several sound sources

Effects Processor: Adds special effects such as reverb, echo, or delay to the sound

Equalizer: Controls the tonal quality of the sound

Power Amplifier: Increases the electrical power of a sound signal

Speaker: Converts electrical signals into sound energy (moving air)

Propagation: Moving of sound through air, affected by building acoustics, audience, surfaces of floor, walls, and ceiling, air temperature, and humidity

Audience: People who are expected to hear the program

Section 3 ….. Goals for Sound Systems

- To go unnoticed (A perfect sound system should not be noticed by the audience)
- To allow the speaker or performer to feel comfortable (not intimidated by or afraid of the sound system)
- To sound as natural as possible (tonal quality which does not "color" the sound)
- To sound as clear as possible (intelligibility, lack of unwanted echo or bounce back)
- To help people to hear portions of speech/music which are naturally quiet
- To help people in the back of the room hear (even volume throughout the audience)
- To make sound louder for artistic reasons (i.e. greater impact)
- To allow someone to communicate with a large group of people without having to YELL!
- To have every sound source reproduced at the volume which is appropriate for that sound (blend)

Section 4 ….. Job of the Sound Team

- Before you start doing Church Sound, Light, or Video to see if that is what God has called you to do. If you feel it is, pray that God will give you the talents you need to do it well.

  Proverbs 21:30 There is no wisdom, no insight, no plan that can succeed against the Lord.

- Get the training you need. Look for information to read. Talk to other people who have experience. Watch other people run Sound, Light, and Video (not just in Churches, but other places like concerts). Look for courses at local schools.

- Be a servant - you are there to serve, not to be served

- Be humble - people love to point blame at an arrogant sound man

  Proverbs 27:1 Do not boast about tomorrow, for you do not know what a day may bring forth.

  Romans 12:3 If it is possible, as far as it depends on you, live at peace with everyone.

- NEVER upset the Pastor or any performer before the service or program - your job is to SERVE!

  Proverbs 15:18 A hot-tempered man stirs up dissension, but a patient man calms a quarrel.

  Proverbs 29:11 A fool gives full vent to his anger, but a wise man keeps himself under control.

- Do the job as a service to the Lord, not as a service to your ego

- Pray each time before you run sound, light, or video that you will do it to God's glory

  1 Thessalonians 5:17 Pray without ceasing.

- Pray afterwards, Thank God for the privilege of serving Him!

- Work as a TEAM. Support, help, and encourage each other. Give and accept CONSTRUCTIVE criticism gracefully. This is NOT a place for EGO.

  Proverbs 23:23 Buy the truth and do not sell it; get wisdom, discipline and understanding.

- The Sound Technician is to serve the performer or speaker. Help the performer to be as comfortable as possible so that they can concentrate on the message they are delivering and not be distracted by the sound system or Sound Technician.

  Proverbs 15:30 A cheerful look brings joy to the heart, and good news gives health to the bones.

- SMILE (it really does make a difference and it costs so little)

- Give out a complement or word of encouragement now and then.

- The Sound Technician is to serve the audience. Ensure that the audience can hear the performance as well as possible.

- Determine any special needs of the performers / program and be prepared well before the program starts (i.e. additional mikes, monitor speakers, connection of music instruments, cassette tapes, CDs, Video tape, etc.).

- Set-up and check-out the sound system before the performance, check every mike, check every instrument, listen to monitor levels, queue every tape.

- Check for feedback and adjust equalization if necessary.

- Obtain an agenda of what is to happen and when so you can be prepared (ask questions if the agenda is not clear - the sound technician doesn't need surprises!).

- Instruct performers on the proper use of microphones.

- Cooperate with the performers to meet their needs, help them feel at ease, and give them confidence that they will sound good.

- Run a sound check with the performers to set levels, tonal quality, and adjust monitor mix for performers.

- NEVER antagonize the performers!

- Pay attention during the entire service or performance. Do not allow yourself to become distracted. Take your service seriously.

- Listen, Listen, Listen,
  Listen to the program and make adjustments as needed during the program. Listen for correct volume, blend, and tonal quality.

- Watch, Watch, Watch,
  Be prepared for unexpected changes. If someone picks up the wrong mike, make sure you are there to turn it on. If the performer moves to an area that is not lit, make sure you are there to adjust the lighting.

- Tear-down, put-away, and secure the equipment after the performance.

- Take good care of the equipment - treat it like it belongs to God (because it does)

Section 5 ….. The Sound System - Sound Source

When it comes to sound systems, all of the sources for sound fall into one of two main categories: a) Electrical sound sources and b) Acoustic sound sources.

Electrical sound sources are those where the sound originates as an electrical wave generated by some piece of electrical equipment. Examples of Electrical sound sources include: recorded sound (CD, record, or cassette tape player) and electronic instruments (keyboard, electric guitar, electric drums, etc.). Many Electronic sound sources cannot be heard at all without the aid of a sound system.

Acoustic sound sources are those which naturally make sound that can be heard (at least to some degree) without a sound system. Examples of Acoustic sound sources include: voices (talking or singing) and acoustic instruments (acoustic guitar, horns, drums, etc.).

Below, we will discuss the various ways to connect Electronic and Acoustic and sound sources to a sound system:

A. Connecting Electronic Sound Sources

In order to connect an Electronic sound source to a sound system, you usually just "plug it in". That is, an Electronic sound source usually outputs an electronic signal suitable for direct connection to a sound system (the mixer).

CD, Record, and Cassette Tape Players

CD, record, and cassette tape players all have "RCA jacks" on them which can be used to connect them to the sound system. Some mixers also have RCA jacks for input, but many only have XLR and 1/4" jacks. In this case, an RCA to 1/4" adapter is needed.

If you have enough channels on the mixer, you may connect the left and right signals from the CD, record, or cassette to two mixer channels. However, if channels are scarce or if the system is being run mono (rather than stereo), then you may use a Y-adapter to bring both signals into the same channel.

If you are playing "Split-Track" performance tapes where the music is on one track (i.e. the right) and a rehearsal version of the singing is on the other track (i.e. the left), then you MUST connect to two mixer channels in order to be able play just the music for the performance.

If you try to connect a record player to a mixer, you should be aware that older record players put out a special signal called "Phono" which requires special equalization to make it sound normal. If your mixer does not have a "Phono" input (and most don't), then you may need to make drastic changes to the tone controls (equalizer) to get the records to sound OK.

Electronic Instruments - Keyboard, Electric Guitar, Electric Drums, etc.

"Electronic" instruments such as guitars have a built in microphone or magnetic "Pickup". Keyboards output an electronic signal. These signals may be connected to a guitar or keyboard amplifier and/or the sound system mixer.

If the distance between an electronic instrument and the mixer is much more than 25 feet, a "direct box" should be used to convert the unbalanced high impedance signal from the instrument into a balanced low impedance signal which can travel a longer distance (hundreds of feet) without picking up significant noise. The direct box has an input jack that accepts the 1/4" plug of a guitar cord and an output jack that accepts the XLR plug of a microphone cord.

Usually, an electric guitar player and often a keyboard player will want to have their own guitar / keyboard amplifier. This serves two purposes: a) the musician can hear what they are playing better with a dedicated amplifier than they can through the monitor system and b) the musician may want to add special effects that are only available on the guitar / keyboard amplifier. If the guitar / keyboard amplifier is being used to add effects, then it a direct box cannot be used. In order to get the "sound" that the musician wants, you have to place a microphone in front of the guitar / keyboard amplifier to pick up the sound of that instrument for the "House mix". Usually a desk mike stand (6") with a Goose neck (12") works best in this application. If the amplifier has treble and bass speakers, place the microphone half way between them. If it has several full range speakers, place the mike directly in front of one speaker.

Another way to connect electronic instruments to the sound system is via an FM transmitter and receiver. This is most appropriate for guitar and horn players that want freedom of movement. The FM transmitter has a guitar cord which plugs into the instrument and the FM receiver has another guitar cord which plugs into the mixer.

B. Connecting Acoustic Sound Sources

To connect an Acoustic sound source to a sound system, you usually need to use a microphone to convert the "acoustic" sound into an "electronic signal".

Acoustic sound or "sound waves" are rapid minute variations in air pressure created by a person's voice or an acoustic instrument. A microphone is used to convert these sound waves into electrical signals suitable for input into a sound system.

Voices - Talking or Singing:

To get a person's voice into a sound system, you need a microphone. There are many different types of microphones which can be used depending on the particular situation.

Wired / FM Microphones:

Most microphones require a cord to carry the electrical sound signal from the microphone to the mixer.

When the cord presents a problem such as a trip hazard or limits the movement of the performer, then an FM microphone should be used.

FM microphones contain a small FM transmitter which sends a radio signal through the air to an FM radio receiver. The FM receiver is then connected to a mixer input.

Placement of the FM radio receiver and its antenna(s) is very important. It should be close to the microphone (20 to 100 feet) and not near sources of electronic interference (i.e. CD player, effects processor, keyboard, or computers).

You get what you pay for in FM microphones! The more expensive units are called "True Diversity". That means they have two antennas to receive the radio signal from the microphone. The purpose of two antennas is to minimize the multi-path cancellation effect. If one antenna is receiving the signal directly from the microphone and also receiving a reflection of the same signal (i.e. off a metal object), the two signals may be out of phase and cancel each other out, causing "drop-out". However, with a second antenna (properly located), it should not be receiving the same reflected signal and therefore its signal will not be canceled out (at the same time). The receiver automatically switches to whichever antenna is receiving the strongest signal.

If several FM microphones are used at the same time, each must operate on a different radio frequency.

The newest series of FM microphones are called UHF. This means they operate on a very high radio frequency around 900Mhz. Usually they will provide better performance than the VHF units.

Unidirectional / Omnidirectional Microphones:

Unidirectional (also called Directional or Cardoid) microphones "hear" or pick up much better from the front than they do from the sides or rear.

Unidirectional microphones are usually best for sound reinforcement to prevent feedback (because they don't pick up much sound from the main or monitor speakers)

Omnidirectional microphones hear or pick up sound from all directions equally.

Omnidirectional microphones cause feedback easily.

Omnidirectional microphones sometimes sound more natural for recording purposes.

Lavaliere Microphones:

A Lavaliere microphone is a small microphone which usually has a clip to attach it to the performer's clothing. Lavaliere microphones provide "hands-free" operation.

FM Lavaliere microphones provide the greatest freedom of movement.

Lavaliere microphones usually do not sound as natural as hand held mikes and are less desirable for singing.

Depending on the clothing being worn, location of a Lavaliere mike is sometimes a problem. It needs to be located as high as possible and centered with the mike pointing straight up. The best location is clipped on a tie, just below the knot. If the performer is not wearing a tie or a buttoned shirt, it may be difficult to find a place to clip the mike.

In theatrical performances, they often hide an FM Lavaliere mike in the performer's hair either over one ear or above the forehead.

It looks better if you get the performer to wear the FM transmitter in a pocket or under their clothes and route the wire to the microphone through their clothes to minimize visibility of the microphone.

Headset Microphones:

Recently, Headset microphones have been gaining popularity. Many professional performers use a Headset microphone.

A Headset microphone provides the "hands-free" operation of a Lavaliere with superior sound quality for singing and better feedback control.

An FM Headset microphone provides the best sound quality along with "hands-free" operation for a singer.

When combined with an FM Instrument pickup, an FM Headset mike gives a singer / instrument player complete freedom of movement.

PZM Microphones:

A PZM (Pressure Zone Microphones) looks like a flat metal plate with a small raised area containing the microphone element. It may be placed directly on a flat surface such as a floor or table top to pick up sounds from as far as 10-12 feet away. Sometimes, a shield is used to limit the sound pickup angle.

Tonal quality of a PZM microphone may not be as good as other mikes, but they are useful when it is not practical to directly mike the performer.

Acoustic Instruments - Piano, Guitar, Horn, Drum, etc.

There are a great number of ways to mike acoustic instruments. Experiment to find out what works for you and the performer!

Acoustic Guitars:

First, hope that the performer is using a guitar with a good quality built-in Pickup. This can be directly connected to the mixer. If the mixer is more than 25 feet away, a direct box should be used.

If the guitar doesn't have a pickup (or has a pickup that doesn't sound very good), use a directional microphone on a boom mike stand. Place the mike as close to the opening in the guitar as possible without being so close that the performer will hit the mike when playing the guitar.

Similar to electric guitars, an acoustic guitar can be used with an FM transmitter and receiver to connect it to the sound system and allow the performer freedom of motion.

Horns:

A good quality Lavaliere microphone clipped onto the bell of the horn often works well (be careful not to scratch the metal of the horn or you will have a very angry performer on your hands!). An FM Lavaliere gives the performer freedom to move. Some Lavaliere style microphones come with a special clip for horns which is padded with felt so it doesn't scratch the horn. It also allows the microphone to face into the bell of the horn.

A directional microphone on a straight, goose-neck, or boom mike stand can also be used.

Pianos:

On a Grand Piano, the lid should be opened (probably on the short stick). An Omnidirectional microphone on a boom stand can be positioned so that it hovers near the middle of the strings. Tonal quality can be changed by moving the mike towards lower or higher strings.

An alternative is to use a PZM microphone attached to the inside of the Piano lid or possibly attached to the bottom. If you are in a hurry, you can even place a PZM on the floor under the piano.

Since the wood of the Piano tends to direct sound from the speakers back to the microphone, feedback may be a problem.

An upright Piano can be miked with a PZM inside the lid or a mike on a boom stand behind the Piano. Tonal quality varies greatly with mike placement.

Before we complete the discussion of sound sources, there are some related areas we should discuss:

Phantom Power:

Some microphones contain circuitry which requires power to make the microphone operate. Condenser microphones are one example. Some mixers provide "Phantom Power" for this purpose. Phantom Power is usually 9 to 48 VDC.

If the mixer has Phantom Power, it usually has a switch to turn it on or off. Sometimes there is a switch for each channel and sometimes the switch applies Phantom Power to 4 or 8 channels at a time. Be sure the Phantom Power switch is turned off if you are not using microphones that need it. Try not to leave Phantom Power turned on to a mixer channel connected to a Direct Box or to a microphone that has an On/Off switch - it may cause problems.

Microphone Proximity Effect:

Directional mikes always have a "proximity effect". When a person is in close proximity to the microphone (2" or less), there is a dramatic increase in the low frequency (bass) response.

The proximity effect can be used to advantage by constantly staying close to the mike (less than 2") for a more POWERFUL sounding voice.

Occasionally getting close for effect adds variety.

When a large number of microphones are used for a singing group, the proximity effect is usually a disadvantage: Most of the time, singers are to far away from the mike (8" to 18") for the proximity effect, so the sound system is set up to sound best without the proximity effect. On solos, singers get in close (2" to 4") and the proximity effect makes them have too much "bass".

When the system is set up primarily for people using mikes at distances 8", then individuals should not get closer than 3" to prevent excessive bass.

Miking Techniques:

As mentioned above, the distance the sound source is from the microphone is very important. For singing and speaking, 2" to 4" is the optimum distance.

Equally important to distance is the angle of the microphone. When using a hand held microphone, it should be positioned just below the mouth, even with the chin. The microphone should be held almost vertical with only a slight tilt toward the mouth. You should speak or sing across the top of the microphone. If you speak directly into the top of the microphone it will cause excessive breath noises and "P-popping".

Mike Stands:

When a hand held mike or a lavaliere is not appropriate, then a mike stand is probably the best solution. Again, the proper choice of mike stands can make a big difference:

Straight stand - usually suitable for a speaker or singer or a group of people.

Boom stand - helps get the mike closer to a person playing a piano, keyboard, guitar or other instrument.

Goose-neck stand - often useful to get a mike close to an acoustic instrument.

Hanging mikes - sometimes the best way to mike a choir is to hang microphones above and in front of them.

Podium - often a lectern or podium has a goose neck microphone built in.

Connecting Cords:

It is important to understand the different types of cords that are used to make the connections between the various parts of a sound system. All cords used to connect inputs to a sound system or to connect the various parts of the sound system together MUST be shielded cable in order to prevent the system from picking up hum and noise. The only exception to this is the connection between the amplifier and the speakers which does NOT need to be shielded.

The cords used to connect sound sources to a mixer fall into one of two categories: Unbalanced and Balanced

#1 Unbalanced:

The term "unbalanced" refers to a two conductor cord where one conductor is a grounded shield and the other conductor caries the sound signal. However, the sound signal must also return via the shield. The connection is referred to as "unbalanced" because one conductor is grounded and the other is not.

Unbalanced cords may use RCA jacks, or 1/4" plugs.

Unbalanced connections generally should not be used for distances much greater than about 25'. At greater distances, an unbalanced cord picks up too much electronic "hum" and noise. Although the shield protects the inner conductor from picking up hum and noise, since the sound signal must return via the shield, it is affected by the hum and noise picked up by the shield.

#2 Balanced

The term "Balanced" refers to a three conductor cord where the outer conductor is a grounded shield and two inner conductors carry the sound signal. The connection is referred to as "balanced" because the two inner conductors that carry the sound signal are "balanced" at the same voltage level.

Balanced cords can carry sound signals much further than unbalanced cords, typically several hundred feet. Balanced cords are not nearly as susceptible to picking up hum or noise because the sound signal only goes through the inner two conductors and never uses the shield. Therefore hum and noise picked up by the shield is simply grounded.

Section 6 ….. The Sound System - Mixer

- A mixer is an electronic device which combines the electrical sound signals from microphones, instruments, Tape, CD, etc.

- With the mixer, you can adjust the volume and tonal quality of each input source to achieve a harmonious and pleasing blend ("mix") of all the sound sources.

- The output of the mixer goes through the compressor (optional), the digital delay (only for large distributed speaker systems), the 31 band graphic equalizer (or parametric equalizer), the cross-over (only for 2-way or 3-way speaker systems), to the amplifier(s), and then to the speaker(s). If you are using a stereo speaker system, all the equipment is doubled!

- Most mixers provide separate controls for the main speakers and one or more sets of monitor speakers. Each set of monitor speakers also needs a compressor (optional), 31 band graphic equalizer, amplifier, and speaker(s). Sometimes, a separate mixer is used for the monitor mixes.

- Usually, the mixer provides the ability to connect effects processors to add reverb, echo, delay, etc. to the sound.

- It is very desirable that the house mixer and sound technician be located somewhere near the middle rear of the audience area so that the sound technician will hear the same sound the audience hears (but not block the view of the audience). It is best if the mixer is located on the main floor with the audience, but sometimes it is necessary to locate it in a balcony. The house mixer should not be located in another room or behind a glass wall.

- If a separate mixer is used for the monitor mixes, it should be located to one side of the stage where the monitor mix technician can have eye contact with all the performers (yet not be seen by the audience).

- The microphones are connected to the mixer either directly via mike cords or through a "snake" if the mixer is too far from the mikes.

- Instruments (guitars & keyboards) are connected to the mixer through a special adapter called a "direct box" (used to convert unbalanced high impedance 1/4" cord connections to balanced low impedance XLR microphone connections).

- A "snake" is a multi-conductor cable that caries a number of microphone lines (usually 16 or 24) from the performance area in the front to the mixer location in the back.

- If a separate monitor mixer is used, the snake must have splitters on each channel to feed both the Monitor Mixer and the House Mixer.

- A mixer usually handles 8, 16, 24, 32, 40, or 48 channels or inputs.

- Below, we will attempt to describe most of the controls found on the average GOOD quality mixer that might be found in a church environment:

There are separate controls on each channel for:

Trim Pot - The Trim Pot or channel gain control is used to compensate for the difference between the various input sources. Generally, you should start with the Trim Pot for each channel set at the 12:00 (or straight up) position. During practice, you should adjust the Trim Pots to compensate for channels with louder or quieter signals. For example, the keyboard may put out a stronger signal than a guitar. In this case, the trim pot for the keyboard may need to be turned down or the trim pot for the guitar turned up.

During practice, listen to each channel one at a time with the headphones (using the PFL or AFL switch). Adjust the Trim Post so that each channel sound about the same loudness. If your mixer has per-channel meters, they can make it a lot easier to adjust the Trim Pots. On most mixers, when you operate the PFL or AFL switch, it shows that one channel on the main L/R meters. This can also be used to help you adjust the Trim Pots for nearly equal levels from all channels.

When adjusting the Trim Pots, you need to be careful not to over-drive the channel. By setting the Trim Pot too high, you can cause the channel to over-drive and distort or clip. This is not a pleasant sound! Some mixers have a little red LED on each channel to let you know when it is set too loud. You can also use the per-channel meters or press the PFL or AFL button and look at the main L/R meters. If the meter goes into the red, turn the Trim Pot down a little.

Mike / Line switch - The Mike / Line switch is not included on all mixers. It is used to select between the low level mike signal connected to the XLR jack or the high level line signal connected to the 1/4" jack. If this switch is not present, usually both of the jacks are active.

Phantom Power switch - The Phantom Power switch is not included on all mixers. It is used to turn Phantom Power on or off. Sometimes one switch controls Phantom Power for 4 or 8 channels. Normally, the Phantom Power switch should be left in the off position unless you are using a microphone that requires Phantom Power to operate. Sometimes the Phantom Power switches are beside the channel input jacks.

Equalizer or Tone Control - The per channel Equalizer controls are used to adjust the tonal quality of each input and to reduce or eliminate feedback. Low end (low cost) mixers may only have treble and bass tone controls. Better mixers have high, mid, and low frequency controls. Top of the line mixers have additional equalization controls which include frequency adjustment capability and sometimes even an adjustment for the width (or Q) of the control. Often low-cut and sometimes high-cut switches are also provided.

> **High-cut** - The high-cut switch should usually be left turned off. It is only needed if the input to that channel has an unusual amount of high frequency energy (i.e. a low quality guitar pickup).
>
> **High** - The high frequency rotary tone control is often useful to smooth out a channel which is particularly shrill. Seldom should it be used to boost high frequency sounds because this can quickly cause feedback. However, when a keyboard or guitar with a pickup is connected to the channel, the high frequency control can be used to brighten the instrument without danger of feedback.
>
> **Sweepable controls** - Sweepable tone controls are very helpful in eliminating feedback. The best mixers have several Sweepable controls on each channel.
>
> Before practice starts, you should check each channel for potential feedback. Start with all the tone controls set to null (straight up or 12:00 position). Turn the channel on to normal operating level. Then slowly move the channel fader up the whole way. If you get feedback before you reach the top, move it down just a little to keep the feedback quiet and stable.
>
> Decide whether the feedback is a high frequency or a low frequency. If it is high, use the higher Sweepable tone control. If it is low, use the lower Sweepable control (if you have one). Turn the level control down about 2 hours (i.e. from the 12:00 position to the 10:00 position). Then "sweep" the frequency control until the feedback goes away. It takes a little getting' used to, but with practice you will get the hang of it.

Then move the channel fader up a little more and see if you get feedback at another frequency. If you have another Sweepable control, you can eliminate that feedback too.

Low - The low frequency rotary tone control is useful in eliminating "boomy" sound in mikes. It can also be used to add or reduce bass for keyboards or guitars. Occasionally, it can be used to eliminate a really low frequency feedback.

If you turn the bass up, you need to be careful not to over-drive the channel and cause it to distort.

Low-cut - The low cut switch should usually be enabled (pushed in) on most channels except keyboard, bass guitar, drums, and CD. This switch takes out low frequency rumbles (like breathing noise and air conditioner sounds).

Monitor or Auxiliary send(s) - The Monitor or Auxiliary sends are most often used to provide a separate "mix" for the monitor speakers.

This allows the performers to hear a somewhat different "mix" than the audience hears. Often a performer will want to hear more of their own voice or instrument than is in the house mix. Possibly the instrument which sets the tempo for the whole group needs to be louder in the monitor mix than in the house mix. Sometimes different members of the performance have different monitor requirements. This can be solved by providing multiple monitor mixes and setting up several monitor speakers (each with its own equalizer and amplifier).

The Monitor or Auxiliary controls are used to produce one or more Monitor mixes. Low end mixers may only have one set of Monitor or Auxiliary controls. High end mixers often have as many as 8 Monitor or Auxiliary sends, sometimes even more.

To set up the Monitor mixes, first you must decide how many Monitor mixes you are going to use. You must connect each Monitor or Auxiliary send through a compressor (optional), through a 31-band graphic equalizer (highly recommended for feedback control), to the monitor amplifier, and then to the appropriate set of monitor speakers (which have been placed in front of the correct person or group of people).

Start by turning the main Monitor or Auxiliary send controls off. Then, for each Monitor mix, you must decide which of the inputs should appear in that mix. Start by setting the monitor control for each input you want in that monitor to the 12:00 position. If some inputs should be louder in that monitor, start them at the 1:00 position and for those that should be quieter, start them at the 11:00 position.

Next, turn the main Monitor or Auxiliary controls up slowly. SLOWLY find the point at which the monitor begins to feed back. Reduce the control at least 9db below the feedback point or an eighth of a turn if the control is not labeled in dB.

While the group is practicing, go on stage, listen to the monitors. Ask the performers if they are hearing what they want to hear. Adjust the Monitor or Auxiliary controls for each set of monitors to suit the performer's needs.

Generally, Monitor, Auxiliary, and Effects sends do not have meters or overload warning lights on them. Generally, you won't have trouble with overload if you keep most of the controls near the 12:00 position.

Pre / Post switch - The Pre / Post switch near the Monitor or Auxiliary sends determines whether the signal sent to Monitor controls is affected by the main Channel Fader (slider). When the switch is in the "Pre" position, the Monitor signal is taken from a Pre-Fader position. The Monitor level is NOT affected by the Channel Fader. When the switch is in the "Post" position, the Monitor signal is taken from a Post-Fader position. The Monitor level IS affected by the Channel Fader. Generally

Rock Music groups prefer the "Pre" position and choir type groups prefer the "Post" position.

If the performance contains solos where the sound technician must move the Channel Fader up for the solo, then if the switch is set in the "Pre" position, the monitor for the solo will NOT get louder when the house mix gets louder. If the switch is set in the "Post" position, the monitor for the solo will get louder the same as the house mix gets louder for the solo.

Effects send(s) - The Effects sends are used to select certain channels that should be sent to effects processors such as reverb, chorus, or digital delay. Generally, processed sound is returned from the Effects processor to the mixer's Effects Return inputs. If you want to have better control over the effects sound, you can connect it to an unused channel.

Low end monitors generally only have one set of Effects sends. High end monitors usually have multiple Effects sends. Monitor, Auxiliary, and Effects sends are all similar in nature. The main difference is whether the send is "Pre" or "Post" Fader. It is best to read the manual on your mixer to determine whether its Monitor, Auxiliary, and Effects sends are Pre or Post fader or whether you have a switch to make the selection yourself.

Subgroup Select switches - The Subgroup Select switches (not included on all mixers) are used to group several channels into a Subgroup (i.e. Subgroup 1 or S1 = soprano, S2 = alto, S3 = men, S4 = instruments). Sometimes there are Subgroup switches labeled L/R which bypass the Subgroups and send the signal directly to the main L/R output.

Left / Right Pan control - The Left / Right Pan control is used to select whether the signal should be sent to the odd numbered Subgroups (Left), even numbered Subgroup (Right), or both (center or 12:00 position). If the signal is sent directly to main output, then the pan control determines the location of the signal in the stereo mix.

If you are just sending the sound to the house speakers, usually you will use a mono send (unless the house speakers are set up for stereo). However, if you are also making a recording of the sound, it may be desirable to make a stereo mix where the sold on the left and right are somewhat different. By turning the Pan controls part way to the left or right, you can position each sound within the stereo field. When you mix for stereo, generally you need to use the Subgroups in pairs.

Channel Fader The Channel Fader, slider, or level control is used to adjust the level of each channel in the main mix.

During practice, many sound engineers like to setup the mixer so that they have a good well balanced general mix with all the Channel Faders are in a straight horizontal line. This way, after making any changes in Channel Fader levels needed for a particular song or solo, they can return all the Faders to a straight line. This is easier than having each Channel Fader set at a different level and after a solo having to remember where that Fader was before you changed it for the solo.

Generally, you should start with the main house L/R control off and all of the Channel Faders up at the position marked either "0" or "U". This is usually about 3/4 of the way up. Then, slowly raise the house L/R controls to their "0" or "U" mark. If you get feedback before you get there, then move the house L/R controls down to about 9db below the feedback point.

While the group is practicing, adjust the Trim Pots on each channel so that you get a good blend or mix. It is helpful to listen to each channel individually with the headset by pressing the PFL or AFL button.

If there are solos or other areas in the performance where one or several performers need to be accentuated, move their Channel Faders up for that portion of the program and then return them back to the "0" or "U" mark.

With most mixers, there is about 10db of gain left between the "0" or "U" mark and the top or maximum position of the Channel Fader. With some performers, this is not enough extra gain to accentuate a solo adequately. You may find it works best to setup the mixer with all the Channel Faders normally set at "-5" or "-10" so that you have 15db or even 20db of room left to accent a solo. It is all a matter of what works best for you!

Pre-Fader Listen or After-Fader Listen - The Pre-Fader Listen (PFL) or After-Fader Listen (AFL) button allows the sound technician to listen to one channel at a time using a headset. If the button is labeled PFL, then the level heard in the headset is independent of where the Channel Fader control is set. However, if the button is labeled AFL, then the level heard in the headset is dependant on the setting of the Channel Fader.

During practice, you should listen to each channel using the PFL / AFL button. The Trim Pot on each channel should be adjusted so that the volume (and meter readings) on all channels appear nearly the same.

The PFL / AFL button can also be used to look for troubles, like the source of a hum or buzz. Pressing PFL / AFL on a Subgroup allows you to hear that group alone without the rest of the mix. If your mixer has a PFL / AFL button on the Monitor, Auxiliary, or Effects sends, you should listen to each of them to see that there is no distortion (overload, or clipping) and to see that the mix sounds like what you expected.

Mute or Channel On switch - The Mute or Channel On switches are used to enable / disable individual channels. Not all mixers have these switches. Mute switches are more prevalent than Channel On switches. If the Mute switch is pressed, the channel is turned off (muted). Generally, a Mute switch has a light which illuminates when the channel is Muted.

Some mixers provide Mute Automation. These mixers include a computer which can remember a number of snapshots of mute switch settings and recall them at will. This is particularly useful in large drama presentations with many FM microphones worn by all the performers. All the FM microphones can normally be muted (to eliminate background noise and minimize feedback possibilities) and the system can be programmed to unmute the correct mikes for each scene.

If your mixer has Subgroups, there are separate controls for each Subgroup:

Subgroup Left / Right Pan control - The Subgroup Left / Right pan determines where this subgroup is positioned in the main L/R stereo mix.

Subgroup Left / Right Select - Some mixers have Subgroup Left / Right select buttons instead of Subgroup Left / Right Pan controls. The L/R select determines whether the Subgroup is sent to the Left main, Right main, or both outputs.

Subgroup Fader - The Subgroup Fader (level control, or slider) is used to adjust the level of the subgroup within the main mix. This allows the Sound Technician to raise or lower the level of a whole Subgroup of inputs with one control. For example, if the altos have a verse by themselves and need to be a little louder, you can raise the alto Subgroup for that verse.

Subgroup PFL / AFL - The Subgroup Pre-Fader Listen (PFL) or After-Fader Listen (AFL) button allows the Sound Technician to listen to one Subgroup at a time in the headset. If it is a PFL button, the level in the headset is independent of where the Subgroup Fader is set. If it is an AFL button, the headset level is controlled by the Subgroup Fader.

Mute or Subgroup On switch - The Mute or Subgroup On switches are used to enable / disable individual Subgroups. Not all mixers have these switches. Mute switches are more prevalent than Subgroup On switches. If the Mute switch is pressed, the Subgroup is turned off (muted). Generally, a Mute switch has a light which illuminates when the Subgroup is Muted.

The main L/R output usually has only one control:

Master L/R Faders - The Master L/R Faders are used to control the main output of the mixer to the house speakers. Most Sound Technicians prefer to normally operate the Master L/R Faders at the "0" or "U" position. In order to do this, the volume or gain control on the house amplifier should be adjusted so that the system operates at a normal comfortable level with the Faders set at "0" or "U".

Depending on the type of mixer you have, there may be additional controls, but since these vary greatly with different mixers, we won't try to address them all here.

The mixer controls listed above are used to adjust, blend, and tune the sound, but it is the input and output jacks that are used to connect the mixer to the various other components of the sound system:

Channel Jacks - Each channel of the mixer usually has two input jacks, one is usually a low impedance XLR jack and the other is usually a high impedance 1/4 inch jack. Normally, microphones are connected to the XLR jack and instruments, keyboards, tape decks, and CD players are connected to the 1/4 inch jacks.

Frequently, each channel will also have "insert" jacks. Better mixers usually have two 1/4 inch insert jacks labeled "send" and "return". Some mixers use a single three conductor 1/4 inch jack for both functions. The purpose of the insert jacks is to send the signal from one channel to an effects device or a graphic equalizer that is dedicated to that one channel.

Effects Jacks - If the mixer has one or more Effects Sends, each of these will have an output jack. Each Effects Send has a corresponding Effects Return with an input jack.

Generally, the Effects Send output jack is connected to the input of the Effects Unit and the output of the Effects Unit is connected to the Effects Return input jack. If the Effects Unit is stereo, you have two choices: You can use only the left channel output jack (which makes the unit operate in mono mode) and connect it to one Effects Return input jack. Alternatively, you can use both the left and right output jacks and connect it to two Effects Return input jacks, but only do this if you are doing a stereo mix.

Monitor / Auxiliary Jacks - If the mixer has Monitor or Auxiliary Sends, each of these will have an output jack. The Monitor and Auxiliary outputs are generally connected through an Equalizer to the Power Amp for stage monitor speakers or auxiliary speakers.

Sub-Group Jacks - Usually, if the monitor has Sub-Groups, each will have its own output jack. The Sub-Group outputs are most often used when doing multi-channel recording. In this case, each Sub-Group is sent to a different track on the recorder.

Main Output Jacks - All mixers have output jacks for the main mix. Usually, they have left and right jacks. Frequently, they will have both 1/4 inch and XLR jacks (it doesn't matter much which type of jack you use). Some mixers have a Mono or Sum output which is a combination of the left and right outputs.

The main output jacks should be connected through an Equalizer to the Power Amp for the main speakers.

If you are doing a Mono mix, you can use just the left output jack to feed the main speakers (because the left and right are identical). You might want to use the right output jack to feed the Narthex, Cry-Room, or cassette recorder.

If however, you are doing a Stereo mix (for the purpose of making a stereo recording), then it is best if your mixer has a Mono or Sum output to use to feed the main Equalizer and Power Amplifier. If you MUST use the main left and right outputs, you need a way to "sum" them together. If you simply use a "Y" cord, your recording will become Mono also. The best way to "sum" the left and right channel is to build a special "Y" cord with a resistor network (probably about 1Kohm in series with each side and a 1Kohm shunt).

```
                ------1Kohm-----O-----1Kohm-------
                                |
   Left Output                1Kohm                     Right Output
                                |
                ----------------O-----------------
                             To Mains
```

If you are doing a Stereo house mix (and we recommend against that), then you will need to connect the left and right main outputs through two separate Equalizers to two separate Power Amps.

Recording Jacks - If you are recording the mix, you have a number of options: Some mixers have a separate set of RCA jacks labeled Record Out, If your mixer has both 1/4 inch and XLR main output jacks, you can use the 1/4 inch jacks for recording and the XLR jacks to drive the house, If you are running a Mono mix, you can use the left main output for the house and the right main output for recording, you could also use an effects or monitor send for recording (but that requires you to set up a separate mix for recording), and finally, you could use "Y" adapters to split the main output for recording.

Headset Jacks - Most mixers have a headset output jack. If the mixer has PFL buttons (Pre-Fader Listen), then you can listen to one channel at a time or the whole mix. You can also listen to each effects and monitor send.

Finally, there are a few special items that are provided on some mixers:

Talk-Back - Some mixers support a "Talk-Back" microphone which allows the Sound Technician to talk through the main and/or monitor speakers.

Clear-Com - Some mixers support a "Clear-Com" system which is an intercom that can be connected between the Sound Technician, Lighting Technician, Monitor Mixer, Stage Manager, Director, etc. as needed.

Before we leave our discussion of mixers, we will go over some basic Mixer Operation principles:

Mixer Pre-set - Before starting to connect a mixer, you should go over all the controls and pre-set them so that when you turn the system on, you won't get disastrous feedback. Make sure all the Trim Pots are set all the way off (left), set all the equalization controls straight up, start with all the effects, monitor, and auxiliary sends turned off (left), turn off (down) all the channel Faders, sub-groups, and mains.

Equalizer Pre-set - Set all the sliders on the main house equalizer and all the monitor equalizers to their mid-point.

Power Amp Pre-set - Turn the volume control on the main and monitor power amps all the way off (left).

Mixer Connections - Next, connect all the mixer inputs and outputs, equalizers, main amplifier, monitor amplifiers, main speakers, and monitor speakers.

Power Connections - To minimize the chance of encountering "ground hum", it is best to connect the entire sound system to a single AC outlet (assuming your system isn't too powerful for one circuit breaker). Select an AC outlet near the power amplifiers and put a multi-outlet strip there. Run 3-conductor extension cords as needed to reach the other equipment. Never cut off the third prong! However, you should have a supply of ground lift adapters (3 prong to 2 prong) handy.

Power-Up - Turn on the mixer, tape or CD player, the effects units, the equalizers, and finally the main and monitor power amps.

System check-out

Start by bringing up the main Faders to normal (0 or U) position.

Then bring up the sub-groups to normal (0 or U).

You should have a tape or CD player connected to the system. Turn the trim pot on the tape / CD channel all the way off (left) Now play some music (but don't expect to hear it yet).

Make sure a sub-group is selected for the tape / CD channel and bring the tape / CD channel fader up to normal (0 or U).

Slowly turn the tape / CD channel trim pot up until the main left / right meters read high in the normal (green) range without going into the red.

Now, with the amplifier volume turned all the way off (left), turn on the main amplifier power. Slowly increase the volume until you hear the music out of the main speakers.

Check that all the main speakers are working and increase the amplifier volume setting until the music is at the desired performance level. When you are satisfied, turn off the main Faders to turn off the music in the main speakers.

Next, turn the tape / CD channel control for the first monitor mix up half way (to the 12:00 position). Also set the first monitor send level to half way (12:00).

Now, with the amplifier volume turned all the way off (left), turn on the first monitor amplifier power. Slowly increase the volume until you hear the music out of the first monitor speakers.

Check that all the first monitor speakers are working and increase the amplifier volume setting until the music is at the desired performance level. When you are satisfied, turn off the tape / CD channel first monitor send to turn off the music in the first monitor speakers.

Repeat this process for each set of monitors.

You still have the effects units to set, but we'll leave that to you to figure out!

Section 7 ….. The Sound System - Effects Processors

Reverb: -Gives the sound a more spacious or roomy feeling

Digital Delay: -Provides one or more echoes of the original sound.
-A delay set for just a 10 to 30ms makes the sound fuller, like two people are singing.
-Delays greater than 40ms are clearly noticeable.
-Multi-tap delays give a repeating echo.

In large auditoriums where the main speaker system cannot adequately cover the whole audience area, additional speakers may be distributed to other parts of the room. Digital Delays are used to delay the sound sent to the distributed speakers so that the audience hears the sound from the main speakers and the sound from the distributed speakers at the same time. If all the distributed speakers are the same distance from the main speakers, a single Delay can be used. If the distributed speakers are at different distances, multiple Delays are needed.

Multi-Effects Processor: -Provides a large variety of effects in a single unit.
-Usually includes reverb, delay, multi-tap delay, chorus, flange, tremolo, vibrato, gates, and other effects.
-Some units allow effects to be combined.
-Some units allow you to program your own effects (if you know how).

Compressor / Limiter: -Especially useful for a performer whose voice varies from very quiet to very loud, beyond the dynamic range of the sound system.
-A compressor / limiter reduces the dynamic range of the sound. It allows low volume sounds to pass through unchanged, but loud sounds are reduced in level.
-A compressor / limiter can be used to prevent over driving the power amplifier. This prevents distortion and possible equipment damage.

Gate: -Especially useful to eliminate low level background noise
-A gate keeps the signal turned off until its volume reaches a pre-set threshold, then it is turned on. When the signal becomes quiet again it is turned off. This process happens very rapidly and can be used to eliminate background noise when the person is not speaking (even between words).

Section 8 ….. The Sound System - Equalizer

Why is equalization needed?

-To increase gain before feedback.
-To compensate for Acoustical problems in the room.
-To tailor the tonal quality of the performance.

Always start with no equalization (all controls set "flat").

-Equipment manufacturers design their equipment to be used with NO EQUALIZATION.
-Only use equalization to eliminate problems (i.e. feedback, too much bass, too sharp, etc.).
-After problems are eliminated, there is seldom need to change equalization to enhance the sound.

Room factors which may require equalization:

-Sound reflections off of hard surfaces (walls, ceilings, and floors) causes uneven frequency response because of out-of-phase reflections (i.e. drop a stone in a pond and watch the ripples as they hit the shore and reflect back).

-Sound absorption by soft surfaces (seats, people) causes loss at high frequencies.

-Every room is different! Some seem to increase high frequencies, others absorb highs. Some resonate with base, others seem to have no bottom end at all.

Individual channel equalization on the Mixer.

-Adjust for differences in different types of Mikes.
-Adjust for specific instruments (keyboard, guitar, sax).
-Adjust to optimize sound for particular voices (men, women).
-Eliminate feedback points unique to one Mike.
-Reduce breath noise and "pops".
-Compensate for "proximity effect" when performer "eats the mike".

Monitor equalizer

-Adjust to increase gain before feedback.
-Eliminate "hollow" or "ringing" sound.

Main System equalizer

-Adjust tonal quality of main speaker system to compensate for room acoustics.
-Only if necessary, adjust to increase gain before feedback.

Automatic Feedback Eliminator

-This is a special type of equalizer which "listens" for feedback, determines its frequency, and automatically sets a narrow notch filter to eliminate it. Multiple notch filters are set at different feedback frequencies.

Section 9 ….. The Sound System - Amplifier

- The amplifier receives the combined or mixed signal from the mixer.

- The amplifier is the last component in the sound system before the speaker. Generally, the sound signal progresses through the sound system as follows:
    - Microphone or Instrument
    - Mixer
    - Compressor (optional)
    - 31-Band Graphic Equalizer (or parametric equalizer)
    - Digital Delay (only for remote speakers that must be time-aligned with the main speakers)
    - Cross-Over (only in the case of bi-amped or tri-amped speaker systems or a sub-woofer)
    - Amplifier
    - Speaker(s)

- The amplifier supplies the power to drive the speakers.

- Most speaker systems, especially lower power speakers only require a single amplifier. However, some speaker systems are designed to work bi-amped or tri-amped. In the case of a bi-amped system, two amplifiers are needed, one for low frequency sounds and one for high frequency sounds. In the case of tri-amped, three amplifiers are needed for low, midrange, and high sounds. If a sub-woofer is used for very low sounds, another amplifier is needed for it.

- Separate amplifiers are needed for the Main System, remote speakers (if required), and the Monitor System.

- When multiple monitor mixes are used, each mix requires its own amplifier.

- In order for an amplifier to make the sound twice as loud, it must supply four times as much power. For example, it takes a 400 watt amplifier to be twice as loud as a 100 watt amplifier.

- Always use an amplifier capable of supplying more power than you need. Distortion increases dramatically when an amplifier is operated at its maximum power. Having plenty of "head room" or reserve power reduces the chance of distortion.

- It is important that an amplifier have very low background noise. Even a small amount of "hiss" can be very objectionable.

- The power rating of the amplifier and the speakers must be similar to reduce the chance of damaging the speakers.

Section 10 ….. The Sound System - Speaker

Speakers are usually classified as full range, tweeter, midrange, woofer, or sub-woofer. A full range speaker is designed to handle the full range of sounds most people can hear. A single amplifier is all that is needed to power a full range speaker.

However, tweeter, midrange, woofer, and sub-woofer speakers are only designed to handle a portion of the sound spectrum. These speakers require a device called a Cross-over to work properly.

- **Cross-Over** - The Cross-over splits the full range audio signal into two, three, or four ranges to be delivered to separate speakers. A 2-way speaker system consists of only a tweeter and a woofer. A 3-way system consists of tweeter, midrange, and woofer. And, a 4-way system consists of tweeter, midrange, woofer, and sub-woofer.

  There are two ways that the Cross-over can be connected: either after the amplifier or before. Most low to medium power speaker systems connect the Cross-over after the amplifier. Frequently, the Cross-over is inside the speaker cabinet. High power speaker systems often use a Cross-over connected before the amplifiers. This also makes it necessary to have a separate amplifier for each speaker. A 2-way system must be bi-amped (that is, 2 amplifiers). A 3-way system must be tri-amped (that is, 3 amplifiers). If a sub-woofer is used, it also requires a separate amplifier if the Cross-over is connected in front of the amplifiers.

- **Sub-Woofer** - The Sub-Woofer reproduces extremely low frequencies from about 100 Hz down to 20 Hz. These frequencies are "felt" more than heard. The Sub-Woofer gives the bottom end "beat" to music and the thunderous effects to movie sound tracks. If your sound system is not used for music with a heavy beat or movie sound tracks, then you may not need a Sub-Woofer.

  Sub-Woofer may be placed wherever it is convenient since the human ear cannot tell what direction bass sound comes from

- **Woofer** - The Woofer reproduces low frequencies from about 500 Hz down to 100 Hz, the bass sounds. Speaker placement is not critical because Woofers are Omnidirectional.

- **Mirage** - The Midrange speaker reproduces midrange frequencies from about 500 Hz to 6000 Hz. The midrange area contains most of the sound for voices and instruments. Speaker placement is more critical because mid-range sound is more directional.

- **Tweeter** - The Tweeter reproduces high frequencies from about 6000 Hz to nearly 20,000 Hz. It is responsible for the brilliance in the sound, mostly associated with harmonics. Speaker placement and angle are critical because high frequency sounds are very directional.

- **Full Range** - A Full Range speaker is a single speaker which attempts to reproduce the entire audio spectrum, usually not as well as a multi-speaker arrangement. Full range speakers are practical for low power speakers, but not for high power.

- **Main or House Speaker System** - The Main or House Speakers deliver the sound to the audience. Usually the a combination of tweeter(s), mid-range(s), woofer(s), and possibly sub-woofer(s) designed for smooth frequency response over a wide frequency range and able to operate at high volume levels is used for the Main or House Speaker System.

  Usually, a central cluster is best for the main speaker system. A number of multi-speaker cabinets are arranged in an arc and suspended from the ceiling just in front of the center of the performance area. The number of cabinets, angle of speakers, and angle of cabinets is critical for even coverage.

  If the room is not suited to a central cluster or the main system must be portable, a distributed system with one or more multi-speaker cabinets just in front of each side of the performance area should be used. The height of the cabinets, angle of speakers, and angle of cabinets is important for even coverage.

A distributed system will experience the "comb filter" effect to some degree. Each audience member hears sound from both the left and right speakers. These sounds arrive at different times and therefore are out of phase with each other. The amount of phase difference depends on the frequency of the sound. Therefore, from each individual audience seat, some frequencies are louder (in phase) and some frequencies are quieter (180 degrees out of phase).

- **Monitor System** - The Monitor Speaker System is located in or aimed at the performance area. Monitor speakers enable the performer(s) to hear themselves and other necessary elements of the program (i.e. music).

  Usually Monitor speakers consist of a combination of a tweeter and a woofer in a slanted cabinet designed to aim the sound back toward the performers.

  Monitor speakers should be placed to satisfy the performer(s). Wedge shaped monitor speakers should be placed on the floor in front of the performer so they aim the sound towards the rear of the performer's mike. Sometimes side fill monitors are needed to cover a larger area (i.e. if the performer moves around). Avoid placing monitor speakers where they face the front of a microphone.

  If multiple monitor mixes are required, each performer may have their own monitor speaker(s).

Section 11 ….. Understanding Sound

1. The Decibel - dB

- The Decibel (dB) is the unit of measurement used in sound systems

- A Decibel (dB) describes a ration between two quantities expressed as a logarithm. Logarithms are used because our ears hear differences in loudness as a Log function.

- In simple terms, 3 dB represents twice as much power, and 10 dB represents 10 times as much power
  -2 * power = 3 dB
  -4 * power = 6 dB
  -8 * power = 9 dB
  -10 * power = 10 dB

- To make sound twice as LOUD requires 4 times as much power which is 6 dB
  -If you have a 100 Watt stereo, and you want one that is twice as loud, you need a 400 Watt stereo

- Loudness of a sound system is measured in dB of Sound Pressure Level (dB SPL)

- Loudness decreases by 6 dB (half as loud) every time the distance from the sound source is doubled
  -If you start 3 feet from the speaker and move back to 6 feet, it will sound half as loud (-6 dB)
  -If you move from 6 feet to 12 feet, it will reduce in volume by half again
  -By the time you are 48 feet from the speaker, the sound will be 1/16th as loud (-24 dB) as it was at 3 feet

2. Frequency Range of the Human Voice

- Voice range covers 300 Hz to 3500 Hz

- Most energy concentrated below 1000 Hz

- Vowels have most of their energy below 1000 Hz

- Vowels contain the "power and impact of the voice"

- Consonants have most of their energy above 1000 Hz

- Consonants are responsible for intelligibility

- Harmonics in voice can go above 3500 Hz

- Poor high frequency response reduces intelligibility

3. Recognizing the Frequency of Sounds

It is important for a sound technician to learn to recognize the frequency of sounds so you are able to quickly and correctly adjust equalization when there is a problem (i.e. feedback, hollowness, nasal sound, boomy, etc.)

When you have time alone with a sound system, put on some good quality music and play with the graphic equalizer to learn what frequency range each control affects. Memorize the tonal quality each control affects.

40-60 Hz	Boomy - a sound over abundant in low lows. These waves move a lot of air, hence
60-150 Hz	Fat - the octave above Boominess. Makes things sound big, but not earth- shaking.
125-250 Hz	Woofy - a somewhat nebulous term for sounds that are sort of "covered" - masked by low-end energy.
250-500 Hz	Puffy - is like an octave above Woofy. It's still sort of a cloud, but not as big.
200-400 Hz	Warm - obviously a positive characteristic often found between 200 and 400 Hz. Could easily degenerate into Woofiness or Puffiness if overdone.
500-1kHz	Boxy - seems to remind one of the sound in a small box-like room.
500-1kHzLow end of	
500-5kHz	Power range - mid-range band which contains the 1st and 2nd harmonics of most important sounds.
1.5-2.5 kHz	Telephony - accentuating the limited bandwidth characteristic commonly associated with telephones with a roll-off both above and below.
2.5-4 kHz	Cutting - Here, "cut" means to put an incisive "point" on the sound.
2.5 kHz	Punch - Accentuating this range punches through vocals.
3-6 kHz	Presence - Anywhere from 3-6 kHz can be used to make a sound more present.
7-10 kHz	Sibilance - Dangerous "s" sounds and lots of other trashiness can often be found at 7-10
10-12 kHz	Zizz - refers to a pleasantly biting high-end resonance (think of a "harpsichord"-type brightness found around 10-12 kHz.
12-15 kHz	Glass - A very translucent, but palpable brilliance associated with 12-15 kHz.
15-20 kHz	Sparkle - A real smooth stratospheric brilliance almost beyond hearing, but can certainly be sensed.
Above 10	Brightness - Most generally achieved by a global (shelving) EQ of everything above 10
Below 10	Darkness - The opposite of brightness (a general lack of highs at 10 kHz and beyond).
125-500 Hz	Muddiness - Actually a compound problem: Woofiness plus Puffiness (excess low end and also low mids).
125-500 Hz	Thinness - The opposite of Muddiness (a deficiency of lows and low mid frequencies).

4. Ear Sensitivity

- Sensitivity of most people's ears is relatively smooth between 500 Hz and 5000 Hz
- Ears are most sensitive to sounds between 3000 Hz and 4000 Hz
- Below 500 Hz and above 5000 Hz, hearing sensitivity drops off

- At louder listening levels (rock concerts), the frequency response of the ears becomes more equal over a wider range
  -This is why you really can't notice the bass or cymbals in quiet music, but they are quite evident in loud music
  -Many stereos have a "loudness" switch to compensate for this effect at low volumes

- A Sound Pressure Level (SPL) of 120 to 130 dB SPL is the threshold of PAIN for most people
  -Children and women are more sensitive to loud sounds than men

5. Dynamic Range

- Singers and instruments are capable of performing as quiet as 50 dB SPL and as loud as 110 dB SPL
  -This represents a 60 dB dynamic range from the quietest to loudest

- The quietest parts of a performance must still be kept louder than the room noise (called the noise floor)
  -The noise floor is typically 50 to 60 dB SPL
  -The quietest parts must be amplified enough that they can still be heard above the noise floor in the back of the room
  -Assuming 24 dB of loss from the front to the back of a 50 foot long room, the quietest parts would need to be amplified 24 dB to be heard in the back of the room
  -Note: It may not be possible to provide 24 dB of gain before feedback (see Feedback Control)
  -The speakers must reproduce the quietest parts at 74 dB SPL in order to be heard at the back of a 50 foot room at 50 dB SPL

- The loudest parts of the performance must not be so loud as to be obnoxious or painful
  -The loudest should not exceed 110 dB SPL
  -The loudest cannot exceed the capabilities of the amplifier and speaker system (also about 110 dB SPL)
  -However, since the quietest parts must be amplified 24 dB, the loudest parts (110 dB SPL) will also be amplified the same amount, obviously making them much TOO LOUD

- The useful dynamic range of the speaker system is limited by:
  -The quietest part of the program must be amplified to 74 dB SPL to be heard in the back of the room
  -The loudest part of the program exceed 110 dB SPL in the front of the room
  -This leaves a useful dynamic range of 110 - 74 or 36 dB, much less than the 60 dB dynamic range of a typical music group

- Solutions to the Dynamic Range problem:
  -Make the room quieter (requires expensive sound insulation)
  -Get a louder sound system (could annoy the audience)
  -Use a compressor / limiter circuit (expensive, especially in a system with many microphones and instruments in one system)
  -Have the sound engineer turn quiet parts up and turn loud parts down
  -***** Have the performers reduce the demand for dynamic range by getting closer to the mike on quiet parts and backing off from the mike on loud parts, similarly, have instrument players control their own volume according to the dynamic needs of the program

6. Feedback Control

- Feedback occurs when the sound from the microphone is amplified too much.

- Feedback is caused by a repeating circular process of a microphone picking up a sound from a speaker, the sound system amplifying it (too much), the speaker reproducing the sound again, and the microphone picking it up again.

- Feedback usually occurs at one frequency at a time. The frequency of the feedback is affected by:
  -Direction the microphone is facing
  -Distance between the microphone and speaker
  -Frequency response characteristics of the room
  -Equalization of the microphone channel
  -Equalization of the monitor speakers
  -Equalization of the main speakers

- Sound systems should be operated no louder than 6 dB before the beginning of feedback (that is half as loud as when feedback starts)

- Operating closer to the feedback point causes a "hollow" or "ringing" sound

- Adding more microphones increases feedback problems. Every time you double the number of microphones, the maximum gain before feedback is reduced by 3 dB. Apparent loudness of each mike is cut in half when the number of mikes is increased 4 times.

- To go from 1 to 4 mikes halves the maximum volume of each mike before feedback.

- To go from 1 to 16 mikes quarters the maximum volume of each mike before feedback.

- Feedback is controlled by:
  -Using directional microphones and carefully aiming them away from monitor and main speakers to reduce feedback
  -Performers must be careful not to re-aim mikes towards speakers during performance
  -Performers must be careful when hand holding mikes not to point them towards the monitor speakers
  -Decreasing the distance between the sound source (performer) and the microphone (so the microphone does not need to be as loud)
  -Increasing the distance between the speakers and the microphones
  -Using equalizers to reduce the system's gain at the frequencies where the feedback occurs
  -Installing acoustic dampening material in the room to reduce sound reflections back to the microphones (expensive solution)

7. Factors Influencing Clarity and Intelligibility

- High monitor levels on stage get into the microphones and muffle the sound because the monitor sound is out of phase with the original sound (don't set louder than necessary)

- High monitor levels can also cause sound to be hollow or ringing

- Instruments playing music louder than necessary on stage causes too much music to be picked up by vocal mikes, making the music muddy (keep music as quiet on stage)

- Excessive use of equalization to prevent feedback effects the clarity of the overall sound (again, don't set monitor louder than necessary)

- Acoustical characteristics of the room (reverberation) affects the intelligibility of the sound. Large flat hard surfaces reflect sound which is out of phase with the original sound. Acoustical treatment of walls and ceilings is desirable.

- When performers are too far from the mike, the mike gain must be increased which causes pickup of more background noise and muddies the sound (the optimum distance from the microphones is 6" except for solos where "quiet" singers should be 2" from mike)

- Microphones aimed at monitor speakers, instrument speakers, or drums (tilt mikes up slightly so they point at a "quiet" ceiling rather than behind you at a guitar amp or drummer)

- Breath noises and "popping" on solos (tilt mike up and sing/speak over the top of it, stay 6" from mike)

Section 12 ….. Set-Up of a Sound System

- Make a diagram of the stage layout with the location of all people and all instruments identified and all mixer inputs numbered
- Decide what equipment needs to be transported
- Carefully pack equipment for transport
- Unload equipment at destination
- Place all equipment (mike stands, speakers, amps, mixer) in its desired location
- Resolve equipment and people placement problems
- Run "snake" from stage to mixer (if snake is being used)
- Put mikes on stands (with stands set too high so that there will be enough slack in the cords)
- Select correct length cord for every mike, instrument, and speaker and lay out cords
- Label all channels on the mixer board
- Run all mike, speaker, power, and instrument cords NEATLY
- Connect mixer to main and monitor amplifiers
- Connect auxiliary equipment such as FM microphone receivers, equalizer, effects unit, taper recorders, and CD
- Power up all equipment
- Perform a sound check with a tape or a CD, confirming that all main and monitor speakers work
- Check all mixer controls to see that they are in the correct position. Set channel trim pots to expected operating level (from previous experience). Set equalization flat (unless there is a reason not to). Set all monitor send controls to half-way position (unless there is a reason to set them different). Determine which channels should be assigned to each sub-group. Set any other controls unique to the mixer to desired starting point. Start with all channel Faders off!
- Perform a sound check on all microphones and instruments (one at a time) to see that they work and are connected to the correct channel
- Carefully position all microphones and speakers in their final positions
- If time permits, use the graphic equalizer spectrum analyzer, a pink noise source, and a calibrated microphone to analyze the room acoustics and set the equalizer to compensate
- Play a good CD through the main speakers and adjust equalization ONLY if needed so that it "sounds good"
- With master control and monitor control off, set all mikes at their expected working level (normally the "0" position on the channel sliders), then slowly turn up monitor and master levels
- "Ring Out" monitor system using graphic equalizer spectrum analyzer

- "Ring Out" main speaker system ONLY if necessary to get enough gain before feedback (this will effect the overall tonal quality of the performance)
- Perform a sound check with all people in their final positions
- Watch carefully for any overload lights and adjust channel trim pots if necessary.
- Listen to each input channel (mike or instrument) individually and adjust its channel trim pot to balance with the rest of the system
- Listen to what the performers hear on stage and adjust monitor mix to suit the performers
- Listen to tonal quality of the sound from the position of the audience and adjust ONLY if necessary

Section 13 ….. Operating a Sound System

- You should have completed sound check before the audience arrives
- If appropriate, play some background music as the audience enters
- Once the room is full of people, the sound is going to be different!
- If a lavaliere mike is being used, make sure the performer is wearing it as high as possible, centered, and facing straight up
- If battery operated mikes are being used, make sure they are turned on
- Keep all mike channels off until the program begins
- When the program begins, listen first for correct overall volume and make any necessary adjustment quickly
- Check to see that the proper lights are turned on. Are the main lights supposed to be dimmed?
- Listen to see that all performers in the program can be heard and that none are significantly too loud or too soft
- Look at the performers to see if they look at ease. Are they straining to hear the monitors? Are they backing away from monitors that are too loud?
- Listen for serious tonal quality problems - too bassy, too sharp, or hollow sounding
- Follow your program/agenda/Q-sheet to make sure you don't miss any queues
- As time permits, use a headset to listen to the PFL for each channel and make any necessary adjustments to volume or equalization
- If it is a musical program, listen constantly for a proper blend
- Stay alert!

Section 14 ….. How the Performer can Help

- Help pack up equipment for transport when we go on the road
- Help get out the equipment for practice
- Learn where equipment is normally placed so that you can set it at location which will probably be correct
- Position mike stands higher than normal for set-up (so the cords will be long enough later)
- Put mikes on the mike stands (the type of clip indicates which type of mike to use)
- Set up music stands, guitar stands, and keyboard stand
- Help with running mike, speaker, and power cords (AFTER they have ALL been laid out)
- Get guitars and horns tuned well before warm-up starts
- Help with setting guitar and keyboard levels BEFORE warm-up starts
- Show up on time for warm-up and be in your designated position
- Position the mike at the right height for you and any others sharing the mike, but don't change the up/down angle of the mike or point it to the side (because it may pick up sound from the monitor speakers and cause feedback)
- Tell sound man if you have any problems he may be able to fix, especially monitor problems
- Tell sound man if you have any problems with your designated position
- If you are alone on a mike, try to stay a constant, consistent, never changing 6" from the mike, except for solos or speaking parts when you should usually be 2" from the mike
- Sing at a consistent volume (except where dynamics of the music require change). DON'T get closer to the mike and sing louder on songs you know well. DON'T back up and sing quieter on songs you know less well.
- When singing SOLO, control your own dynamic range. Get closer to the mike on quiet parts and back off from the mike on loud parts.
- If you are grouped on a mike with others, allow the quietest singer to be centered on the mike, louder singer(s) may be off to the side a little. If all singers on a mike are the same loudness, they should be equidistant from the mike. Stay as close to the mike as possible. If you are more than 15" from the mike, you cannot be heard.
- DON'T move to another mike unless absolutely necessary. The volume of every mike has been balanced for the people who normally sing on that mike. If you must move to another mike, tell the sound man (when convenient), don't get closer than 6", and don't sing louder than you normally would.
- If you are normally grouped on a mike and the other person(s) leave for one song, don't move closer to the mike than you would have been if the others were still there.
- After practice and after the show, unplug your mike and put it away, also put your mike stand and music stand down the whole way and carry them out.

- Roll or carry other equipment out.
- Help wrap up cords. THIS MUST BE DONE NEATLY! If the cords are not wrapped neatly, they will knot when used the next time (increasing setup time) and they will not lay flat (causing a trip hazard).
- After wrapping a cord, place it in a pile with similar cords. Group speaker cords (heavy cords with 1/4" plugs) together and group mike cords (XLR connectors) together by the color of their tie string.

Section 15 ….. Trouble Shooting the Sound System

What do you do if the sound system doesn't work?

-First, check the obvious!

-Are the mixer, equalizers, amplifiers, and all other equipment plugged in and turned on?

-Are the microphones and instruments connected to the mixer correctly?

-Check all mixer controls for proper settings: channel trim pot, pad switch, line/mike switch, sub-group select, left/right pan, channel on/mute, channel fader, sub-group on/mute, sub group left/right pan, sub-group fader, main on/mute, main fader

-Are both ends of the snake connected correctly?

-Is the mixer output connected to the equalizer input?

-Is the equalizer output connected to the power amplifier input?

-Is the power amplifier output connected to the speakers?

-Is the power amplifier gain control set correctly?

-Are the pilot lights lit on all equipment?

-Are there any blow fuses?

-When you speak into a mike, does it indicate on the mixer's meter?

-Can it be heard on the headphones when the PFL switch is operated?

-Does anything work as a starting point?

-Can the trouble be isolated to one microphone, instrument, or speaker?

-Try replacing microphone, cross-connect, or speaker cables (in case there is a broken wire)

-Try playing a tape or CD

Section 16 ….. Everything Else!

Sound Technician Supplies:

Adapters: A sound technician should have a good variety of Radio Shack audio connector adapters to convert between 1/8", RCA, 1/4", XLR, male/female, and mono/stereo connectors

Cables: A variety of short (3') cables are needed to interface between mixer and FM receivers, tape deck, CD, equalizer, effects processor, and amplifiers. Most use 1/4" to 1/4" cords, but some use RCA, XLR, or 1/8"

Tools: Variety of screw drivers, wire cutters, adjustable wrench, hammer

Cable Tester: To look for broken wires in cables (it happens more often than you'd think)

Batteries: Keep several 9V Alkaline batteries for battery powered FM mikes

Flashlight:

Guitar Tuner

Patience and a cool temper

AC Power:

If your total sound system does not require more than 15 amps, then connect EVERYTHING to one outlet to reduce the possibility of ground hum.

You need several AC power ground-lift adapters (3-prong to 2-prong adapters), a few multi-outlet adapters, several multi-outlet power strips with surge protectors, and a variety of 3-conductor extension cords (of different lengths).

You should have a 3-prong AC power tester and always test an AC power outlet before you use it (prevents equipment damage due to faulty wiring).

If you do get a ground hum, use your power tester to test the power feed to every unit first, then try a ground lift adapter on each unit of the system, one at a time, until you find the source of the hum.

Intercom System:

An intercom system may be helpful if it is necessary for the Sound Technician to talk to Stage Crew or Lighting Technician

Projection System:

A projection system may be useful if song words are to be displayed for the audience

Possibilities include overhead projector, slide projector, or projection video

Projecting on a movie screen will give a brighter image

Carefully aim other lights so they don't shine on the screen

Documentation and Labeling:

Label all permanent wires

Put a strip of tape across the mixer and label all channels

Label all equalizers, effects units, amplifiers, etc. as to what they are connected to (i.e. Left Main Amp).

Controls which are always to be left set at the same place should be marked with a dot at the proper setting. (white-out works well)

Engrave a permanent identity name or number on all expensive equipment (to identify it in case of theft).

Buying Equipment:

Read audio magazines, go to various performances and see what equipment other people use, ask other sound technicians what equipment they like or don't like, go to music stores and ask questions

Before buying any equipment, write down all requirements for the sound system

Budget, Evaluate the requirements in relation to the available budget
Determine which items are most important to have first

Insurance / Inventory / Labeling:

Make sure that all equipment is covered by insurance (you may have to list all equipment)

Keep an accurate, up to date, and complete inventory of all sound equipment with manufacturer, model, and serial numbers

An identification name or number should be engraved on the equipment in a visible, but not unsightly location

Other Topics:

Lighting Control:
Video Taping:
Audio Taping:
Direct Boxes
Impedance Matching
70-Volt Line
Clear-com
Talk-Back Mike
Light for the sound console
Soldering iron
Sound feed for Video, ambience
Sound feed for Radio, ambience
Recording
Video, camera position, sound source, ambience
Grounding, hiss, hum, buzz, dimmers
Setup, connecting the wires
Tear down, wrapping the wires, shoe strings
Microphone care
Pink noise generator, Real-time Spectrum Analyzer
Guitar tuner, importance of tuning