

Sound Systems for Services

This article is intended for *the start-up congregation* that may not yet have a permanent facility in which to meet, but wants their messages and music to be heard to benefit all who come. It explains when a sound system is necessary and what a person needs to know to select and operate one. Using this information can help an amateur produce pleasing results as well as save hundreds of hours of frustration and thousands of dollars in equipment.

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Do You Need a Sound System?

Sound systems are also called “public address” or “P.A.” systems. A good system can make the difference between a service where everyone can comfortably hear and pay attention, and one where people are frustrated by the sound. But money spent on a sound system that is not what you need can be a big waste. When a new congregation is just beginning to meet, it is best if they borrow, rent, or do without a sound system for the first few meetings—until they know about how many people they will have and where they might be meeting. Resolve the major spiritual issues first, and then obtain sound equipment to support the decision you have made. (It is not sensible for two groups of differently-thinking people to continue meeting together just because they both own part of a sound system.) The goal of any sound system is to render the appropriate service needed by a congregation now, with the possibility of easy expansion for the future.

If you are meeting in a home or a small hall without a sound system, and you have had no previous problems with people hearing, then you do not need a sound system. If you desire only to record the service, but not amplify it, then see the sections on **Microphones and Stands** and **Tape and CD Recorders**.

Even small groups may have soft-spoken teachers that need amplification to be heard. (We use the word *teacher* in this article to denote any person speaking; we reserve the word *speaker* here for the box-like thing that sound comes out of.) Other small groups may have nearby noise (traffic, trains or their own youngsters) and need amplification so the messages can always be understood. Also, small groups sometimes combine together in a larger hall for other special occasions for which they may need a sound system. Finally, many singers rely on amplification—not many sing loud enough to be heard over pianos or other accompaniment instruments. What you need depends greatly on the nature of the room you use, the number of people, the teachers and the music. Some rooms (especially movie theaters) are very sound absorbent and amplification is necessary for all but

the loudest of teachers.

If there are people in your congregation that really understand sound systems or at least have a gift for working with electrical and technical things, let them make the decisions on what is really needed. Unfortunately, for every person that really understands sound systems, there seem to be about five more who work with them and claim to understand them even though they are lacking in much fundamental knowledge. This writer has met many people that either set up or sold sound systems for a living, but did not understand many of the principles covered in this short article. The danger of dealing with people like this is that you can find yourself spending money on equipment that will not work correctly for you, or replacing equipment that could work, but is simply not being used properly. If you read this article, then talk to a “sound system” person (be it a person in your congregation or a salesman) that seems to be totally unfamiliar with some of the concepts written in this article, then encourage them to read this article or look for someone else to help make your decisions.

Unless you are certain that you will be in a specific building for a long time, you probably want to obtain a portable sound system rather than a permanently-installed one. The best place to obtain such equipment is usually an audio specialty store or a music store that deals in PA systems. The advantages of local purchase is that you can get hands-on experience with the equipment in the store before you buy it, the sales people can show you exactly how to operate your equipment, and there is a minimum of difficulty replacing any equipment that fails. You do not need to pay the asking price in most of these stores—if you tell them you are a religious group and indicate that you will do repeat business, they often can give you 10 to 30% off their posted prices. Unfortunately, these stores are usually found only in larger cities. Radio Shack stores are available almost everywhere, but tend to have only lower-end P.A. equipment (though they are a convenient source of audio connectors and parts.) Mail order is another option, try: Carvin (California) 800-854-2235, Freeport Music (New York) 800-624-5141, Long’s Electronics (Alabama) 800-633-3410.

Most of the rest of this article covers equipment that you may need and how to use it. But first, an understanding of sound and frequency is vital—all the other sections will refer to information in the following section:

What Is Sound?

When acquiring a sound system, there may be a tendency to say, “Skip the technical stuff and tell me what equipment I need.” However, this writer has witnessed many thousands of dollars and hundreds of hours wasted on sound systems that did not provide the desired results because there was a basic lack of understanding of the nature of sound and the desired results to be achieved. This lack of understanding is frequently found in both those who use and sell equipment. Most electronics store clerks have never been introduced to the information in this article—only in professional audio stores would one be likely to find individuals who understand it. Therefore, it is important that you, a potential sound system purchaser/operator, understand it.

Sound is nothing more than vibration of air molecules. If you bang on a table, you are causing it to vibrate, and that vibration of the table is transmitted to the air, and then your ears hear it. The vibration stops shortly after it starts, and the sound goes away. The table is not designed to produce any particular sound or for any duration. However, if you were to strike a string on a guitar or violin, you would hear a distinct note that would last for several seconds. These instruments are expertly designed to produce certain musical pitches and to transmit them into the air where they may be pleasantly heard. Each string on these instruments produces a distinct musical pitch. Similarly, if you were to blow across the mouthpiece of a flute, the air would go in and out of the flute rapidly, producing a musical pitch that can be controlled by the keys on the flute. This is not going to be a music lesson, but we are starting with these musical sounds because they are the *simplest* sounds.

The concept of pitch, or **frequency of sound** is vitally important to understanding sound. “Pitch” is the term used in music, “frequency” is the term used in science and in the specification of sound systems. What is sound frequency? Air molecules can vibrate slowly back and forth (a low pitch), or very quickly back and forth (a high pitch). The **lowest** frequency an average

person can hear is **20 vibrations per second**, also called 20 hertz (abbreviated 20hz). (Elephants can hear down to about 10hz; so anyone who wants to attract or repel elephants from their congregation might consider this.) The **highest** frequency people can hear is about **20,000 vibrations per second**, also called 20,000 hertz or 20 kilohertz (abbreviated 20,000hz or 20khz). Many animals can hear higher frequencies—a dogs-can hear “dog whistles”, be people cannot.

To understand the range of sounds that we can hear, suppose that we had a piano keyboard that was set to play the normal piano pitches, but with sound of a flute (one can actually purchase an electronic keyboard to do this). The reason we chose a flute is because it is very close to a pure tone—just one frequency. The lowest note, all the way to the left, on this piano (a musical “A”) is 27.5hz—near the low end of human hearing. If our piano had five more keys to the left, the note would be 20.6hz, at the lower limit of human hearing. Middle “C” on the piano is 261.6hz. The highest note, all the way to the right, is 4186hz (or about 4khz). If this piano keyboard had 27 more keys to the right side, its highest note would be 19,912hz (or 19.9khz) at the upper limit of human hearing. As you can see, the middle pitch that we can hear is **not** half way between the highest and lowest (half way between 20 and 20,000 is 10,010). Sound frequencies operate in what mathematicians call a “logarithmic scale”. If we want to produce a sound that is an “octave higher” as the previous one, we do not add some number of hertz (hz) to it, but we multiply its frequency by two. Here are some typical ranges of the *fundamental frequencies* of common sounds.

30 to 150 hz:	bass instruments and drums:
60 to 400 hz	men’s voices
120 to 1000 hz	women’s voices
200-3000 hz:	typical musical instruments
200hz-20khz	cymbals, scraping noises,

We must explain one more item of essential complexity before ending this section on sound: **Very few sounds in nature are pure tones consisting of only a single frequency.** The closest sounds to a single frequency are flutes, organs making flute sounds and various kinds of whistles—including the wind whistling around the corner of a building. Most other musical in-

struments, such as a violin, trumpet or saxophone produce a great variety of **overtones**—frequencies that are multiples of the **fundamental** frequency (main or lowest frequency). For example, if a violin plays the musical “G” above “middle C”, the fundamental frequency will be about 400hz. However, that same note will include a certain amount of 800hz, 1200hz, 1600hz, 2000hz, and every other multiple of 400 clear up to 20khz! Violins are distinguished by their many overtones—that is why they do not sound good when played through poor sound systems. Trumpets, saxophones and other musical instruments also produce overtones, but their patterns of overtones are all different. Our ears are good at distinguishing these things—that is the way that we can distinguish one instrument from another. A saxophone does not produce many overtones in the 8khz to 20khz range, so it will still sound like a saxophone even through a poor sound system.

Most sounds consist of a fundamental frequency and “overtones” (multiples of the fundamental). That is why pianos and virtually all other musical instruments do not provide any means for playing notes above about 4khz. Frequencies this high sound mostly like high-pitched squeaks. The range of frequencies that we hear between 4khz and 20khz is primarily useful as **overtones** for distinguishing the character of lower frequency sounds.

Human speech or the simple banging of two hard objects together produce even more complex mixtures of frequencies than musical instruments—many of which are not multiples of each other. When a sound system does not adequately reproduce all of the overtones (higher frequencies) of an instrument, you can still hear the melody, but the quality of the sound will just be poor. But if the overtones of human speech are poorly reproduced, you can hear sound, but may not be able to understand what is being said. Similarly, complex sounds, such as the cymbals used by a band often have a sharp sizzling sound when heard live, but sound like pots and pans clanging through a poor sound system. Cymbals produce a complex set of frequencies

up to 20khz (the limit of human hearing), and when those frequencies are lost in poor sound reproduction, they sound like pots and pans.

We have explained that musical instruments, voices, and the striking of a hard object produce certain frequencies of sound. These sounds are often unique enough that one can learn to hear the difference between musical instruments, between individual’s voices, and between a cup, fork or pan falling to the floor. Also, each room has a unique set of frequencies associated with it. If you clap your hands in the same way, but in different rooms, you will notice a slightly different sound of the echo that occurs after the clap—due to the different frequencies that different

rooms emphasize. However, when we look for quality sound systems, we look for something completely opposite. We look for equipment that **does not have certain frequencies emphasized, but that will reproduce all frequencies equally.**

When a singer sings or a teacher speaks into a microphone, we usually want to hear the exact same combination of frequencies coming out of the system’s speakers that went into

the microphone. We do not want an object with unique characteristic frequencies, but objects that treat all frequencies the same. In technical terms, this is called a “flat response”. The ideal microphone, speaker, tape recorder, or amplifier should reproduce all frequencies equally well throughout the range of human hearing. In practice, amplifiers and recorders can now do this quite well. A few microphones and speakers can, but they will cost quite a bit. Devices called “equalizers” can be used to compensate for some frequency response problems. Learning to make the right compromise is important.

Another important concept of sound is technically called “amplitude”—it simply means the intensity or the loudness of the sound. Everyone naturally understands the difference between a loud and a quiet sound. It is important to realize that frequency and volume are completely independent: any sound frequency can be created at any possible volume.

To sum up the essential elements of sound relevant to selecting sound systems:

Do not let things like **440hz** scare you. An “hz” (“hertz”) is simply one vibration of sound per second. 440hz is the musical “A” above middle “C” to which orchestras “tune up”. The next higher “A” is 880hz, then next lower “A” is 220hz—you can keep multiplying or dividing by 2. Humans can hear between 20hz and 20,000hz.

1. Humans hear sound in frequencies between 20hz and 20,000hz (20khz).
2. Most musical instruments produce a fundamental frequency in the 30hz to 4000hz range, along with overtones that may extend up to 20khz. Music can be appreciated without all of these frequencies present, but it will not sound as good as live music.
3. Human voices produce fundamental frequencies in the 60hz to 1000hz range with complex overtones extending up to about 12,000hz. Hearing these overtones, especially in the 1000 to 2000hz range, is vital to understanding speech.
4. The goal of sound systems is to amplify or record the full spectrum of frequencies as they were originally produced. The equipment needed to do this will vary greatly depending upon many different factors.

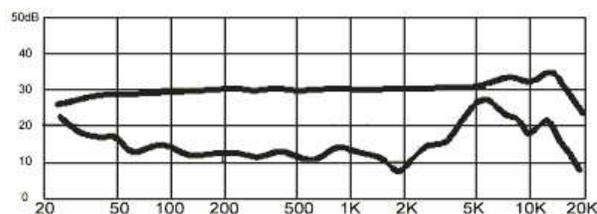
Microphones

A poor-quality microphone is likely to create more trouble for a sound system than any other single component. The purpose of a microphone is to convert sound into tiny electrical voltages that can be amplified by an amplifier. A perfect microphone would produce the same voltage for the same loudness of sound, for any frequency of sound. A microphone like this is said to have a “flat response”.

Poor microphones have a “peaked” or “peaky” response—often producing 2, 3 or even 10 times as much voltage for one particular frequency range. These microphones will sound “tinny” and have a much greater tendency to “feedback.” **Feedback** is the deafening high pitch squeal or roaring sound that occurs when the sound coming out of a sound system’s speakers is picked up by the microphone, and amplified again, even louder. (The sound continues to go from the speakers, into the microphone, to the amplifier, back to the speakers even louder, etc. It will continually get louder until the amplifier reaches its maximum output.) Poor microphones will “feedback” at their peaky response frequency. The easiest way to eliminate feedback is to turn down the volume, but with a peaky microphone, it must often be turned down so low that you cannot adequately hear the person. You simply want to avoid microphones with “peaky” response.

Another characteristic of good microphones

is that they will respond to higher frequencies than poor microphones. The best microphones will reproduce frequencies up to 20,000hz (20khz), which is the limit of human hearing. A “high end” response above 14khz will work quite well and anything above 9khz is usable for human voices. In the curves below, the top graph is very flat, with only a small peak around the 10khz area—a minor problem easy to fix with an “equalizer”. The bottom graph would have a very “peaky” response in the 5khz area—it would feedback there easily and have a harsh sound overall. Both graphs tend to drop off greatly from 15khz to 20khz, but that would be a problem only if they were to be used for cymbals or other instruments with high frequencies.



In general, more expensive microphones will have a flatter response and better high frequency response, but do not assume you are getting quality just because you are paying a lot. Good microphones used to cost hundreds of dollars, but they are now designed by computer and may cost as little as \$50. But, some \$50 mics are nearly worthless. Microphones that “come with” low to medium quality equipment are often very cheap and not worth using.

Before buying a microphone, ask the sales person to plug it into a good PA system. It should sound natural and produce plenty of volume without feedback. Try comparing its sound with that of the best microphone in the store. Try making an “s” (hissing) sound in both microphones, holding them to the side of your mouth so the air does not blow into them.

There are three other important options that you will have when buying a microphone. Some microphones are designed to clip onto clothing (“lapel mics”) and others are designed to be held either by hand or on a stand. Lapel mics are useful for teachers that move around a lot. Also, they reduce obstruction for video taped messages. However, since most services involve a number of people that speak, pray, sing, etc., it is usually impractical to use a lapel mike for all of a service. Your first microphone should be

one that mounts on a stand.

Some microphones are **unidirectional** (they pick up sound mostly in one direction) and others are **omni-directional** (they pick up sound equally in all directions). If your purpose is simply to record services—including questions and comments from everyone in the room, use an omni-directional microphone. For most other speaking and singing situations, you want to use a unidirectional microphone (sometimes called just “directional” or “cardioid pickup pattern”). These microphones greatly reduce the amount of unwanted noise and “feedback”.

The two most common internal constructions of microphones are **dynamic** and **condenser**. Condenser microphones can be very small and have excellent high-frequency response. However, they require a battery or special wiring in the mixer or amplifier to provide power to them (called “phantom power”). If you are going to use condenser mics, it is best if they use “phantom power” and that your amplifier or mixer be able to provide it. If not, you must make sure that the microphones are turned off when not in use and that you always have a few extra batteries available. It is very easy to accidentally leave these microphones turned on and then find dead batteries the next time you want to use them.

Microphones come with two different types of cords and plugs. Cheaper ones use the common ¼" plug that is also  found on guitar cords or large stereo headphones. These are “unbalanced line” microphones, and can be either “high impedance” or “low impedance”, but “high” is much more common today.

Microphones with a 3-pin XLR plug are well worth any extra cost. These are “low impedance” and “balanced lines” microphones. They use a technically superior method of electrically transmitting sound that greatly reduces the amount of hum and noise. Also, very long lengths of cable can be used without loss of sound quality. Even if your amplifier does not accept a 3-pin XLR plug, you can buy an “XLR to ¼” conversion transformer”  from Radio Shack or other audio stores for about \$14. Plug

it into your amplifier gives you nearly all of the benefits of the superior cabling system. Keeping noise out of sound systems is very important—if people like the messages but are irritated by a noisy sound system, they will frequently go somewhere else.

Finally, you may want to consider wireless microphones. A wireless lavalier microphone is very helpful when teachers walk around a lot. These units contain a microphone with a transmitter and a small receiving unit that plugs into the amplifier. Inexpensive wireless microphones may be available from Radio Shack and other stores for less than \$100. However, they may be prone to noise or drop-outs (the sound is lost for a few seconds)—especially in crowded business areas where there may be interference from other radio devices. A major audio store or the mail-order store that we listed will have wireless microphones in the \$150 to \$500 range that will be reliable.

Hand-held wireless microphones also have uses. Compared to wireless lavalier microphones, they produce a better sound quality and are much less likely to have “feedback problems”. Many singers prefer a wireless hand-held mike. They are a disadvantage for teachers to have to hold them, but they work very well for assistants to use, walking around the congregation to solicit questions and comments in interactive sessions. Hand-held wireless microphones cost about the same as wireless lavalier microphones.

The big disadvantage of wireless mics is their cost and the fact that they consume a 9-volt battery for every 3 to 10 hours of use. Using many wireless mics that are left “on for the whole service” could easily cost \$15 a meeting just for batteries.

Microphone Stands

Many lecterns already contain a place to attach a microphone. If you have a lectern that you always use, it is best to buy a microphone stand designed to fit on it. If you do not always have the same lectern, try a floor stand. A boom attachment (a metal tube that can extend at an angle to the main upright stand) or long “goose-neck” (a strong but flexible tube) is essential for getting the microphone close to the teacher over a lectern. These stands are also good for musicians who are singing while they are playing the piano, guitar etc. A straight microphone

stand is useful primarily for singers or teachers who do not use lecterns. Booms and gooseneck attachments can be added to strait microphone stands, but it is usually cheaper to buy a "boom stand" than buy the pieces separately.

Another important factor that relates to all microphone stands (and to some degree, microphones) is audio insulation of the microphone from whatever the stand is resting on. If you tap on the case of most microphones (when they are hooked up to a sound system), you will hear a fairly loud sound from the speakers. If the microphone is tightly connected to its holder, which is tightly connected to a stand, which is tightly connected to a lectern or sitting on a noisy wood floor; a lot of noise may get into your microphone through the stand. You know you have this problem if you hear loud thumps in your sound system whenever someone hits the lectern or bumps the floor near the microphone. Low frequency feedback (rumble) can result from the worst cases. You can buy microphone clips designed to stop this problem, but they may be large and clumsy. Some stands have acoustic insulators in the tubing or thick rubber feet to prevent floor noise. Placing the microphone stand on a small piece of thick rug can largely eliminate floor noise. In the worst cases (a meeting room is next to a gym where people are continually hitting the floor), you may need to use several of these techniques to eliminate floor noise.

Amplifiers and Mixers

An **amplifier** takes the small electrical signal from a microphone or tape player and turns it into a powerful electrical signal that can be connected to speakers and easily heard. A **mixer** is a device that takes small electrical signals from many sources and allows you to combine them in any manner desired. A mixer by itself would be useful for making tape recordings, but an amplifier is needed for "live sound"—people hearing what is being said (or sung) as it is happening. Frequently, the amplifiers and mixers are contained in the same unit, called a "**powered mixer**". Also, many pieces of equipment sold as "amplifiers" also have built-in mixing facilities. For most church congregations, there is usually little savings or convenience in purchasing a separate *mixer* and *amplifier*. However, if a congregation already has access to one or the other of the two, they can save money by

buying the other.

Any amplifier produced within the last 15 years, as long as it has *sufficient power and audio connections*, will probably work. (Modern electronic design has made this economically possible; such quality was expensive 20 or more years ago.) The necessary **audio connections** vary quite a bit depending upon what you are trying to do. If you have a separate mixer, all your amplifier needs to do is connect to the mixer output—that is usually easy. But the mixer (whether it contains an amplifier or not) must be able to accept all of the various inputs that you need, and provide all of the outputs that you need.

There are three different power-levels of signals that mixers and amplifiers deal with. The weakest is **microphone level**, which may be as little as millionths of a watt. (At the microphone level, there are "high impedance" and "low impedance" signals—which are similar in power, but have different voltage and current levels). The next is **line level**, which is used to connect various pieces of electronic equipment, and is in the thousandths to tenths of a watt range. Finally, there are **speaker level** signals, which can range from less than a watt to hundreds even thousands of watts. It is best to only connect signals of the same level. Unfortunately, all of these levels of signals sometimes use the same types of audio plugs (male) and jacks (female)—so nothing stops you from making connections that will not work. If one plugs a microphone into the wrong impedance input, or into something designed for line level or speaker level input, there simply will not be enough volume. Conversely, if one connects a



speaker output to something designed for a microphone or line level input, the signal may be-

come distorted because the device is not able to handle that strong of a signal. In other cases, it may work fine, but instead of being able to adjust a volume control through its entire range, one may find that the maximum volume is already achieved when the volume control is turned to its first notch. Most equipment has “speaker level” and “microphone level” connections clearly labeled; the rest are usually assumed to be line level.

Amplifiers and mixers come with all kinds of features, some of which you may need and some which you probably do not. The single most obvious feature is the number of channels. A “**channel**” is section of the equipment that contains an input jack with associated volume control, tone and special effects controls. You will need a separate input “channel” for each microphone that you use. Some channels are for only microphone level or only line level; others can be switched between the two. Some mixers have an input “trim” control—a pre-volume control that can adjust all the way from microphone to speaker level.

Units tend to be sold with 4, 6, 8, 12 or more channels. For a small service, 4 will be plenty. If you have singing groups that would use 3 or 4 microphones now, then you should plan to get more channels. Cheaper mixers may not have tone controls for each channel. This can work if you only use one or two mics and have master tone controls on your amplifier. But with several mics used for different purposes (speakers, singers, instruments) separate tone controls will be needed.

Tape or CD input and output connections are a necessary feature if you are going to play and/or make tapes, though any line-level input can be used for a tape/CD input. Many systems also have a built-in reverberation feature. This feature is not necessary, but it will add depth and warmth to singer’s voices. Some systems also have a separate “monitor” mixing control on each channel. This is very helpful for larger singing groups—see the section on **Remote Speakers and Monitor Speakers**. Also, more expensive units can do stereo mixing—the ability to send any sound to the right speaker, left speaker, or any combination of both (by using “pan” controls). Using a stereo mixer and stereo amplifier is noticeably better for playing back pre-recorded stereo tapes and CDs, and can im-

prove artistic presentation of live music. If you plan to record music someday with your system, stereo is essential. But for speaking and occasional music, a mono (“non-stereo”) system will be simpler and cheaper to use..

It is best if an amplifier uses ¼" plugs to connect speakers. It is the most common system and the most convenient for quick setup and take-down. Systems that use simple wire connections might be slightly cheaper, but they are slower to connect and wire gradually breaks.

Home stereo amplifiers with a power of 10-watts or more will probably work for someone talking. A **watt** is a unit used to measure electrical power. (When examining the power of an amplifier, make sure you are reading how much power it delivers to its speakers—there may also be a rating in “watts” for power consumed from the electrical outlet, which will be considerably more than the usable audio output. Also, some amplifiers are rated in “peak” power which can be 3 times as high as their **RMS power rating**—use only RMS power ratings for comparison.) To amplify multiple singers and/or musical instruments, you will need an amplifier with more output power. About 50 watts is fine for conservative music, but you may need 200 watts or more to make loud “joyful noise” music free of distortion. You know an amplifier is underpowered when you hear distortion (raspiness) only during the loudest part of music or speaking. (If you hear distortion all of the time, some other problem is occurring.)

Amplifiers should have a sturdy case, especially if you must move it every week. Covered controls or controls recessed into the case of the amplifier prevent their accidental breakage during transport. Home stereo amplifiers and tape decks are not designed for continual moving—it is best if you use their original shipping cartons or some other kind of padded box.

Lastly, some “amplifiers” contain speakers built into them. These units are usually designed for guitars and electronic musical instruments. If you already have access to one and if it works well for you as a P.A. system, use it and save some money. However, it is **not** a good idea to purchase such an amplifier instead of a P.A. system because they have several inherent disadvantages: 1) The speaker cannot be separated from the amplifier; if someone needs to adjust the volume while the service is in progress, they

must walk up to the front of the room (where the speaker should be placed). 2) If a CD player is connected to such units, it usually should be done with a short cord, requiring CDs to be changed at the front of the room which can be distracting. 3) Speaker/amplifier units sometimes lack the necessary connections for CD's and recording. 4) The speakers in such units are often not designed with the level frequency response needed to avoid feedback when using microphones. 5) It is difficult to set the speakers up high—if the whole unit is set high, the amplifier controls become inaccessible.

Speakers

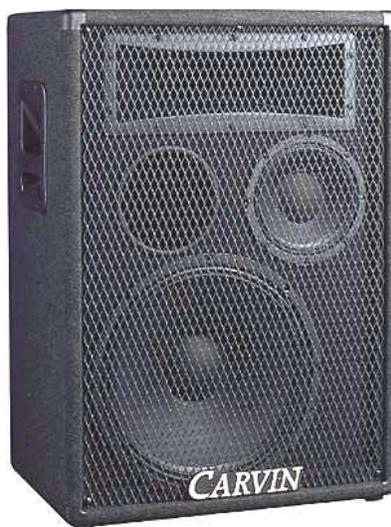
Speakers are the second most critical part of a PA system. But since a good speaker costs more than a good microphone, it is the most important purchase decision. The purpose of a speaker is to convert electrical energy from the amplifier into sound. The perfect speaker would be much like the perfect microphone: it would convert a certain amount of voltage to the same sound volume at every frequency (musical pitch). Like a microphone, this would be called a “flat response.” Unfortunately, no such speaker exists. It is physically too difficult for one device to adequately reproduce both very low (bass) and very high (treble) sounds. Consequently, most speaker systems have two or three “driver” elements in them: one for low sounds (a “woofer”), sometimes one for mid-range sounds, and one for high sounds (a “tweeter”). A “crossover network” is used to route the various electrical frequencies to the right “driver” element that can change it to sound.

As with microphones, a speaker can have a “peaky” response—it may generate a much louder volume for a narrow range of frequencies (pitches), resulting in “feedback” at that frequency. Much more emphasis on “flat response” goes into PA speaker design, compared to home stereo or musical instrument speaker design. The more expensive speakers tend to have a flatter response, but, like microphones, spending more

money is no guarantee. A frequency response graph is very helpful. The middle of the graph should be as flat as possible. One can expect the line to drop off to the left, indicating the limit of the bass response. It may also drop off to the right, though a good “tweeter” will extend it out to nearly 20khz. The scale at the left of these graphs will be in decibels (“db”). If there is a peak (upward projection) of more than 10db in a speaker response graph, you will probably have feedback difficulties with that speaker.

While microphones tend to have more trouble producing good crisp high notes, speakers have trouble producing very low bass notes. Even speakers that do not come with graphs will almost always have a rating that gives their lowest (and highest) frequency response. Speakers that can produce very low notes (low frequencies) must be physically large and strong—which makes them heavy. For spoken voice applications, speakers rated down to 100hz or lower will work fine—most PA and medium-size stereo speakers fall into this category. For male singer's voices, speakers should go down to about 75hz. If large musical instruments (piano, organ, bass, drums) are to be amplified or if taped music is to be played as an accompaniment to singing, the lower the speaker's frequency response, the better. This is another area where you can start simple, with smaller inexpensive speakers, and then as your group grows and money is available, replace them with better speakers that will be appreciated by all.

One of the most common ways of evaluating speakers is to compare them with other good ones. Electronics store clerks should be able to connect a good microphone to an amplifier and the best PA speaker they have. Stand a few feet away from the speaker with a microphone facing away from the speaker and have someone turn the volume up as high as it will go without “feedback” or ringing. Talk or sing. Then, connect the amplifier to the speakers that you intend to buy and stand the same distance away from them that you did from the other speakers. If you have to turn the volume down right away to prevent feedback, the



speakers probably have a “peaky” response. If you have to turn the amplifier volume up slightly to achieve the same sound level, that means that the cheaper speakers are not as efficient as the best ones (they do not produce as much sound for the same amount of power). Unless you meet in a large hall where high power is needed, this is not a major concern. If the cheaper speakers sound acceptable to you and sound nearly as loud as the store’s best speakers without feedback, they will probably work for you.

Most PA systems use two speakers. This gives better sound coverage, and will still provide at least minimal function if one speaker completely fails. If your group is small and budget is a problem, you can buy one speaker now, and add a second later. For low-ceiling, long, narrow rooms, you are probably better off with one speaker and one speaker-stand, than two speakers and no stands.

Column speakers can be particularly useful in some PA applications. Column speakers use 2 to 6 identical drivers mounted in a single speaker box. These usually handle the bass and mid-range, and one tweeter is included in the box for the high frequencies. These speakers tend to disburse sound over a wide area, whereas traditional single-driver speakers tend to send sound waves deep. Nevertheless, column speakers have more difficulties with frequency distribution, and may sound worse and produce more feedback in long room situations. Therefore, column speakers should not be purchased unless one clearly has a “wide room” situation.

A few other speaker pointers: **1)** It is very nice to have two jacks on the back of a speaker; one can connect the first speaker to an amplifier, and then connect two speakers together. This is sometimes easier than running two separate cords or installing a “y” connector on the floor where it might get kicked apart. **2)** Speakers with adjustable tweeter or midrange controls on the back are not that helpful. Some such controls make an almost imperceptible difference in

the sound, and those that are significant run the risk of being accidentally altered when the speaker is being moved or set up. It is better for all such corrections to be made at one equalizer, rather than some on the equalizer and some on the speaker controls. **3)** Buy speakers that have either side or bottom mounting holes so that they can be put on a stand. If you do not need stands now, you might later. For about \$10 each, you can buy stand mounts that can be placed in a hole drilled in the bottom of a speaker, but it is easier to have mounts there from the beginning.

Speaker Stands

These components are often ignored, but they can make the difference between a pleasant sound system and one that people complain about. People need to hear the high frequencies in the 1000hz to 3000hz range to understand what a person is saying. However, these frequencies are easily absorbed by people and their clothing. Also, they tend to travel in straight lines, whereas low frequencies will tend to fill a room no matter where the speakers are located. **If you set speaker units on the floor, it will be difficult to achieve an even distribution of the higher frequency sound from the front to the back.** Speakers on the floor produce sound that is “too loud” in the front of a room and “too quiet” or “hard to understand” in back. Placing speakers on the floor may be effective in small rooms or less-crowded rooms (lots of space between chairs). **If everyone can see the speakers, they can probably hear them effectively.**



When placing speakers on the floor becomes ineffective, setting them on chairs or tables is a great improvement—especially if they are on a level with the heads of the seated audience. However, the best method is to place speakers quite high. Why? Because this guarantees a line-of-site to everyone in the audience; they will all be able to hear the high frequencies. Also, it creates a greater distance between the speakers and the people sitting in front to the audience—this helps prevent the sound from being too loud for those in front, but still allows it to be loud enough to reach those in the back.

Speaker stands will not be needed when a

hall is small, or if tables, a stage or some other building feature makes it easy to elevate speakers. But if a hall is large, especially if it is long and narrow, and has no place to elevate speakers, **then speaker stands are a must-have item!** See the later section on “Equipment Placement” for important information.

Equalizers



Equalizers or “graphic equalizers” are sometimes built into amplifiers and sometimes separate units. Equalizers have a large number of sliding controls that all look the same—except for a different number written on each of them. They are nothing more than very specific “tone controls”. Most people are familiar with tone controls on a stereo—the *treble* control provides a way to independently adjust the volume of high frequencies and the *bass* control provides a way to independently adjust the low frequencies. A typical inexpensive equalizer may have ten controls, each labeled with a different “center frequency”. The labels might read: 30, 60, 120, 240, 480, 1K (1000), 2K, 4K, 8K and 16K. Each control adjusts a range of frequencies centered around the number specified. For example, the “60” control might adjust from 45 to 90, then the “120” control would adjust 90 to 180, etc. This is like having a separate volume control for each octave (each group of 12 notes, including black keys) of a piano keyboard. Why would anybody want such a thing? There are at least six reasons for using an equalizer:

1. To reduce microphone feedback. This usually is related to the next three points, but even with “perfect” equipment, an equalizer would still have some use in reducing feedback.

2. To adjust for deficiencies in the rest of the sound system. This is one of the most essential uses. Frequencies where microphones or speakers are “peaky” (have a greater than average response) can be compensated by adjusting the corresponding equalizer controls down. Similarly, when microphones and speakers are weak at certain frequencies, they can be in-

creased on the equalizer. There are limits to what can be increased—especially for speakers. If a speaker’s usable low frequency response is rated at 70hz, a modestly improved bass response probably can be achieved by turning the 60hz equalizer control about half way up. But boosting the 30hz control all the way up is much more likely to cause distortion or destroy the speakers than it is to produce thunderous bass.

3. To adjust for deficiencies in a room. Each room is acoustically different. It may be “live” at certain frequencies and/or dead at others. The technical reasons for this are rather complex, so they will not be covered here. One important thing to realize is that a room full of people often has different characteristics than an empty room—so equalizers sometimes need to be readjusted after the people arrive.

4. To enhance the sound of a speaker or singer. Decreasing bass frequencies will sometimes make it easier to understand a man with a “boomy” voice. Decreasing treble frequencies will make a shrill woman’s voice easier to listen to for a long time.

5. For special effects. Boosting the frequencies around 100hz can be used to make a voice sound “powerful”. Turning down the frequencies below 300hz and above 2500hz can produce an “old time radio effect”. There are others effects one might want to achieve.

6. To protect speakers from damage. In some situations, one might willingly give up good bass response in order to save equipment. Nearly all speaker cone destruction occurs due to too much power at low frequencies. Turning the lowest few equalizer frequency controls all the way down greatly reduces the chances that speakers will be damaged. (For example, with a 500 watt amplifier and two speakers designed for only 100 watts and a lowest frequency of 50hz, turning the 30hz control all the way down will protect the speakers.) This can be a particularly helpful when a singer’s microphone is in front of drums or a bass guitar speaker and inadvertently picks up those instruments. Some equalizers or amplifiers have a “low cut” switch for this very purpose. (If you set equalizer controls all the way down for this reason, leave them down, even when future instructions in this section say, “set all controls to flat”.)

The more controls an equalizer has, the more effective it can be at solving feedback or other

audio problems. However, the more controls, the more complicated it is to use. An amplifier that has a bass, midrange and treble control can be treated like a “3-band equalizer” and some of the following techniques can still be used.

How to use an equalizer: Always begin using an equalizer by setting all of the controls in the middle or “flat” position (volume at all frequencies is equal). Never arbitrarily set controls. This writer has seen people say, “I like bass”, and then watched them turn all the low frequency controls all the way up and the high frequency controls all the way down—and then watched them play a tape and wonder why their system sounded awful. **If you do not use an equalizer correctly, you can end up making your sound worse rather than better.**

One of the best ways you can become skilled with an equalizer is to practice with it. This would be best done in the room where you use it, but anywhere will help give you experience. Turn off the microphones from your sound system and play a good quality music CD or tape—one with both low bass and high cymbals. Start with the equalizer set “flat”—everything in the middle. Then, one at a time, slowly move each control as far up as it will go, then as far down as it will go. Listen to the change in the music. This will give you an idea of the specific effect of each control. Make sure you move them back to the middle. After you have finished, you may conclude that your system sounds better if you make certain adjustments to the controls. Write these down (e.g. “60hz, 2 notches up, 240hz, 1 notch down, 4K, 2 notches down”). If you have time, try this again with several different CDs or tapes and compare your results. This will help you distinguish between settings you made to improve a particular song and settings that improve your system.

For the next exercise, return all the controls to flat, and attach all of the microphones as you would normally use them. Ask a long-winded person to talk while you again try moving each control all the way up and then all the way down. **Be careful—your system may feedback when you begin turning some of the controls up.** If that happens, quickly turn the control back down. You may also note that the person speaking becomes either easier or more difficult to understand with the controls in certain positions. All of these are good things to write

down. When you are done, compare these results to the ones from playing music. If there are certain settings that you made to your equalizer in nearly all of the previous situations, they are probably settings that you will want to keep permanently—you may wish to write them on your equalizer in some way.

If you do not have problems with “feedback”, you may keep these equalizer settings for a long time. Most equalizers provide an “in/out” switch to temporarily bypass their effects. You can always try this to be sure that your settings are actually an improvement to the music or voice to which you are listening.

Using an equalizer to reduce feedback is sometimes a complex task, but doing so can make a huge difference in the effectiveness of any service or program. To review, feedback occurs when sound from the speakers gets back to the microphone and is amplified and sent out the speakers again even louder—the volume increasing until the maximum amplifier power is reached. Feedback can always be eliminated by turning down the volume, but then one may not be able to hear the teachers or singers at the desired level. Using an equalizer makes it possible to eliminate feedback by turning down just the offending frequency, rather than the entire audio spectrum. Feedback is more of a problem when someone is very soft-spoken, when it is desirable to achieve high volume levels, or when microphones and speakers must be placed very close together. Feedback can be reduced by changing positions of microphones and speakers, but that is not always possible—sometimes the job is left to the equalizer.

“Tuning” an equalizer to reduce feedback should be done in an environment as close as possible to the actual conditions in which a sound system is used. The speakers and microphones should be in the same positions and set at the volume levels where they are typically used. When professionals do this job, they sometimes go as far as placing stuffed sacks on every chair to simulate the acoustical properties of people. Set all your equalizer controls to flat—or to the settings that you have carefully determined make your system sound better. Then turn up the amplifier’s master volume control slowly until your system just begins to feedback—if it starts to squeal loud, turn it all the way down, then turn it up again to a slightly

lower point where the feedback does not occur. If you had to turn it up quite a bit from your normal setting to make it feedback, you should not need much equalization. If your system begins to feedback with any increase of the master volume control from normal, then you probably need to alter the equalizer settings a lot.

With the master volume at “near-feedback”, one at a time, try slowly turning up the first equalizer control (if you have some deliberately set all the way down because your speakers do not handle that frequency, skip them—leave them down). If feedback begins right away, then this is a problem area, and it should be set down a couple notches from where it was. If feedback does not occur until the control is moved half way up, then the control should be returned to where it previously was. If turning the control all the way up does not produce any feedback, it probably represents a “dead spot” in the sound system and you might turn it up a notch or two.

Repeat this process for each of the equalizer controls. After you have finished it, you should be able to turn the master volume up even further without feedback. If feedback begins to occur right away, use your past experience to guess at which frequency control needs to be moved down to eliminate the feedback. If you guess wrong, move the one you changed back to where it was and try another one. Sometimes, when the feedback point is right between two controls (for example, the feedback is at 3K and you have only 2K and 4K controls), you may need to move both of them together to eliminate the 3K feedback. This is a difficult situation. If too many other high frequencies (around 2K and 4K) are lost, it may be necessary to obtain an equalizer with more bands, or new speakers or microphones.

After the feedback has been eliminated, turn the master volume control back to its normal level. Write down your current equalizer settings. If there are other significantly different microphone configurations that you use, you might try redoing the equalization in some of those setups. Compare your results to see if anything is significantly different. You may find that one or more frequency controls may need to be increased with one microphone position and decreased with another. You can either make careful notes and change the settings each time the microphone configuration changes, or possibly

just set the equalizer control in the middle of the optimal settings. Finally, it would be good to use the newly established setting to listen to the sound of speech through the microphones and to play recorded music.

You may have to make some last minute adjustments when somebody unexpectedly moves a microphone or speaker to a new place. That is one good reason for writing down settings. As a room fills up with people, its acoustic properties change. If feedback problems suddenly occur, experience will often provide a good idea of which frequency needs to be decreased. But once the problem situation goes away, it is good to be able to return to the original settings.

Another type of equalizer, sometimes called a “parametric equalizer” or “adjustable center equalizer” provides a way to adjust the center frequency of one or more controls. This type of equalizer is more difficult to understand and use, but it can be far more effective in removing feedback or compensating for other specific weaknesses. It will have **two** knobs for each band of frequencies. One knob, like the other equalizers, determines how much that frequency band will be boosted or cut; the other determines the center frequency of the band (which is not adjustable on the other equalizers). There is usually some limit as to how much the center frequency can be adjusted. Some units have several overlapping bands, one from 20 to 200, another from 100 to 1000, another from 500 to 5000 and another from 2000 to 20,000. One can search for potential feedback problems by turning an amplifier up almost to the point of feedback, then turning the first frequency control about half way up and adjusting its center frequency throughout its range. If there is one frequency—or a small range of frequencies—where feedback occurs, then it can be eliminated by leaving the “center frequency” control at the problem point and turning down the level control for that band. The procedure can be repeated for each control. This system does not have the problem of a feedback frequency being “between two controls”.

There are fancy equalizers (sometimes called anti-feedback systems that have lights showing how much audio signal is in each frequency range—some will even automatically detect feedback and adjust for it. They are expensive and probably needed only in extreme situations.

Remote Speakers and Monitor Speakers

Although the purpose of these two items are quite different, their technical solution is similar, so we discuss them together. **Remote speakers** are speakers located outside the main hall. The most common use for these in relation to a worship service is various kinds of nursery rooms where parents can go to feed, change, or calm down young children. **Monitor speakers** are speakers placed near musicians and singers at the front of the auditorium so they can more easily hear other musicians and singers. Since the main speakers of a PA system are usually pointed at the audience and away from the front of a hall, it can be very difficult for a piano player located on one side of the lectern to hear singers on the other side of the lectern, and vice versa. Monitor speakers are aimed at the **back** of directional microphones where they do not pick up sound well at all.

The essential requirement of both of these speakers is that their volume can be independently controlled from the main speakers. A nursery may need the volume adjusted loud so people can hear over crying babies; it may need to be adjusted softly so as not to wake up sleeping babies. A monitor speaker must be adjusted so it is just loud enough to be effectively heard by the musicians—if it is up too loud, it can cause “monitor feedback”. (This is feedback from the monitor speakers to the microphones, which will also be heard through the main speakers—turning the main volume down will quiet this feedback, but not stop it. You must turn down the microphone or the monitor speaker volume to stop monitor feedback.)

The simplest way to connect a remote or monitor speaker is to find a speaker with a volume control on it. They are not common, but they are available. Your amplifier must have sufficient power to run all of the speakers. If the amplifier you are using now occasionally distorts when someone is speaking or singing loud, adding another speaker will only make the problem worse. Also, the net number of “ohms” (impedance) presented to your amplifier must be consistent with its rating. Most speakers are 8-ohm. When you plug more than one into an amplifier, the net number of ohms presented to the amplifier can be calculated by dividing the ohms of one speaker by the number of speakers. If you add a third 8-ohm speaker, then 8 divided

by 3 is 2.67. If you add a fourth 8-ohm speaker, then 8 divided by 4 is 2. Many amplifiers set 2-ohms as their minimum—they will over-heat if you connect speakers with a combined impedance of less than 2 ohms.

Two things to be careful of: some amplifiers support a minimum load of 4 ohms—two 8-ohm speakers is all you can hook up to them. A few speakers are 4-ohm—you can hook up only one to a “4-ohm minimum” amplifier or two to a “2-ohm minimum” amplifier. If you put too many speakers or the wrong “ohms” on an amplifier, **it will probably appear to work**. But it may distort or overheat. Overheating will either cause it to temporarily shut down if it has “thermal protection” or else cause permanent damage. You do not want any of the above to happen during your service.

The other approach is to use a separate amplifier for the main speakers and the remote or monitor speakers. Some PA amplifiers actually have separate output sections that can either be used together or separately. By flicking a switch or using a patch cord, you can have separate volume control of two different speaker outputs. A stereo PA amplifier can also sometimes be configured to work in this manner (the “left” channel can be used for both of the main speakers and the “right” channel for the monitor speakers.) Most amplifier manuals will have instructions and diagrams on how to hook up monitor speakers.

If your amplifier does not have a separate power output for monitor speakers, you can usually run an audio cord from your main amplifier to another amplifier for your remote or monitor speakers. This will provide separate volume and tone controls for these speakers. An inexpensive public address amplifier or even a home-stereo amplifier usually works well for auxiliary speakers. Most main amplifiers have several jacks to which you can connect an auxiliary amplifier. For **monitor speakers**, you will want to use your main amplifier’s “monitor output”—this will give independent control of the volume of each microphone into the monitors.

Remote speakers should contain the same sound heard in the main room, so you want to use the “main output” rather than the “monitor output”. If your amplifier contains an equalizer, you may have a choice of obtaining an output that is “pre-EQ” (before the equalizer) or “post-

EQ" (after the equalizer). If your equalizer is set largely to adjust for the room or deficiencies in the main speakers, then you probably want to use the "pre-EQ" jack for your remote speakers—the equalizers corrections are not needed for those speakers. However, if your equalizer is primarily compensating for your microphones, then equalized sound from the "post-EQ" jack may be the best. If you are not sure, you can simply try all ("main out," "aux out," "pre-eq out," "post eq out," etc.) to see which produces the best sound for your remote speakers.

If your remote speakers are hundreds of feet from your main amplifier, you may have some difficulty running a cord that far. If you are using a non-powered remote speaker with a volume control, then all you need to do is run a speaker cord. Do not use a musical instrument, microphone or other audio cord. Use a speaker cord with at least #16 wire (#14 or #12 is better). If you can solder or screw on the necessary connectors on the ends, you can buy power cord by the foot from a hardware store. (It is often even cheaper to buy an extension cord and cut the ends off).

If your remote speakers have their own amplifiers, then you will need to run audio cord of some kind. In order to avoid the loss of high-frequencies on such a long run, you should use balanced-line microphone cord. You will probably need to obtain transformers to connect the microphone cord up to the ¼" or RCA output jacks on your amplifier. (You can obtain these from Radio Shack, or the mail-order vendors mentioned earlier). If physical obstructions make it too difficult to run a cord between rooms, consider using an FM transmitter and receiver. These are available from audio stores, Radio Shack, etc.

Complete Packages

Today, many audio stores (both local and mail order) will sell complete packages that include microphones, amplifiers, speakers, stands and all the connecting cords. Buying such a package can often save time and money. You can be fairly sure that they equipment will work well together. However, you must be sure that it contains what you want and does not duplicate items that you already have. For example, if you need a sound system with one microphone and a tape recorder, and you already have a good pair

of speakers, it may not be the best to buy a discounted sound system that comes with three microphones and speakers but no tape recorder—you will be buying stuff you do not need, and still missing the tape recorder that you do need. If you find a complete package that has basically what you need, it is still a good idea to review each of the components separately, using the principles in this article, to see if they are what you want. Also, realize that not every "complete package" is a good deal—some stores are used to charging church-groups a lot because they figure that they do not know much about sound systems and might not comparison shop.

Groups with a small budget might consider using a "Karaoke" machine. They contain microphones, an amplifier, tape and/or CD players and speakers. Some of the better units will come with microphone and speaker stands. Please realize that the quality of Karaoke machines varies drastically, and so does the price. Cheap machines under \$50 are not worth trying. Machines in the \$100 to \$200 range may work well, but some of those will still be poor quality. Try any machine out first—in your actual location if possible. Many karaoke machines are not expandable. Be sure that yours will handle the greatest number of microphones that you would need, provide a way to spread its speakers far apart, allow speakers to be mounted on stands, etc. Some Karaoke machines do not use standard microphone and speaker plugs and are therefore hard to upgrade. Also realize that the more expensive Karaoke machines do video projection of song words—a feature which you will pay for, but may not need for your service.

PA Equipment Placement

This is an extremely vital part of providing an effective sound system. Sometimes, a person who knows little about sound systems will be in charge of setting up a hall. He may want the microphone to be placed far away from the lectern so it does not get in the teacher's way. He may want the speakers to be placed against the wall behind the lectern so they do not spoil the decor and do not obstruct people walking to and from the lectern. These are all noble goals and can sometimes be achieved, but other times they must be sacrificed if people are to hear what is being said. Where you are able to place compo-

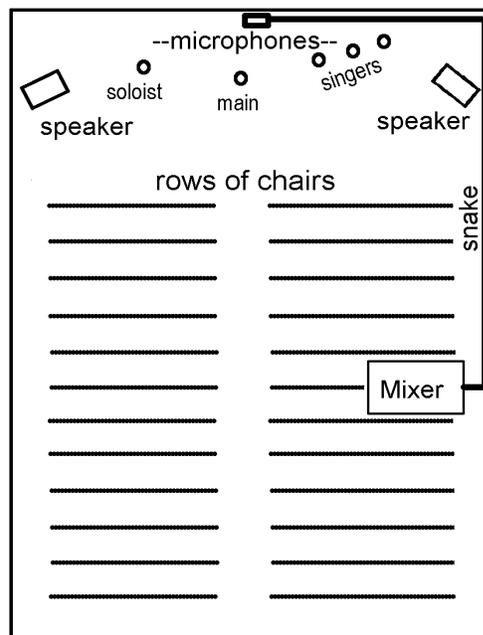
nents of the sound system depends on many factors. If you have very good equipment, a well-designed room, and a teacher that talks loud, you will have a lot of choice in equipment placement. If, on the other hand, you have poor equipment, a difficult room and a soft-spoken teacher, than you may have to place your equipment strictly to optimize your sound, and let everyone else work around it. We will describe the best possible placement for acoustic purposes, then you make adjustments as your system permits.

Microphones should be placed far from noise sources. You do not want them to be near a noisy open window, a noisy air duct, or some other noise source. You may have to change how you arrange a room to prevent this. Also, parents with children that may cry loudly should not sit in front rows very close to a microphone—their loud little voices will be picked up and sent through the PA system and recorded on tape as well. The closer a microphone is to the teacher or singer, the less the likelihood that it will pick up noise or “feedback.” This is the technical reason why professional performers keep their mouth only an inch or two away from their microphone. However, this close distance is not practical for most lecture-type sessions. People that are speaking have a tendency to turn and face different parts of the audience—they may move around a lot behind the lectern. If the distance between the speakers mouth and the microphone varies greatly (from less than one foot some times to several feet at other times), the volume level will fluctuate greatly. For speakers that do not move too much, a microphone mounted on the lectern or on a “boom stand” will work quite well. However, if a teacher moves around a lot, you will need a lapel microphone. If there is so much movement that tripping over the cord becomes a problem, then you may wish to consider a wireless microphone. If you are having much difficulty with feedback, the best thing is a mounted microphone and a person that speaks loudly and stays close to it. The next best method is a lapel

microphone, and the worst is a mounted microphone with a speaker that is far away. One other thing to watch: a microphone pointed at a hard, flat surface (like a wall behind the speaker) can sometimes be a source of feedback since the surface reflects the sound from the room back into the microphone. Pointing the microphones at an angle toward the place where the wall and ceiling join helps to eliminate this problem. Also, hanging a thick curtain or decorative tapestry on the wall will reduce feedback.

Speakers should be placed so that the sound from them does not get back into the microphones. Placing them above the heads of the audience (on stands or tables) allows the sound to “get out” to the audience rather than overpower those on the front row and be too quiet for those in back. High placement also helps to reduce feedback. Placing speakers far to the side of the room also helps to eliminate feedback problems. So far we have two recommendations: high in the air, and wide away from the lectern. But how far forward or backward should they be? (For purposes of this discussion, the microphone and lectern are located at the front of the room.) The further toward the back of the room that speakers are placed, the more feedback will be reduced. However, far-back placement also means that the musicians will not be able to hear the singers at all through these speakers, creating the need for additional monitor speakers (see section on them).

Also, placing speakers too far back can be distracting to the audience. To understand why, we must know something about how the human ear works. If you are in a room with your eyes closed and someone calls to you, you can probably figure out what direction they called from—even though their sound is reaching you from many different directions. The sound comes directly from their mouth, but it also reaches you after bouncing off of walls,



ceilings, etc. All of the “bounced” sound takes a slightly longer path through the air than the sound that reaches your ears directly. Since sound travels at 1100 feet per second through air, the “bounced” sound arrives in your ears a few thousandths of a second later than the direct sound. Your ears marvelously sort out all of this and tell you that the sound came from the place where it arrived first—not necessarily from the place where it arrived the loudest. Therefore, when a PA system is used, a listener will attribute the direction of the sound to the person talking as long as they are closer to that person than they are to the speaker units.

If you would like the audience to feel like they are listening to a person and not a speaker box, then place the speakers at the edges of the room, but a little more toward the audience than the microphone. If you do not have any problems with feedback, great. However, this is a trade-off. If you have feedback problems, then you may need to move them toward the audience. What usually happens is that moving the speakers changes the number of seats where people will attribute the sound to the speaker system and not to the person talking. People sitting in the middle of the room will almost always be closer to the person talking than to the speaker units. People at the left and right front corners of the room will often be closer to a speaker unit than to the person talking. The higher, wider, and further forward you can place your speakers, the fewer the number of seats that will be consciously aware that they are listening to sound coming from a speaker system.

Once speakers have been positioned, it is also worth considering the direction in which they should be “pointed”. High frequencies tend to be directional, so they will be heard best in a line directly in front of the speaker. They fade gradually as one moves around toward the side of the speaker. Very little high frequency sound is heard from the side or back of a speaker—reflections off of the walls will still be heard, but they are much more difficult to understand. The general rule for aiming speakers used for speech is: **Aim speakers right into the middle of where the people are.** Sound aimed at walls or ceilings does not help anyone hear, but simply adds to the total sound that might cause feedback or produce unwanted echoes. This means that in a square room, speakers in the

front should not be aimed straight back, but each should be pointed at the opposite corner of the room. If speakers are mounted high, it is best if they are angled down at the people, but many speaker stands do not allow this. Speakers used for musical instruments do not need to follow this “directly at the people” rule as much because reverberation (additional short echoes) often enhance music. That is why musical instrument amplifiers are designed to rest on the floor and point up into the air.

Amplifiers and Mixers, from an electrical point of view, should be placed in the closest place between the microphone and speakers so the amount of cable needed to connect them will be minimal. For a simple one-microphone PA system, this may work fine. However, if a variety of people talk, or if the system is to be used for multiple musicians, then there will be a need to adjust volume levels frequently during the service. You probably do not want someone doing this right in the middle of the front of the room. Therefore, it is best to place the mixer out in the room when someone can hear the sound in the room and react accordingly. This is especially true for musical selections when a singer using a microphone has to be balanced with a person playing a piano or other instrument that is not using a microphone. The middle of the room would be best, but this is often the most difficult place to run microphone and speaker cords. Cords must be covered by rugs wherever they are exposed to foot traffic—both for safety and to prevent cords being accidentally disconnected while they are in use. Therefore, the side of a room frequently becomes a reasonable compromise, or sometimes the back (but then cords must be quite long). These long cord runs certainly require “balanced lines” with 3-pin XLR plugs. Other types of cords will be too noisy and lose high frequencies over long cable runs.

If you use more than three microphones, you may want to consider buying a “snake”—a long cable that contains many microphone cords inside it. “Snakes” are available in 50’, 100’ or longer lengths with 6, 8, 12 or even more microphone cords inside. The individual connections are numbered at each end so you can keep everything straight. Some snakes also contain speaker wire and jacks. It is much easier to use a snake than it is to run many different cords.

Digital, CD and Tape Players

Most congregations will want the ability to play recorded messages or music to the congregation. If this is a regular occurrence, then you want to strive for good quality sound—otherwise the members of the congregation will become frustrated listening to it. For a once every few months occurrence, putting a microphone in front of a boom box might be good enough. If you do this, turn the boom box up as loud as it will go **without** distortion and point the microphone directly at one of the speakers. If there is significant stereo content—different things in each speaker—then use either a microphone for each speaker or place one microphone a foot or two in front of the boom box, centered between the speakers.

(If the microphone is off to the side of the boom box, or close to it but not pointed at a speaker, you will lose much of the high frequencies, making music unpleasant to listen to and voices hard to understand.)

The best way to connect CD players, digital audio players or older tape recorders is through a cord to your amplifier or mixer. By digital audio players, we mean MP3 players, Ipods, cell phones, computers or anything else that plays digital music and has an output jack. It is best to have these devices located near the mixer or amplifier when using them. (Sound quality may begin to degrade if they are connected with cords longer than 6 feet.) The person who runs the mixer can stop and start the player, and adjust the player volume and/or tone controls.

Even though most players can run on battery power, it is best to use one that works from outlet power as well. A rechargeable device that is regularly charged is also good. Batteries eventually wear out and you do not want to distract the service by replacing them during it. On the other hand, outlet power is sometimes temporarily lost; a player with batteries, too, will not power off if someone blows a breaker or kicks your cord out. If a tape player loses power, you just hit the play button and keeps on going from you where. With CD or digital players, you may have to do a lot of listening and searching to get back to a previous place in a long recording. Keeping a player continuously powered up is a worthwhile goal.

Better quality CD players and tape recorders will have stereo "line out", "output" or "play"

jacks that are designed to connect to an amplifier. Use them if they are available. Virtually all players will have "headphone" outputs—usually a 1/8" stereo mini plug. These outputs will generally work if connected to an amplifier, though you may need to turn the player volume **down** somewhat to avoid distortion in the mixer by sending it too strong of a signal. "Speaker" outputs can be used, but require even more "turning down" and will frequently be noisy.

A local department store or an audio store will have the cords you need to connect your player to your mixer or amplifier. Even if your mixer/amplifier is not stereo, it will usually have a stereo "tape in" jack. If not, buy an adaptor to combine the stereo channels, or use two channels of your mixer and control them together.

Cassette tape players and recorders are rapidly disappearing from the market, as they have been replaced by CD and digital players/recorders. Nevertheless, some people will probably want them to play older sermon tapes. If you use a tape player/recorder, remember to periodically clean and demagnetize its heads.

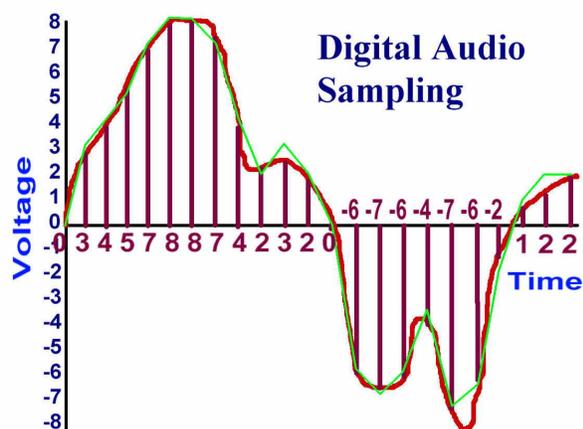
Understanding Digital Audio

Since more and more audio devices today are digital, rather than analog, it is important to know enough about how digital audio works you can understand device ratings and properly select and use digital equipment.

Old fashioned audio equipment is "analog" equipment. A varying sound wave in the air is converted by a microphone to a varying voltage that is, ideally, the exact image of the sound wave. This varying voltage is then amplified and applied to speakers, which convert it back to nearly identical, but louder sound. The image of that varying voltage could also be impressed into vinyl (old-fashioned records) or changed into a varying magnetic field on cassette tape. The difficulty with these analog methods is that the vinyl and the magnetic tape do not do a perfect job of recording that electrical image (distortion), and they introduce unwanted noise of their own. When an audio message is recorded and played back several times in one of these processes, a significant loss in quality is noticed.

The approach used by digital audio is to take the analog electrical signal that is the image of a sound wave and, as soon as possible, convert it to a series of numbers that mathematically de-

scribe that sound wave. The process of converting to numbers is called digitizing or DA conversion (Digital/Analog). This process does alter the sound somewhat, but **once the sound wave has been reduced to digits, these digits can be transmitted and copied any number of times while maintaining all of the quality of the original signal.** This ability to copy without sacrificing quality is very useful for making copies on demand, editing for professional recordings, and transmitting over the Internet. It is wonderful for spreading the Gospel free of charge. Secondly, the ability to store digital recordings on memory chips, with no moving parts, **greatly increases reliability and virtually eliminates maintenance.**



The thick red line, above, represents an analog audio voltage waveform (which is the image of a sound waveform). The digitization process involves sampling the voltage of the wave at regular intervals. The quality of this process depends upon three things:

1. **The number of samples taken per second.** The samples are represented by the brown lines and numbers, above. There are only 23 samples, not near enough to accurately describe this waveform. Standard CD audio takes 44,000 samples every second, a sample rate of 44Khz. Better quality digital audio will use 96Khz, 192Khz or even more. It is possible to hear the difference!
2. **The precision with which each sample can be mathematically represented.** If you are measuring with a yardstick, you could give a measurement to the nearest foot, to the nearest inch, or with a very good yardstick, to the nearest $1/64^{\text{th}}$ inch. All audio digitization processes have a stated maximum precision with

which they measure, usually expressed in the number of "bits" (mathematical base 2) that are used to describe one sample. CD audio uses 16 bits, which gives 2^{16} or about 64,000 possible different values for each sample. 24 bit audio has about 16 million possible values and clearly sounds much better than 16 bit audio. 32 bit audio is also available. In the above graph, there are only 17 possible values (-8 to +8, including 0). This is like 4-bit audio, which would sound bad. Some older telephone answering machines used 8 bit audio—one can understand the voice, but it is distorted and scratchy. **The thin green line** in the graph indicates what would happen if this 4-bit, low sample rate audio were converted back to an analog wave form. The playback equipment only knows about the numbers that were recorded, represented by the 23 single-digit numbers. The playback will produce the stored whole-number voltage of each sample, rather than the more accurate in-between voltage (for example 3 volts instead of 2.6) that was in the original waveform. Your ears could easily hear the difference between the original thick red waveform and the reconstructed green one. The latter would sound bad.

3. **The accuracy of the digitizing equipment.** It cheap digital recorder may produce a file that is 24bit with a 96Khz sample rate, it may not sound anywhere near as good as a recording studio instrument with the same ratings. The cheap equipment may use a real sample rate of 48Khz and then just mathematically create a set of samples in between the other samples—which have little correspondence to the actual sound wave. Expensive equipment provides samples that much more accurately represent the sound being digitized. The best equipment uses a technique called "oversampling", where the digital-to-analog converter may actually take eight times as many samples as it needs, and then use a sophisticated computer program to figure out the best values for the one out of eight samples it will save.

Practical things to know for digital audio:

The initial conversion from analog to digital audio is the most important. If poorly done, there are no good ways to make the bad audio better. Simply converting it to a higher sample rate or more bits will do nothing.

1. Conversions among digital formats always reduce the quality—sometimes significantly. For example, if you transfer a CD to a digital recorder via a computer, and use a different digital sampling and bit rate on the digital recorder, the quality will get worse. Also, if you simply connect the CD to a digital recorder with an audio cable, the CD will be converted back to analog and then re-digitalized by the recorder. The quality will suffer—sometimes considerably.
2. There are now several different standards for connecting digital audio devices together. These are found on more expensive equipment. For example, you can connect a digital mixer, a digital effects processor, a digital recorder or a computer. This avoids multiple digital/analog conversions. Unfortunately, there are a variety of jacks and digital protocols. Most of them are not interchangeable. Even though some use the standard RCA audio plug, you most certainly cannot connect any digital input/output to an analog audio input/output. However, if you intend to record multi-track music in the future, you should look for equipment with digital inputs and outputs.
3. Many digital devices have options for different recording sample and bit rates. Selecting a higher quality always reduces the total amount of recording time that any given storage device can handle. Also, stereo usually takes twice as much storage as mono recordings. Do not be fooled by a digital recorder advertised to record "up to 100 hours"—that amount of time may be available only with mono 12-bit recording and a 16Khz sample rate. Recording time at the highest quality may be only a few hours.
4. Most digital recording devices offer some forms of "compression". Traditional audio CD's do not use any compression, they have 44,000 16-bit numbers for every second of sound recorded on them. Since the CD format was established, many complex means have been developed to store the same information in less space. By using various mathematical techniques, it is possible to compress an audio file to less half of its original size, without losing anything. This is called "lossless compression"—the original file can be created just as its was. There are also "lossy compression" methods, such as the popular MP3, that can compress audio data down to 1/10th of its original size, losing only a minor amount of quality. Other compression methods go even further, but the audio quality begins to noticeably degrade. Again, be careful of devices that advertise large amounts of recording time at high sample and bit rates—they may be using compression that reduces audio quality.
5. The way to achieve the best audio quality for a given storage size is generally to reach toward the upper (but not highest) levels of sample and bit rates, while using only slightly lossy compression. You will have a hard time finding fault with the quality of 24-bit music sampled at 96Khz with MP3 compression, but it will take up only 1/3 the space of standard CD audio which is 16 bit at 44Khz and no compression. For voice recording, usable quality is possible at lower sample and bit numbers with more compression. If you make files for the Internet, you will want to use a lot of compression and smaller sample and bit rates so that files are smaller and easy to download or "stream" to others.
6. It is difficult to choose the equipment you need from ratings alone. Have several people listen to any digital audio equipment that you intend to use and be convinced that it will produce the quality that you need.

Recording Some or All of a Service

If the only purpose for recording a service is to keep a record of what was said, and to have tapes available for a few people who missed the service, then almost any means of recording will work. But if your goal is to encourage people to regularly listen to some or all of your service on tape, CD, or the Internet, then producing better quality will increase the chance that people will listen to it. (Hopefully the main reason people listen is for good content. If the content is bad, it is a mistake to try to dress it up with first-rate audio. But if the content is good, quality audio will help people hear and understand it.)

Before one decides what kind of recorder to buy, there are two important questions to ask:

What will you record?

1. Only a teacher
2. The teacher and members who interact with him

3. Music

What will you do with your recordings?

1. Distribute cassette tapes
2. Distribute CDs
3. Put recordings on the Internet

If you have an older congregation that still wants cassette tapes, and you plan no other use for your recordings, then a tape recorder will probably be best. One with "Dolby noise reduction" will have a cleaner sound. If your recordings are longer than one side of a tape (45 to 60 minutes), an auto-reverse recorder will automatically keep recording when it reaches the end of the first side of the tape.

If you want anything more than cassette tapes, you will be recording digitally.

If your goals are to provide un-edited CD's of your service, then obtain a stand-alone CD recorder—a device that records directly to CD. From there, you can use a CD duplicator. For services more than 80 minutes, a dual-deck CD recorder is good, it will automatically switch from one CD to another. This author's experience has shown that rewritable CDs fail more often than read-only CDs, and recommends that the cheaper read-only CDs be used exclusively. This also prevents someone from accidentally recording over the only copy of a master recording.

If a CD recorder is used, the CD audio can be uploaded to a computer and converted to formats for the Internet. It can also be recorded to cassette tape, if some of those are still desired. However, CD recorders do have more failures than most of us would like—due to bad blank CDs, occasional dust, and the recorder itself. Also, most CD recorders lose the entire recording if there is a power failure while it is being recorded. This happens more often than we would like—especially if power cords are in a position where they can be accidentally kicked out. (Some people simply plug all of their recording equipment into a computer UPS—an Uninterruptible Power Supply—a \$100 to \$300 device that will continue to supply AC power for 5 to 20 minutes when outlet power quits.)

The most reliable method of making good recordings is using one of these small digital memory recorders that operates both from batteries and outlet power. They are available in a great variety of price ranges, recording qualities, and recording time durations. Most of them are easy to upload to a computer, which can then be

used to edit the recording, burn it to a standard CD, burn it to a high-capacity MP3 CD, or upload it to the Internet. Many of these devices have interchangeable memory cards or memory sticks which can extend their recording times and transfer recordings to computers. They also come with built-in microphones, which are usually sufficient if the recorder can be placed on the speaker's lectern (provide he doesn't hit it with his books or notes).

Use the guidelines in the last section to help understand what kinds of sample rate, bit rate, and compression are acceptable in recording devices. If saving all of your recordings is important to you, make a plan to always save two copies of each recording: upload it to the hard-drive of a computer that has a regular back-up procedure, upload it to two computers; save each recorder memory chip as well as upload it to a computer, or upload it to a computer and burn it to a CD right away.

You can also record your service directly to a computer with the appropriate software. Such software can be free to very expensive (try download.com). See the section below on Computers and Duplicating Equipment.

Whatever your means of digital recording, consider your final product when selecting your recording sample rate. If your main use is making CDs, then recording at 44Khz is a good thing—no conversion will be required. If you want to record at higher quality, it is better to use an exact multiple (88Khz) or something that is considerably better (96Khz or 192Khz). Digital conversions of nearby sample rates (like 48Khz to 44Khz) produce a marked decrease in quality.

Finally, realize that the method of controlling the volume level on any recording device is very important. If you send too powerful of a signal to the recorder, the recording will become distorted. If you send too weak of a signal, there will be too much background noise.

This is still a factor with digital recording media. Digital recorders have an absolute highest number that then can record (both positive and negative) and if a voltage larger than that largest number comes to them, they just record the highest number available until the voltage drops down to a usable range. This causes a flat spot on the top of the waveform and sounds much like an over-driven amplifier.

The opposite extreme is too weak of a signal.

With digital recording, this means that there a large amount of the possible values digital values are never used because the voltage never gets high enough. Yes, you can turn the volume up when you play it back, but a 16-bit recorder with a "way too low" level will sound like an 8-bit recorder because only 8 bits are being used.

Most recorders have a series of lights to indicate the volume level—the louder the sound, the more lights go on. (An old tape recorder will have a "level meter"—another moving part to fail.) Typically, green lights mean "O.K.", yellow or orange lights are a warning that you are nearing the maximum volume and red lights mean you have exceeded the maximum level and distortion has occurred. Cheap systems may have just one red light meaning distortion. The ideal volume adjustment is one that is as loud as possible, but that virtually never turns on "distortion lights". If someone accidentally hits a microphone, coughs loudly or drops something heavy on the lectern you might expect to see the red lights come on, but nobody cares if these sounds are slightly distorted. Also, some systems will let their red warning lights come on for very minor distortions that one can barely hear—that's O.K. If the system has yellow or orange warning lights, they **should be** going on during the loudest parts of your recording.

The question always is, how do you know how the loudest sound a speaker or musician will make so you can know where to set the level? The answer is, you don't know, exactly. You can test it ahead of time, but people almost always get louder in front of a live audience than they do during a test. Some recording engineers perform their tests, then back the volume level down a notch before the real thing.

Most recorders now have some kind of electronics that attempts to make these level adjustments for you. The feature may be called "automatic level control" "audio limiting", "audio compressing" (not to be confused with digital compression), etc. These features are excellent for speakers or musicians that have an average normal volume, but occasionally much louder surges. Without electronic level control, the volume must be manually adjusted downward during the loud parts, then back up afterward.

On the other hand, some automatic level controls can be very irritating. Some will try to adjust the volume to the optimal recording level

no matter how quiet the sound is. If a teacher pauses for a few seconds, the electronics may turn up the volume until one hears room fans or air-conditioning system. It may sound like the surf is coming in. Page rustling and crowd noise may also be heard during quiet parts.

The best automatic level controls adjust only a limited amount. That means it is a great help for the sound system operator to get the level control close to right to begin with. Anytime a recording is to be made, a few minutes should be spent with someone speaking into each microphone at about the anticipated volume level so that a basic level can be set. From there, levels can be tweaked to optimum values and automatic level controls can smooth it out.

Level controls are utterly vital if you want to record questions and comments of everyone in the congregation. For best sound quality, those who have questions and comments should step up to a microphone to make them. Alternatively, someone can take a wireless microphone to them. Even with these arrangements, people will still talk out of turn, not hold the microphone close enough, and speak at greatly varying volume levels. In these situations, the recording engineer either needs a very good automatic level control, needs to continuously adjust the recording levels, and/or needs to edit the final recording to adjust the levels after the fact.

Connecting Your Digital or CD Recorder

The cords that you need to connect your mixer or amplifier to your recorder will be similar to those required to connect a player to your system. If you are plugging into the "microphone input" of a recorder, realize that it is designed for very low-level inputs and you may need to turn down your PA system output to prevent distortion. If your recorder has a "line" input, use it.

Nearly all PA systems have a "tape output", "main output", "auxiliary output", "effects output", "monitor output" or some similar jack. You may even have a choice of more than one—a "pre-EQ out" output will not be affected by the equalizer while a "post-EQ out" will be. If your equalizer is set primarily to correct for your speakers and room acoustics, then you will probably want to use the "pre-EQ out". This is the most common method. If your equalizer is set primarily to correct for the microphone or the person using it, then you may want to record from the "post-EQ out".

If you connect your recorder to a "monitor output" or "effects output" of your mixer, you may have completely separate volume controls for that output for each mixer channel. These will allow you to use the monitor volume controls to separately adjust the levels of each microphone for the recording. In some mixers these outputs are "post-fader" (the main fader for each channel controls the monitor level in addition to the "monitor" control). In other mixers they are "pre-fader" (the main fader for each channel has no control, it is all done by the monitor volume controls). Some mixers can be switched to either method. If a microphone is used for the members' comments, it will be possible to turn it up louder on the tape. Or, if a teacher is talking while he walks out into the audience with a microphone, right in front of a speaker, the main volume will need to be turned down in order to prevent feedback, but the "monitor volume" can stay the same to keep the recording level the same. On the other hand, using separate controls for live sound and recording makes a lot of mistakes possible: microphones with nothing important can be accidentally left turned up adding noise to the recording, and important microphones can be accidentally left turned down, leaving wanted material out of the recording. If separate volume controls are used, it is best if the recording engineer continually listens to the recorded sound with headphones to make sure it is all there.

Sometimes, an amplifier may not have enough outputs for your remote speakers, monitor speakers and/or recorders. You can buy a splitter adaptor from a local audio store to send one output to multiple devices. Be careful to check the volume levels every time a device is connected or removed: it may substantially change the volume level to the other devices.

If you need to both play CDs tapes or digital recordings at your service, and also record the entire service to the same kind of device, it is highly recommended you use two separate devices. Why? 1) If you want to record the entire service, you will not be able to record while you use the same device to play back. 2) You can easily create an electronic feedback loop which can damage both your speakers and amplifier (not to mention everyone's ears). Most recorders send the audio signal that they are recording back out of their playback jacks. If you have the

playback jacks going into your mixer/amplifier and that volume control up past "unity" gain, it will continually try to produce a louder version of itself, resulting in a loud, disgusting sound. This can be prevented by turning the playback volume down or by disconnecting the playback cable, but these things are very easy to forget about during a live production.

If you only have one channel to record, you can save memory space on digital recorders by recording in mono mode. However, if you are recording directly to CDs or tape, you should use the necessary splitter adaptors to record your one channel on both left and right channels. Whoever plays your CDs or tapes will expect to hear it from both speakers.

For situations where a stereo mixer is used with mono recording, make sure that both channels are combined when sent to the recorder.

Computers and Duplicating Equipment

Computers now play a significant role in sound systems. Sophisticated programs for mixing and recording audio are available free or to many hundreds of dollars (try download.com). Indeed, if you would like to do professional sophisticated multi-track recording with lots of effects, a computer-based system is probably the least expensive way to do it. This paper will not attempt to cover these things, nor the many ways of placing audio files on the Internet.

But before anyone says, "We can save the cost of buying a mixer and a recorder by using someone's computer with free software, please realize that is not that simple. **You will need a specific computer that will have the necessary hardware and software installed for this task.** Furthermore, you will have to control or keep software off the computer that might interfere with this task. It will probably take quite a few hours to get this computer ready.

Most computers have only one sound card with a stereo microphone input, a "line level" input and a speaker output. A few may have a "line level" output as well. This is generally not enough inputs and outputs for a computer to work as the "mixer", unless you are recording just one person. You can gain more input and output channels with more sound cards or USB adaptors for a few dollars each. Nevertheless, all these connections are 1/8" "mini" jacks, which will need adapters to fit the rest of your com-

mercial audio equipment. Indeed, a metal balanced line microphone connector will cost more than the USB audio hardware. The small jacks do not stand up to much pulling and tugging, which commonly occurs in live sound situations. They will easily break, become noisy or stop working all together.

To turn a computer into an effective live sound mixer, you will need additional USB/Firewire hardware designed to interface computers to professional audio equipment. That will cost \$100 to \$400 depending on how extensive you want it—not all that different than a good mixer. However, when such interface equipment is combined with a computer and the right software, it provides a very good multi-track recording studio—something you cannot do with just a mixer.

You should not use a computer as a mixer/recorder and for something else at the same time. Specifically, do not plan to use it to copy CD's, surf the Internet or look up Bible scriptures while it is mixing or recording. These tasks take a lot of computer processing power, which may rob the audio program of the resources it needs to complete its job in real time. The sound program must process the sound as it comes in; it cannot wait for even half a second while some other program monopolized the CPU, memory or disk resources. Lack of resources usually produces an error message and loses some or all of the recording.

Furthermore, it is often necessary to remove or limit the resources used by "background programs": anti-virus software, automatic disk defragmenters, screensavers, animated assistants, and others that run all the time in the computer and occasionally decide on their own to begin some complicated processing. Even if the audio program is running at the highest priority, these other programs can still keep the disk busy or deplete the system of memory, causing it to miss some of the live audio which is being mixed or recorded.

This is why many people choose regular mixers and dedicated digital recorders to capture their live sound. These devices do not get "busy doing other things" and faithfully record until they are full. The computer with the professional audio hardware can still be used as a

multi-track recording studio to edit the captured audio. If some kind of program interruption occurs during editing, one can find out which program did it, fix or remove the program, and redo the last audio editing operation.

A computer can also be used to make copies of CDs. The recording studio computer can do this also, just not at the same time that a recording operation is in progress. If you intend to make more than a few CD's at a time, it is much better to buy a stand-alone CD duplicating machine. These cost between hundreds and thousands of dollars depending on how many CDs they make and how fast. For high volume operations, you can buy machines that will print the artwork on each CD, and some that will automatically load themselves: just place a stack of empty CDs on one spindle and come back in a few hours to find a stack of completed CDs on another spindle.

The ultimate way to share Biblical messages and music is to put it on the Internet and allow other people to download it, email it and burn it to CD if they so desire. That is a bigger topic than this writer intends to take on at this time, but he encourages you to think about it and do it if appropriate. If you cannot host the website yourself, there are other sites that may find your free materials helpful and post them for you (try www.biblestudy.org).

Summary of Sound Systems

A lot of information has been covered in these pages—only a part of which may be relevant to your needs. If you are planning to purchase equipment, or are having trouble with an existing system, we recommend that you make a copy of this document and mark the points relevant to what you are doing—or that you compile your own checklist. Most public libraries will also have books on this subject. It may help for someone to keep notes on the answers to the problems that you solve—that way you can avoid having to solve them again.

Always remember that the purpose of a sound system is to help accomplish the work of a congregation. It is not a thing to gain power, a status symbol or a toy to amuse the technician. May the Eternal guide and bless your ministry.

Ask for *Starting a Local Congregation* or our *Literature List*, they are both free.

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