An Extended Article on Sound

of Special Interest to those Planning New Buildings

from

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Sound System Concepts and Applications

for Churches & Schools

by Larry Lee Bell

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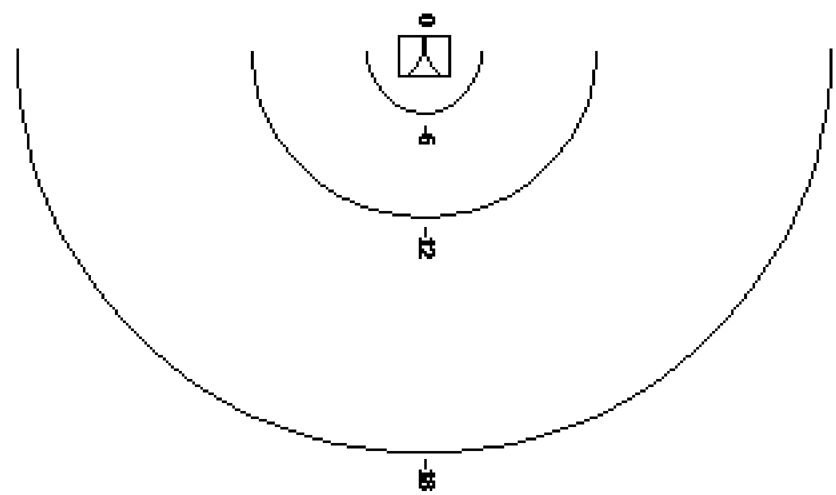
Introduction

What is a "sound system"? In the most basic way the sound system is an extension of the "talker" using the system. As a result, the system should be the most transparent conveyor of information possible. Transparency must, however, take a back seat to the conveying of information, if compromises must be made. Some sound systems have communications only as prime consideration, for example, a paging system in a warehouse has a quite different set of basic criteria from a church sanctuary sound system. A church sound system, obviously, has a greater need of transparency because of the multi-faceted nature of a worship service. Natural sounding reinforcement of the worship experience becomes the major goal. Natural sounding reinforcement is a function of a ensemble of factors which will be dealt with at length in their individual aspects at greater length throughout these discussions.

Relationship of Distance and Sound Volume

The relationship of distance and volume or sound pressure is one of the basics to understanding audio in general. Sound pressure varies in a logarithmic or square law rate (1, 1/4, 1/16, 1/64 Etc.) expressed in dB's (decibels)."What, you may ask, is a dB?" In terms of sound pressure an average person can hear one dB difference or change while listening to a test tone changed in a A to B comparison of tones with a 1 dB difference in volume.Every time you double the distance between the hearer and the sound source there is a 6 dB drop in volume. Simply put as the sound "scatters" from its source it becomes weaker as distance grows but at a rate of decrease that becomes less apparent as distance increases.



Sound Volume and Distance

As a result of this, it is less of a task to have even sound when the sound source is not too close to the listener. An average person can hear a 3 DB change in volume while listening to music or speech, while a 1 DB change can be detected with a pure tone. This principal can be used to provide regulation of coverage by speaker placement that provides a ratio of 1 to 2 in terms of difference in distance between the first and last hearer. Volume before feedback ("howl") cannot be ignored in speaker placement because a speaker that covers evenly, but is not loud enough because of feedback, is a bad compromise.This principal also applies to microphones as well. As a result the distance from the microphone to the talker is important for gain or volume before feedback, as is case with the speaker placement. The closer the microphone is to the talker, the less likely that feedback problems will occur for a given acoustic condition and desired operating volume.

Another factor in gain before feedback is the number of microphones turned on. Every time you double the number microphones "turned on" there is a 3 dB drop in the margin between the usable volume and the beginning of feedback. Making a usable sound system is a bit more complicated than just picking some speakers that sound good and getting them in the ceiling or on the walls on the listener's side of the mics. Putting sound where you need it without generating echoes and lack of definition that renders the system loud but not understandable is often an art. Speaker components come in a limited number of projection patterns and arrangements so speakers don't always fit the room. The room is often the limiting factor to how good or usable a sound system can be. Just throwing money into a sound system without room treatment is often a waste of money and time. Sound in the room that is muddled when no sound system is used is a clear sign that room treatment and a well designed system are a must for just adequate sound. A word about room treatment: carpet on the walls or common acoustic tiles often are useless to help the most common problems, and are often a waste of money and time. Generally speaking, the lower the pitch of the echoes and muddle, the more difficult they are to control with inexpensive means. Another major factor in echoes and muddles is relationship of long parallel surfaces, which are best corrected as part of the original design. Odd shaped rooms present their own set of problem, sometimes doubling the cost of a sound system per seat. Often auditoriums are built for visual beauty or cost per square foot, but are terrible for their purpose of aural communication, often the last consideration in terms of budget or accommodation. How many times have we heard the line "This building won't need a sound system," and seen it designed to eliminate the possibility of a practical system installation. Be sure your building get acoustic help early, not last.

Power and Apparent Volume

Another aspect of sound regulation that is not obvious is the relationship of change in power to sound pressure level. Doubling the amplifier power will result in only a 3 dB increase in potential volume. This factor is probably the hardest to relate because of the relatively small increase in acoustic power for a electrical power doubling. This makes the choice of the minimum amplifier power required much larger than "common sense" might indicate. The difficulty is convincing a person that the 20-Watt PA amplifier that was almost powerful enough requires a 100-Watt replacement is that it just does not seem to make sense at first glance.

Add to that the complication that differences in efficiency of speakers can also create difficulty in determining amplifier power requirement. For a technically extensive discussion of these factors see Sound System Engineering, 2nd Edition, by Don and Carolyn Davis a book published by Howard W Sams available through our association with amazon.com.

The Most Important Element of the Sound / Acoustic System is the Room

One of the most ignored parts of the worship environment, especially in the initial planning and in the design phase is the acoustic environment. Exceptions may randomly occur with certain shapes and sizes of rooms, and sometimes 'dumb luck' results in acceptable performance. What does this really mean in practical terms ? Let me speak of the more typical in anecdotal type examples:

"We have a beautiful building with visual closeness but I can't understand."

"The sound system sounds fine, but looks like an awful wart on our beautiful building."

"We hear and understand clearly but its too loud in the front and not quiet loud enough in the back."

"Our choir loft is spacious but our singers can't hear each other to blend."

"Its loud but all I can hear when I speak is garble."

All the Examples above represent problems with the building and its furnishings. In most of the the cases above the solutions (if any) to these problems could be made best before the building was designed (shape, materials, and configurations chosen). The 'if any' in the statement above is very important because many problems have no solution that is acceptable.As buildings get larger in size and seating capacity the acoustic problems that are built-in can grow exponentially. Larger buildings often have lower pitch acoustic problems that require much more expensive corrective measures, since less expensive materials simply do not work at lower pitches. For example, carpet material has very little effect at low pitches, as many have found in trying to make a multi-purpose building acceptable. Higher-pitched reverberations may respond to less expensive solutions; but, obviously, labor for those solutions is less if the material is installed as part of a renovation or the initial construction. It sometimes seems amazing that so few make priority in planning for the central purpose of communications in there auditorium or worship center.

The sound system, at its most basic, is a communications system for audio information. In the audio world, communication has both objective and subjective components. A telephone or warehouse paging system can provide adequate communication, but few would want to sit in a lecture or sermon with that 'quality' of sound. Transparency, or naturalness, is a major factor in acceptability, especially with the growth in quality of the typical home stereo system. Communication and transparency are normally difficult to achieve simultaneously.

The acoustic environment is like an unmolded sculptures clay. The unfortunate fact of the matter is there are very few great artist in designing great acoustic spaces. Therefore, often the best way to find a good design is to emulate a good design. This process is flawed, however, since seemingly minor changes in dimensions, materials, or configuration that may result in major changes in the acoustic environment. As a result, every decision of a architect, building committee, or internal furnishing committee must be seen in the light of its affect on the acoustic environment. Unfortunately, common sense about acoustics does not exist with most members of the building trades and architect community.

Wall materials like sheet rock, plaster lath, brick (glazed or unglazed), concrete blocks with thick or thin paint (filling or conforming to the surface), and prefab concrete wall each have very different acoustic qualities and have major affect on the acoustic environment. Large flat walls in the rear of buildings are a major source of echoes, even in smaller buildings. Low ceilings often prevent the possibility of good sound in a small or large buildings. Wall-to-wall carpet, as compared to strips and tile, make a major difference in the acoustic environment, while padded pew backs and seats may make the room, the same, whether empty or full, a good quality. Building shapes that tend to focus sounds make a building's acoustic qualities erratic depending on your position, with echoes sometimes being much louder than the desired sound, creating a nightmare unless extensive (read "expensive") treatment is done.

A word about absorption, reflection, the physical property of the conservation of energy, and the frequency-dependance of those qualities: the energy of the sound has three qualities when it encounters a room factor (usually a little of all three). Sound energy can be reflected (move away in a different direction), absorbed (turned into a small amount of heat by an absorber), or refracted (bent into several different directions by a surface or air temperature boundary). Unfortunately, nothing is quiet as simple as it seems because all three of these factors are pitch or frequency-dependant. Therefore, a wall that absorbs almost all of one pitch may almost completely reflect another pitch.

A common mistake based on a false assumption is adding absorptive material without considering the frequency or pitch. It is common to use of carpet on the walls to solve a reverberation problem in a gymnasium. In most gymnasiums, the pitch of the unpleasant reverberations is well below the frequency for which carpet has any useful absorptive qualities. For those responsible, the end result of the expense of installing the carpet is a dull-sounding room with the echoes almost as bad as before. An now the hard facts: very few inexpensive or easy techniques work in this case. Hanging professional acoustic baffles may be the only easy way to solve these problems after a gymnasium has been constructed.

In another example, problems occur because of a seemingly desirable design quality of having a large amount of seats close to the platform or stage. A circular or round design provides the greatest number of seats from the focus of the building. The word "focus" is the key to the terrible acoustic qualities of the round or focused design. Correcting the focal echo effect in the acoustic realm requires that the outer walls be absorptive, or at least diffusive at all audible frequencies. Any lack of care in the choice of materials or shape of the outer walls can result in an acoustic disaster. A room where the message being heard and understood is paramount becomes, in itself, the primary hindrance of communication. There is an interesting aside to designing usable buildings based on a circular model. The location and design criteria for a sound system must be critically determined to blend ascetically and acoustically with the designed space. A 'minor' change in speaker location (including height of the array) can make a major difference. As an example seen in an existing building: the architect changed the height of the sound chamber without determining its affect on the sound. The result was the inability of amplified sound to reach the front of a 1,500 seat building without ugly and expensive 'add on' speakers hung below the designed speaker enclosure. A postulate to the previous example is reaching an over or under balcony seating area without the use of special and expensive speakers. In some building designs a single central speaker array can 'see' every seat in the house while others require the expense and sonic problems of many speaker locations to cover the same number of seats. Another more common problem is locating speakers behind the plane of the primary microphone creating a feedback ("howling") loop.

Lest you begin to still think that simple solutions exist, let us talk of some additional complications in our example. If sufficient diffusion or absorption have not been included in the outer walls, a single source will provide a 'wonderful' excitation to focus echoes. If multiple remote speaker are to be used, provision must be made for their installation, including conduit and location allowance, in the appropriate locations. These speakers must be electronically aligned to avoid echoes and dead spots that result from multiple source locations. It is worth stating that only speakers located and oriented correctly can be accurately aligned in most rooms. Attention to every detail allows a blend of acoustic, aesthetic, and financial considerations. Buildings with similar shapes and configuration may be acoustically quite different after only slight changes.

A very general word about ceiling height in relationship to depth and width ratios: with a speaker array to seating area ratio of 1 unit height to 2 unit of depth, you have a better chance of good back to front volume consistency because of the way sound propagates. As the sound spreads it becomes less dense. Coupled with how our ear and brain perceive sound, there is very little difference between sound volume at 1 unit of distance and 2 unit of distance. Buildings with ceiling heights high enough to provide good volume regulation are preferable to buildings where the ceiling is too low. Very wide buildings will generally require multiple speakers in their speaker array. More than one sound source location always creates dead spots due to interaction of the different sound arrival times at the listener's location.

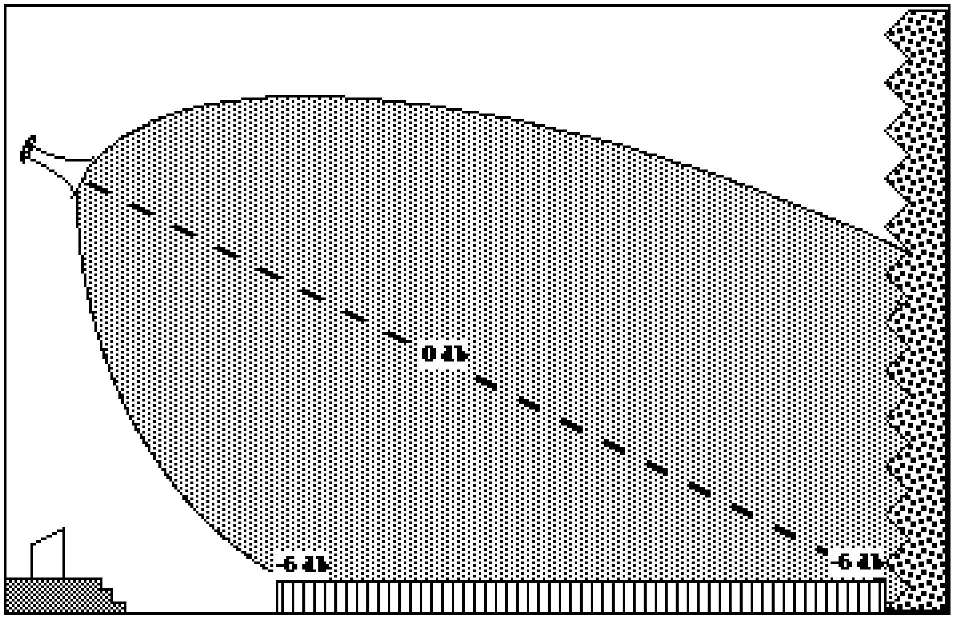
In another instance, a church is using a large open office space to provide a low-cost first sanctuary . Its low ceiling violates the even propagation principals because listeners of a front-located system, sitting in the rear, are a many multiples of the ideal distant when using a single or frontal split speaker location. The result: people in the front of the room are deluged with sound, while others in the rear do not hear well. The usual solution is many inexpensive ceiling speakers with their usual tin-can sound. Ceiling speakers can provide even, understandable speech if properly delayed and aligned, but they can sound terrible for music; while with some engineering and money, a more acceptable solution is possible. For example, special purpose, low- ceiling speakers provide some tools to achieve a better music quality system. A better choice may have been look for a more acceptable room instead. One Church I know took out the existing lay in tile grid and painted flat black the higher upper ceiling and added light fixtures that de-emphasized the less pleasing ceiling giving enough ceiling height for better speaker placement and dispersion.

Why a Central Cluster

Why do central clusters work better than a split system for good sanctuary sound? In the simplest of terms, multiple sound sources create dead spots that are location and pitch sensitive. Sound waves of the same sound but from several sources arrive at various places in the room out of synchronization with each other thus creating echoes and dead spots.

Another reason for a central cluster is the propagation of sound in relationship to distance. As the distance from the sound source to the hearer doubles the volume drops by 6 dB's. If the first hearer is 40 feet from the cluster and the most distant hearer is 80 feet from the cluster then sound is + or - 3 dB's the sought after benchmark by the propagation alone, not taking into account the directionality of the speaker.

Sound Regulation Due to Horn Pattern and Ceiling Height



Even given that speakers do not disperse as evenly as theory would indicate, a properly placed speaker gives your system much less adversity to overcome. However, an improperly placed speaker often gives you problems insoluble, regardless of the amount of money spent or electronic wizardry tried. Some problems, it should be noted, are only cured with acoustic modifications.

The room designer may give you a nightmare that has no good solution; therefore, don't let the building's shape, size, and material be decided before sound or acoustics is considered. It should be noted that visual beauty or economy are often the only considerations, giving the result a house of worship is built where people do not receive the Word with clarity and beauty. It is troubling to note that I have not meet many room designers who had any concept of how to design a room acoustically.

Unfortunately independent consultants are often so politically tied to building design firms that they find it difficult to say to them, "This is completely wrong; start over." For instance, the visually most efficient space is a circle, with everyone being close to the center. Therefore modified circles have become popular auditorium shapes. Circles, however, are one of the most acoustically disastrous rooms conceivable. Even with lots of treatment (read expensive) they are only marginally usable.

Our discussion here, obviously, is not exhaustive, but gives the reader in a compact form a feel for the complications and care that must be taken in choosing shapes, finishes, and accommodations necessary for aural communication and transparency.

Electrical Interference

Light Dimmers and Wire Routing

Some kinds of electrical interference are generated by commercial light dimming equipment. Most such interference requires replacement of the dimmers since filtering doesn't work in most cases. Light dimmers designed for theatrical or television use are generally designed to minimize interference, while most sold for home or commercial use are not. Locating dimmers in areas away from sound and other technical equipment, utilizing a remote control console, is a good idea. Route wiring carrying sound and dimmable circuits well away from each other. Sound cables and conduits should be separated by at least six inches and should cross at right angles. Different types of cables should be in separate conduits. Microphone, line level, speaker, video, intercom, telephone, and other signal carrying cables should each have there own conduit or routing and be separated. Bundling different types of cables together can cause noise and other problems generated by signal interaction.

Other Building Related Problems

Ground Loops and Gremlins that Drive You Crazy

Ground loops are one of the most pervasive problems in church sound systems. It seems through the years if someone is having a strange problem and equipment is located in separate locations the problem is a ground loop. Ground loops, result, in most cases, from not doing it right the first time. It is important to note that many people think that an "under grounded" system is the source of the problem: but, as you will see, that is not necessarily the case.

One of the reasons ground problems are so pervasive is 'doing it right' requires so much coordination between almost all of the contractors during construction. First, is the conventional way that the electrical contractor installs the ac wiring creates ground loops in the sound system. Oversimplified, a ground loop is too many paths to ground. When audio equipment has a path to ground thru the ac plug and again through audio connections, and the locations are separated, you now have a ground loop path. The best cure for this is to use isolated ground ac outlets (special orange outlets fixtures) with a separate isolated insulated ground wire from the outlets to a common ground for all sound, video, and computer equipment . The insulated aspect is paramount because any contact the ground makes with steel building members (for example conduit) creates another path to ground.

Ground loop prevention also requires design features as part of the sound equipment. Balanced inputs and outputs (that is, 3 separate wire connections) are required for truly good interfacing. Equipment that does not provide a separated ground (allowing shield connection on only one end) from the low and high signal connections are a source of ground loops. If your ac was designed without isolated ground outlets audio transformers can be added; this allows you to adopt the standard procedure of disconnected shield wires on one end, thus isolating the grounds. It should be noted that the shields are connected to ground but not each other.

Here are some quick checks that can determine if you may have a ground loop problem. If it is possible to float the ac ground by using a 2 to 3 wire adapter, leaving the ground wire or tab disconnected, listen to the noise level or other strangeness in the system and see if it goes away or down significantly. Also try disconnecting the signal feed on one end and seeing if noise goes away or reduces significantly. These checks cannot be used as a fix, since floating the ac safety ground, as in the test, is a safety hazard; and, obviously the signal wire is required for operations and the noise source may be only be with what you disconnected. These test are not fool proof but can be useful in isolating your trouble. Ground loops, however, defy the conventional wisdom in their strangeness; and, frankly, "Common sense doesn't work with this stuff". The best cure for a ground loop is do it right the first time.

Construction Related Building Factors

Accommodating Speakers Locations and Cable Raceways

Providing accommodation for the appliances of the building's technical systems require effort at the planning stage. Perhaps the hardest part for the professionals involved in the planning is finding out everything in detail at the beginning. Leadership probably should 'over plan' in terms of accommodation and economize on implementation. Conduit, speaker locations, floor pockets for mics, movable monitors, video feeds, video projector locations, projection screens, stage lighting locations, communication outlets, and extra conduit to areas of control center or platform areas etc. Accommodating many or most of these is relatively inexpensive as compared to trying to install them after construction. Unfortunately since these future allowances do not have the same checks and balances of other trades demanding them you must 'bird dog' them yourself.

Providing an adequate number of electrical circuits in the worship area is also much less expensive during construction. Remember that one spotlight may require its own power circuit. Many are surprised when they attempt to set up portable lighting, only to find that every outlet in the building is on one circuit. This is a very common practice, the theory being that you will only use one at a time, so the extra circuit capacity is unnecessary. In most auditoriums, during special programs, you may have special lights, slide projectors, musical instruments, and other electrical needs all at the same time. Again, providing enough outlets on separate circuits is inexpensive during construction, but very expensive after the walls are up and the electric circuit boxes installed. If it is possible in your design to provide an open access way between building parts, it will provide additional flexibility in future. Spare conduits and larger conduit both can provide future flexibility.

Now a word about speaker locations these should be located and accommodated based on a professional design before the plans are approved. So often the sound system looks like an add-on because this coordination has been put off. Sound components' size are not arbitrary but determined by the laws of physics. Their size, shape, and location cannot be changed without affecting performance beyond the acceptable limit. Another factor sometimes ignored is that the aiming required for installation requires space sometimes up to several times the size of the speaker components. Locations of lighting fixtures, ceiling fans, and other possible obstructions to sound must also considered. Sound gradually becomes more directional as it goes up in pitch. Generally speaking, "if you can't see it you can't hear it." Generally lower pitches require larger size speakers. Unfortunately, the window of acceptable performance is very sensitive to the speaker components chosen. Therefore, what fits is often unusable, so here comes the ugly add-ons. Structural consideration also require that the structural engineer provide adequate support structure in his design for speakers, theatrical or television lighting, video projectors, projection screens, and other equipment that must be suspended from the ceiling or located in the attic.

Taking Quality to the Next Level

Attention to detail in making your decision about your building and sound system is the key. There are certain products and designs that are a cut above in quality. Obviously, there are elegant products that are very expensive; but, not as obvious, are some very elegant products at an especially good price. It seems that special values that provide elegant sound are out there that can find application in your system. These products come and go; thus, continuing to be 'on the hunt' for these is an ongoing quest. Unfortunately, consistency among brands is more in the negative than in the positive. Colleagues and specialist are a good source for your search. At this point a word about quality is in order. Sonic quality without lasting qualities is no value. Certain brands have a reputation for continual attention to long term serviceability. Ask the question, "In five years, will it be serviceable?"

When looking for a mixing console for your needs, ask others who have owned that brand for five or tens years how the slides and controls have lasted. Noises and reliability problems are the bane of a good worship environment. Often controls, when they go, are so labor-intensive to replace that the mixing console must itself be replaced.

Sonic quality and noise figures never get better with age. Since most popular music have performers closely miced and relatively constant dynamic volume range, impressions you get in a music store or listening to a popular band may not tell the story of noise when the system is used in the wider dynamic range (loudest to softest volume) of a worship or school environment. When the channel volume is raised enough to hear a podium or choir mic, the mixer may contribute a significant amount of 'hiss' and noise which will never be heard when using a solo mic or recorded music and adjusting the mixer controls down. Unfortunately, many popular brands of the mixers, when used in worship or school, may be distractingly noisy. Again checking out mixers in a installation similar to your own is the best .

Now a word about power amplifiers: two major characteristics of amplifiers in general are sonic quality and reliability. Sonic quality is sometimes achieved at the sacrifice of reliability, and sometimes reliable amplification is at the sacrifice of the sonic quality. Many of the major brands have found designs that balance the need of both reliability with sonic quality. Unfortunately, many amplifiers fall short of the high standards required for reliability and sonic quality. A word should be said here about specifications, because they can be written in a deliberately misleading manner. This is especially true in the auto stereo and music store business. The "standard" for power specification in the auto stereo is a whopping 10% distortion. Spec tricks used in sound system amps are sometimes just as deceptive. One example of this deceit is the power bandwidth spec of 20-20,000 Hz + or - 3dBs or typical. Three dBs is half the power of the amplifier, thus hiding a poor power supply or other amplifier flaws. The power and load specification is another easily deceptive spec. The standard for amplifier power should be into 8 Ω (ohms) with both channels of a stereo amp driven to rated output. Any power rating into a lower load will be larger, so some brands began to show powers into 4 Ω loads with only one channel driven instead, which gave a higher number. The next stage, of course, was to show power into 2 Ω with an even higher powers. Perhaps an apt analogy to deceptive specs would be to say a lawn mower engine could pull a car (which it could with a gearing ratio for a very slow speed but hardly usable on the freeway). Another common practice is to use a number in the amp model number like "400," which may correspond to nothing, or an exaggerated power with both channels added together at a 4 or 2 Ω rating. Another specification that is easily misleading is the damping factor and other specifications at 1 Khz, which generally shows the amp in a better light than full bandwidth specifications conforming to best measurement specifications. Again experience and reading specs with a fine-toothed comb are the best ways to choose an amp. Amp specs that are deceptively written can indicate a product with something to hide.

Equalizers and Equalization

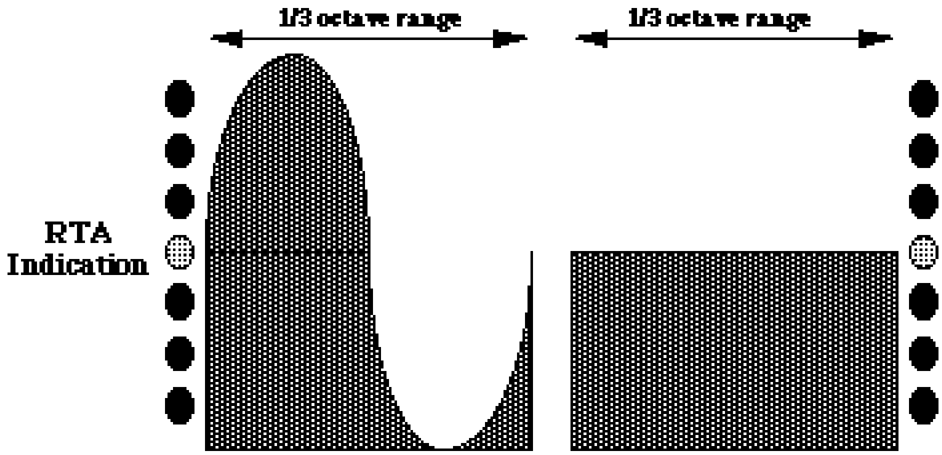
When we talk of equalizers and equalization, the boundaries of definition sometimes become muddled. Equalization may refer to the complex process of adjusting a sound system to a room. This process includes more than just setting the equalizer. Equalization of the room system involves setting crossover frequencies and parameters, adjusting the volumes and alignment of various speaker elements, setting any notch or parametric equalizer, and adjusting the normal 1/3 octave equalizer. Equalization is also sometimes used to refer to the adjusting of the 'tone controls' to obtain some subjective sonic goal.

"A person with a meter and a tin ear is dangerous." Instrumentation is useful, but unless you really understand what it really is indicating, and the nature of the room, you can have a terrible sound by an improper use of a meter. Let's begin with the tool most often used in setting an 1/3 octave equalizer: the real-time analyzer. First, the instrumentation used can vary in quality and price by a factor of ten in price and quality. However, even the most expensive instrument can be badly misused.

The indicators are divided by pitch or frequency ranges. The key word is "range," to begin to understand what really is measured. An ideal analyzer would measure all the energy in a range as a single indication. However sound usually does not have a flat volume across the range but energy above and below within that range. So an equal rise (and width) within the range with an equal fall (and width) within the same range will measure the same as a flat volume across the range. As you can see, the "meter's" measurement is useful, but you must know what it really indicates.

The biggest 'monkey wrench' comes when an extreme rise occurs, usually due to a room's "resonance" mode, or when an extreme low occurs, usually due to an absorber within the room. A notch or parametric equalizer can be used to 'tune out' a room resonance (which has caused a rise). Actually, what occurs is that you 'turn down' the volume in the sound system for the offending pitch range so the sound system doesn't make it worse.

Absorbers (which cause a fall), however, are a different matter because turning up the offending pitch range usually simple results in the absorber absorbing more energy. Turn up the volume on an 1/3 octave range can cause the sound to be "all wrong" because you are turning up the volume on either side of the absorbed pitch only. Remember - the analyzer might indicate flat when you turn the volume up but actually it will sound bad. As you can see, simply lining up the dots on the analyzer with the equalizer can result in very unnatural and unpleasant sound.



Both of the Sound Energy Diagrams Measure the Same

A word about equalizers is now in order. The controls in the ideal should be a volume control for a range of pitches. The cut and boost, unfortunately, does not occur evenly across the range of pitches, but generally favor the center over the edges. The linearity (or 'evenness') of the adjustment varies with the price and quality of the equalizer. Phase response, or the affect the equalizer has over the complex relationships within the sound waves, is the great unspoken gremlin within the world of equalizers, since it can greatly affects how it sounds to us.

With the coming of the TEF machine and the development of digital signal processors, improvements are coming in equalizers and associated instrumentations ability to tell us how it sounds. I, however, still stick to my statement "A person with a meter and a tin ear is dangerous."

Perhaps you can now begin to see how a little knowledge can be dangerous in arriving at a good design and execution of the design. While specifications sometimes seem similar, actual performance may vary considerably. With professional help and experience, you can make more educated decisions about your sound needs and solutions. Excellence is a artistic blend of experience, skill, and careful execution.

Speaker Directivity and Matching the Room

Putting sound where you want it, and away from where you don't, is the function of speaker directivity. Directivity is much easier to achieve at higher pitches. This fact is one of the reasons so much time is spent in these discussions on the room. Generally speaking, directivity requires devices whose size is determined by the wavelength of the pitch (frequency) of the sound you wish to control. Lower pitches have much longer wavelengths. Horns are one way to control the directivity of sound. Horns to control low pitches can be enormous, but do find wide use in large stadiums used by Major League Baseball and NFL football. The size of such devices make them undesirable for most auditoriums but some large horns find use for mids and mid- lows in some medium to large auditoriums. Devices where the high frequency horn is coaxially mounted in mid frequency horn are available from several manufacturers, both with and without digital processors. As with microphones, the more directivity or concentration of the sound, the more the 'bending' of the sound can cause distortion, unless carefully engineering utilizing the best the state of the art is utilized. The science of speaker design and manufacturing continues to grow, providing more tools to the sound system designer. Again, experience with the products can tell you more about how the speaker will sound than will specs. Accurate directional and frequency information in detail, however, is very important for the system designer to have in order to map coverage. A combination of accurate directional information and a knowledge of the sonic qualities of the speaker allow engineering artistry in the acoustic design.

When you talk about directional characteristics, two factors are paramount: volume (acoustic level) at a specific pitch or frequency. There can be great variation among speakers from, pitch to pitch and angle to angle, which affects the speakers ability to provide coverage, in the areas desired, with sonic consistency. In the original horn speaker designs, the coverage narrowed as the pitch increased. Due to the work of D. B. Keel and others, new horn designs were developed with more consistent overall coverage, based on complex geometric angles in the horn shapes. Since that time, the technology has continued to progress with the aid of research utilizing the newest digital technology. Sonic qualities, however, still remain a subjective judgment based on objective facts

What is 3 way, 2 way etc.

It turns out that because of sound wavelength, certain sizes of speakers produce sound more efficiently, and subjectively sound better. Dividing sound into several pitch ranges allows you to utilize a speaker that is optimized for efficiency, sonic quality, and directional factors.

The improvement from going from 2 way - that is, one group of speaker elements for low frequencies and one group for high frequencies - to 3 way is remarkable. The sonic improvement by adding a third speaker for the middle frequencies is especially remarkable in reproducing the human voice naturally, while allowing the use of more efficient bass and high frequency speakers. For high-energy popular music, 4 way is sometimes utilized to improve the highest frequencies which is especially apparent in the percussion and instrument overtones.

The sound is divided into the pitch ranges by a crossover network, which can be before or after the amplifier. Crossovers after the amplifier are called high-level crossovers and may be (and often are) installed inside the cabinet with the speaker elements. Crossovers before the amplifier are called low-level and allow the more efficient use of amplifier power. However, more amplifier sides are required, but it is generally agreed that the sound is cleaner.

A combination of high- and low-level crossovers is sometimes utilized: for example, a 2-way cabinet with a built-in high-level crossover could be used with a low level crossover ahead of a single stereo amp, routing the lowest frequencies to a separate sub-bass speaker and the "higher" frequencies, via the other amp side, to the 2-way cabinet, the 2-way cabinet having been selected for efficient mid and high frequencies.

Microphones Where it All Begins

Mike Directivity Types

The most common type of mike pattern is the cardioid (sometime called unidirectional), which I believe is too general a term to be very useful, so I have divided it into three types of cardioid. The cardioid generally gets its name from the resemblance the polar pattern mapping of the mike has to a heart shape. These mappings, when viewed closely, reveal my reasons for separating it into the three types. The cardioid in general picks up sounds well from directly in front, and less well from the sides and back. The directivity is derived by 'loading' the mike element from the side and rear with small openings, or ports. These ports cause sound to arrive at minor differences in time based on there direction, as related to the front of the mike. Small differences in time produce cancellation of sounds from the rear, and some from the sides. Port designs' not being perfect provides some coloration of the frequency response, especially apparent off the front axis. Generally, the more directional the mike, the more coloration of the frequency response of the mike.The pattern of the mike is not constant from frequency to frequency, so directivity is an average of them, or at 1000 Hz when polar patterns are not printed for several different frequencies. The polar patterns are usually measure at one meter in an acoustically quiet environment, shows 'electrical volume' produced at the mike's output based on a constant acoustic volume at the mike's input at the specified distance and direction. A fidelity curve measure at the front will be different at the side and back of the mike. These fidelity curves are also sometimes printed and, like directivity, are usually measure at one meter. Unless you plan on using your mike at one meter, these curves may tell you little about how the mike really sounds at the distance and direction of your use; therefore, experience is the best standard to use in selecting a mike for a given purpose, with specifications being a way to narrow the field. The first type of cardioid I call the "tight cardioid," because it picks up well to the front, but poorly from the sides and rear. This type of cardioid is best used for close micing of voices , instruments , and other sources you wish to mic in isolation from other nearby sounds. Some tight cardioids are called "hyper-cardioid". Coloration and drop in volume occur quickly as you move off center of the mikes pattern. Tight cardioids find wide use as handheld mikes for a single performer, especially when feedback is a problem.

The second type of cardioid is the "medium cardioid" so named because it picks up sounds from the front and approximately 45° off that axis, with good tonality and level. Medium cardioids provide probably the most universal adaptability, allowing micing of more than one sound source, and provides some freedom of movement for the singular performer. I, therefore, recommend the medium cardioid for most general purpose mics you might purchase.The medium cardioid provides a mike that will work for many purposes, from a single performer's hand held use to a duet or trio utilizing a single mike without major compromise of quality or feedback problems.

The third type of cardioid is the "open cardioid," which picks up well from the front, up to 75° - 90° off that axis, with little tonal coloration and a moderate drop in volume smoothly as one approaches 90° off axis. This type of mic makes a ideal choir reinforcement mike, providing open pickup toward a miced group, while providing an even choral sound when properly aimed . The requirement for micing a choir is, generally, one mike for every 1 : 1 ratio of width to depth of choral performance area. Choral mikes should be one foot in front of the first row of standing singers and two feet above their heads, aimed toward the last row. The effect of the smooth attenuation in volume to the sides provides the compensation for the variation in distance of the various choir members, providing the loft is sufficiently reverberant to provide natural blending in the loft area.

These criteria are general rules, and often must be changed due to complicating factors like a pipe organ, or speakers behind the choir. In a loft where people have difficultly hearing each other, choral sound is always a problem, and is not easily overcome by any electronic means (don't ever carpet or deaden a choir area!).

Open cardioids also make good pulpit and podium mikes allowing a public speaker a lot of freedom of movement with little degradation of the sound. Open cardioids however require mike and speaker placement to be at a near optimum location, which should be the case in a well designed system.

The next most commonly found type of microphone is the omnidirectional. The omnidirectional mike has almost no use in reinforcement, in that 'hears' as well from the back as it does the front . The only commonly used omni mikes found are in lavaliere, or clip-on, use . Omni's find wide use as lavalieres because it is very hard to build a cardioid that small with a flat response. Lavalieres, therefore, must be used only by experienced public speakers, and with care in where mike is clipped on the public speaker (as close to the mouth as is practical without shadowing from the chin) Some cardioid lavalieres have appeared but sometimes have a problem in pickup related to head movement of the public speaker. Omni's are never good choir reinforcement mikes because the are doubly prone to feedback; however, for choir recording and broadcast they have the advantage of smoother response than comparably priced cardioids, and pick up some of the pleasant acoustic qualities of the room (and, the unpleasant qualities as well).

Microphone transducer types - Notes and features

The basic types of microphone transducers (pickup element) are dynamic, condenser (or capacitor), ribbon, carbon (the common telephone design), and crystal. The carbon is not used in pro audio. The ribbon, and crystal transducers are not in common use, so we shall restrict our discussion to the dynamic and condenser transducers.

The most commonly used transducer is the dynamic, or moving coil, microphone. The principle of operation utilizes the same principle as your automobile generator; that is, moving a coil of wire within a magnetic field to generate an electric current. The microphone, unlike your car, uses sound pressure as the motor to move the coil of wire in the magnetic field. The microphone produces a small alternating current, which reflects the sound the microphone hears.

The construction of the dynamic transducer usually utilizes a plastic diaphragm, which resembles a small half-bubble, with a small coil of wire attached to its back, suspended by a very light circular suspension. The coil extends into a circular-slot chamber, surrounded by a permanent magnet, which allows its free movement. As sound strikes the diaphragm (with sufficient power to overcome the inertia of the diaphragm), a small current is generated, which reflects the sound wave that strikes it, with an accuracy depending on the quality of the dynamic transducer and its surrounding acoustic chamber.

The dynamic microphone provides the most rugged transducer normally used in quality sound applications. Dynamic microphones find wide usage in all phases of professional audio, and are often the best choice for field sound amplification, and broadcasting. Dynamic microphone elements do, however, have some drawbacks and, as you will find as your experience grows, there are few panaceas. One of the greatest drawbacks of a dynamic is the large mass (compared to sound's energy) of the diaphragm and its associated coil, which makes it less sensitive to far-away sounds and makes accurate reproduction across the entire the audio spectrum difficult because of the relatively low energy of high frequency sounds. Dynamic choir mikes, as a result, often have less than adequate clarity in the critical consonant range, unless a large number of dynamics are used (which can lead to other problems).

The next most commonly used transducer is the condenser (or capacitor). There are two types of condenser transducers: the electrette or pre-polarized, and the externally- polarized. Both use the the principle of unlike charges attracting, which makes the capacitor operate. The capacitor, simply, is a "temporary" storage medium for electrical charges. The capacitor is usually constructed by placing two plates parallel to each other, separated by air or other materials called the dielectric. It turns out that the distance between the plates (or the dielectric thickness) and plate size directly determine the amount of charge the capacitor can hold.

The externally-polarized condenser transducer uses two plates: one fixed and rigid, the other of a very thin and pliable conductive material. To these plates, a direct current bias is applied to provide the charge to set up the capacitor effect and replenish the charge that is leaked off the plates. A very slight sound pressure will cause a slight depression in the movable condenser plate, therefore allowing more charge to be stored from the biasing supply. This change in charge can be amplified by a very high impedance amplifier to a level and impedance that is usable as a sound source in a standard microphone mixer.

The electrette utilizes this same principle, but does not require the external polarizing potential. It employs a technique discovered by a Japanese scientist, and developed commercially by Sony, to make an inexpensive "permanently" polarized condenser mike element. It did not require a external polarizing potential of 48 to 52 volts, made a battery-powered condenser mike practical, and low voltage phantom power less expensive to use on condenser mikes. These factors have made the use of condenser mike elements approach that of dynamic mikes (almost every portable cassette recorder has a built in electrette mike).

The condenser mike has obvious application in choir micing, picking up a distant sound source, as in a shotgun mike, or in television/film boom and fish pole micing. The condenser also makes an excellent pulpit and lavaliere microphone. The electrette element has made the excellent response of condenser microphones affordable to more users. Condenser mikes do have drawbacks, however. The relatively high pressure sensitivity makes them sound a little too bright in less expensive models. Condensers are also more sensitive to wind, and to breath-blasting. Proximity effect (bass boost) is excessive in models not designed for close micing. However, almost all commercial recording is done with condenser mikes. Another disadvantage of condenser mikes is their increased complexity, which makes then more susceptible to failure under hard use (like on the road with rock and roll bands). The most important drawback not related to performance, is that of power being required for all condenser mikes. Some condenser mikes accept only a battery, or use special multi-pin cables and a remote power supply. Most professional condenser microphones use Phantom™ powering. This technique of powering condenser mikes does not require special multi-pin or non-standard cables, and allows the use of other types of pro-audio configuration mikes. Three-conductor cables are employed, with two signal leads ("positive" and "negative signal") plus a shield not connected to either signal lead. The power supply is superimposed on the signal leads in a manner which makes it invisible to the audio circuit. Although phantom power is possible with all professionally-configured mixers, it is not always provided, and not all mixers with 3-pin plugs are pro-configuration inside. This is especially true of those "bargain" feature-oriented brands sold in the MI business (music stores). Even those that have true balanced inputs may not isolate shield completely from the signal path, making external phantom power impossible without modification, or detrimental to the performance of the mixer. Common problems with using such mixers include hum or oscillation, which result from "ground loops".

Phantom power therefore answers the sticky question, "Where do I get power for my condenser mikes?", but not all condenser mikes accept phantom power. And, not all phantom power supplies provide the same voltage. Some microphones designed for 48 to 52 volt phantom power will not work with 9 to 24 volt phantom supplies. Phantom power also precludes the use of two-wire mikes. These are non-professional types, most commonly supplied with 1/4 inch, 1/8 inch, or Amphenol MC-1 (Screw type) plugs and usually supplied with consumer-grade tape recorders, or sold by hi-fi and electronic specialty stores. They will not work correctly without a transformer which "floats" the signal leads from the shield lead. I, personally, solve this problem by never purchasing these mikes for my own use, and providing transformers in my systems for adaptation of tape recorders and non-standard mikes. I think that they are often inferior in audio quality, because the manufacturer assumes that anyone who values quality will buy standard configuration mikes.

Wireless Mikes

Many things about wireless mikes don't go without saying. Wireless mikes are an investment in a product that requires ongoing maintenance and care. Batteries are an ongoing requirement, much as gasoline is for your car. It should be noted that rechargeable batteries are normally not satisfactory because of the way they loose strength gradually, while the alkaline batteries hold there output until just before they are exhausted. Many wireless transmitters require 8.5 to 8.6 Volts to operate correctly, however, fully recharged regular Nicad batteries only put out 8.3 volts which begins to fall off immediately. These batteries may be fine in electronics that are not so voltage dependant for reliability. Inconsistent battery strength can cause drop outs and other perceived wireless system problems. Battery economy that may cause your wireless to be unreliable at the most critical time is a false economy.

A second area of concern is that the small wires between the body pack and the mike are susceptible to breakage by vigorous or extended use. A backup 'plug-in' mike for your body pack is a fail-safe in the event you break a wire.

A third area of concern is mechanical stress placed on the circuitry in the body pack or handheld by vibrations or 'dropage'. Repair usually requires a trip to the factory repair center, since repair for the tiny circuitry inside requires special tools, FCC certified alignment, and techniques not available in the field.

In general, the more expensive the unit, the more reliable the unit within reputable brands. The expense of a unit however, does not guarantee a trouble free life without a consideration of the factors we have have listed here.

Monitor and Auxiliary Systems

The needs of the hearers in a platform area, soloist, and the instrumentalist are completely independent of the house or the choir. Specifically the instrumentalist and soloist need volume that is above that they generate since as performers they need to co-ordinate this will require a mix 'out of balance' with others mixes especially those in the house area who during solos and instrumental performance need a balanced mix not one tailored for co-ordination. The monitor mix should be independently adjustable input by input so instrument volumes can be set, a track volume can be adjusted loud enough for the performer, each without putting any directly in the house.

The Choir is unique in that they are both performers and a group of worshipers therefore there needs are a blend of the house and worship leaders needs. During many phases of the worship service the choir is worshiper just like the people in the house while at others they are worship leaders with unique monitoring needs. I recommend utilization of a send bus that can select a independent or dependant channel mix (Pre or Post Fader) depending on the choirs and sources place in the worship experience. Therefore tracks, piano, and other sources may be uniquely set while others where the choir is a worshiper group may be selected as dependant mix responding to changes on each channel fader.

Adding Remote Speakers

Auxilary Systems Using 70 & 25 Volt Lines

Adding Speakers to your sound system outside the main auditorium or room is often one of the most confusing things for such a simple task. The difficulty is it envolves 2 widely different environments and power handling devices. Speakers utilized in a typical main room sound system have power handling of between 100-400 watts. small auxiliary speakers (ceiling or wall boxes) typically have power handling of 5 watts.

When you connect 2 - 8 ohm speakers to a power amplifier each will try to draw half the power. If you are using a 200W at 4 ohm amp (2 - 8 ohm speakers in parallel produces a 4 ohm load) each speaker can draw up to 100 watts for a given situation. If one of the speakers is rated at say 150 watts and the other say 5 watts and equal amounts of the potential 100 watts is sent to each and the 5 watt speaker will probably burn out and fail.

Most people when they try to use the method above buy a wall box speaker with a volume control and run a small wire from the main amp to the remote location.

The problems arise because the volume control is also a 5-10 watt device and will also burn up from too much power. The reason failure does not always occur immediately is the very small wire sometimes used "eats up" enough of the power as loss to delay the control and speaker burn out.

When failure occurs since a short trys to draw more power than the good speaker the system begins to have problems because of a shorted remote speaker. If the remote speaker burn out is like a light bulb then you have no sound at the remote location but the amp now "sees" only an open so no further problem occurs in the main room. If shorted speaker and/or control plus the small wire load persist it produces all sorts of problems to main room.

During a service call before a seasonal music program I discovered just such a problem. The complaints was our main house speaker has blown out. When I arrived and discovered a EAW speaker which I had never seen fail in service I became suspicious. Upon measuring the load on the house speaker line I discover a very high load and upon investigation I discovered why. Someone had tied a nursery speaker line across the main speaker line using telephone quad wire. Upon removing the nursery speaker the main house performed flawlessly.

How then do you connect remote speakers to your sound system the right way. If you recall the problem is balancing the power based on the speakers ability to handle power. The best solution is a separate small amplifier which provides power for the Auxiliary speakers. A postulate to this is providing a means to regulate the power drawn by each speaker location. Remember we mentioned that 2 - 8 speakers in parallel produces a 4 ohm load. Similarly 4 - 8 ohm speakers in parallel produce a 2 ohm load. How then do we balance the power load to an array of speakers in remote locations.

The solution engineered by the commercial sound business is the constant voltage distribution system. Specialized amplifiers provide "constant voltage" outputs of 25 & 70 Volts in addition to the 4-8 ohm speaker outputs. Small inexpensive speaker assemblies with transformers with multiple power selection taps (typically between 1/2 to 5 watts) and optional volume controls are provided for connection at the remote locations. These amplifiers produce 25 or 70 volts when the amp is driven to its rated output. The speakers draw the amount of power determined by their transformers power tap selection from either a 25 or 70 V speaker line. Because these inexpensive amplifiers and transformers are not precision it is recommended that the total load be only 80% of the amplifiers rating. To balance or determine the load simply add up the power tap selection of all the remote speakers. The speaker taps should total no more than 80% of your 25 or 70 volt amps rating.

Example of a typical auxiliary system

Narthex 3 - 5 watt taps ceiling speakers with controls 15W total

Nursery area 3- 5 watt taps wall box speakers with controls 15W total

Church hall overflow speakers 6- 2.5 watt taps speakers & controls 15W total

grand 45W total

The Power Amp chosen has a 60W rating times .8 equals 48 watts adequate for the system proposed.

You should use 18-16 gauge wire for auxiliary system wiring to avoid power loss. It should also be noted local codes often require 70 v wiring to be in conduit so 25 v provides an alternative that allows wiring without adding conduit.

Other application for 25 or 70 Volt line system are systems involving multiple speakers like church halls with low ceilings, hotel meeting rooms, Stadiums with multiple or remote speakers, paging systems, and anywhere very long speaker lines are required (70 volt lines have much lower power loss for a given wire gauge and distance). The advantages of 25 or 70 volt lines come with a price . High quality speaker transformers are expensive with price going up considerably with higher power. Background music and remote speakers have lower quality expectations so less expensive speakers and transformers are usually chosen .

Special Hearing Needs

Special Systems for People with Special Hearing Needs

People with hearing impairment can have their special hearing needs met by widely available ALS (assistive listening systems) equipment which provides individual receivers and volume adjustment. The wireless systems add to the convenience to the hearing impaired giving the freedom of choosing any seat in the auditorium.

Outside the room auxiliary systems can provide audio material to hearers in both adjacent and remote locations. Providing provision for all of these systems in the design of your building is sometimes overlooked making later installation impossible, inconvenient, and often ugly cosmetically.

Additional Information is available in books available through our book page links.

Call me at 972 660 2001 about how you can turn challenges into a quality sound installation for your church with LBS assistance. You may also contact a me at Emailto:lbsound@flash.net if you prefer.