Worship the Lord

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Audio, Acoustics, and Video in the Worship Setting

Part 2: Acoustic Definitions and Implications / Microphones

By David Hosbach

In our first installment we discussed basic acoustic principles to maximize sound quality, tone projection, and speech clarity in the worship space.

In this installment we get into the sound system. Coined as a tongue-in-cheek statement by a sound industry icon, the following statement hits home: "If bad sound were fatal, audio would be the leading cause of death." Fortunately, bad sound is not fatal. But it can be upsetting both for listeners and for pastors on the receiving end of complaints from members who cannot hear clearly.

> *"If bad sound were fatal, audio would be the leading cause of death."*

Just what is sound? How does sound behave in the worship space? What does a designer look to accomplish when laying out a sound system? How should I place the microphones? Which microphone designs are used for which purpose?

And the dreaded "gotcha's." Why is the audio quality bad on recording feeds? Why do we have "hot" and "dead" spots in the space? Why do we have feedback? Why is the sound just not clear?

The goal here is to give you things to think about. These will help you to have an idea of what questions to ask and the knowledge that good sound can be achieved. Note: I invented the fictional characters Aunt Tillie and Uncle Charlie some years ago to personalize my talks. I have brought them along here. They, like any worshiper, have one thing in mind when they come to worship: they want to hear the Word—clearly. My job and yours is to make that happen. So let's dive in and see how we can prevent audio pain, futility, and fatality.

There are a few simple goals that any good sound system designer will strive to achieve. But first, an important definition:

Decibel – The decibel (dB) is a comparison with some point of reference. It is *not* a unit of power like a watt or volt. I can say that the electrical power in my house is 110 volts. I can't say that it's 110 dB. What I can say is that normal human speech is measured at about 65 dB. That measurement assumes a reference level of 0 dB, which is the threshold of human hearing. A difference in level of 3 dB is audible; 6 dB is very noticeable, and 10 dB is considered twice or half as loud, depending on an increase or decrease in volume. I cover this definition first because it is at the root of everything we discuss regarding audio design. Following are other clusters of definitions and their implications. These all are inseparable for achieving a goal of optimal sound quality and for microphones to perform well.

Loudspeaker coverage – We seek to deliver even sound levels throughout the seating area. The standard we use is plus-or-minus 3 dB from front to back and side to side in the seating area. If we keep the sound levels within that window, then there will be no audible volume level differences. The "gotcha" will be intelligibility, or clarity.

Intelligibility – How clear the sound is.

Volume – How loud the sound is.

"Gotcha": There is a distinct disconnect between volume/loudness and clarity/intelligibility. Aunt Tillie approaches an usher or the sound operator during worship and says, "I can't hear." The usher or sound operator turns up the volume. But did that make a difference? The sound is *louder*, but how *clear* is it? If the ambient sound is too loud, if the worship space is excessively reverberant or fraught with slap echo issues, or if the loudspeakers are not directional enough, then volume is not going to help. We need to ask Aunt Tillie, "Is it volume or can you not hear clearly?"

Loudness 15 dB above ambient noise level – The clear sound levels from the sound system must be more than twice the level of background or ambient noise. Recall our case study in the previous installment. The ambient noise level from the HVAC system was measured in excess of 70 dB. Normal human conversation is measured at about 65-70 dB, depending on male or female talker and strength of voice. We want the sound from the sound system to arrive at the listener's ear at about 70 dB in order to make the sound comfortable. If the ambient noise level is already at 70 dB, then the sound from the sound system is totally masked or covered up and needs to be at nearly rock 'n' roll levels, at least 85 dB, in order to be heard above the ambient noise. Those levels will drive people away.

"Gotcha": You can count control of HVAC and other ambient noise as major. If you don't control that ambient noise, the sound system, no matter the quality of components, will not be able to make up for it without irritating the listeners. Keep that ambient noise level at about 55 dB. Then I as a designer can deliver a most comfortable 70-75 dB to the congregation.

Direct Sound – The sound we receive directly from the sound source before reflections.

Early Reflected Sound – The sound we hear very soon after the direct sound, maybe off a wall near the loudspeaker. This is usually not a deterrent to intelligibility and is desirable especially for good music projection for choir and organ. (This is why we see shell type walls to the sides of choirs and organ pipe chests.)

Reverberant Sound (also Late Reflected Sound) – The ambient sound, which is made up of HVAC noise, people whispering and rustling paper, and other acoustic and loudspeaker sound that has reflected off walls, balcony faces, and ceiling.

Deliver primarily direct sound – God designed our ear/brain mechanism to receive a syllable, process it, then receive the next syllable. Aunt Tillie, Uncle Charlie, and all of us need to hear the syllable once and let it pass so we can get the next one. When we hear echoes or when the background noise is too strong, then that God-designed hearing process gets messed up. Our ears get tired, our brains shut down, and we tell the usher, sound guy, or pastor that we can't hear.

There is not a written standard to tell us that we need "x-amount" of direct sound for maximum clarity. Experience has shown me and other designers that we need to achieve at minimum 75% direct sound as compared to reverberant sound in order to deliver good speech intelligibility. As mentioned above, we like to have some early reflected sound as well to give us depth for music; this can give dimension to speech as well.

How do we deliver 75% direct sound? We get this result in two ways. First, recall the acoustic principles discussed previously. We need to get the room acoustics right: live but not excessive reverberation, no slap echo and no flutter echo, and control of mechanical noise from HVAC, fans, etc. Secondly, we need to ensure that the sound system receives energy directly from the talker into the microphone, and then that it emits the sound as directly as possible from the loudspeaker to the listener's ear.

How do we make that happen? Let's get into a few more definitions and principles. I think that the answer will become clear with this slightly deeper dive into sound principles. It will also answer some more "gotcha's."

Sound – Sound happens when something vibrates. Stretch a rubber band and pluck the elastic. The elastic moving back and forth creates air pressure zones, like the high- and low-pressure zones you see on a weather map. The sound is carried along in waves (hence the expression sound waves) along those pressure zones.

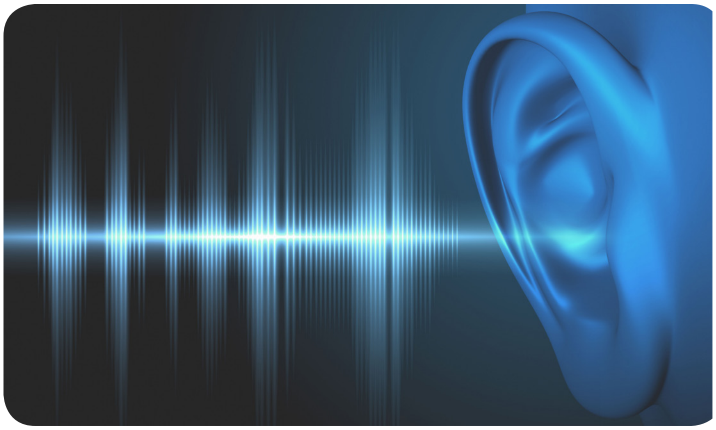
Frequency – The speed at which the elastic moves back and forth, or more specifically, how many times the elastic vibrates in one second. The faster the elastic vibrates, the higher the frequency which is labeled "hertz" (Hz). The slower the vibration, the lower the frequency. We can see those sound waves and their speed on an oscilloscope. The scope will show us wavelength.

Wavelength – the portion of one second occupied by a single vibration. The longer the sound wave, or wavelength, the lower the frequency. Moving in that direction takes us into the bass range. The shorter the wavelength, the higher the frequency, and we move into the treble range.

The table below shows how wavelength and frequency relate:

<table>
<tr><th>Frequency in Hz
(by Octave)</th><th>Wavelength</th><th>Related to Piano Keyboard</th><th>Related to Musical Instruments
and Human Voice</th></tr>
<tr><td>10,000</td><td>1.4 inches</td><td></td><td>Upper end of most musical instruments</td></tr>
<tr><td>8000</td><td>2 inches</td><td></td><td rowspan="7">SPEECH RANGE
'S' sounds in 4000-8000 Hz range
Consonants in 2000-4000 Hz range

Vowels in 250-1000 Hz range</td></tr>
<tr><td>4000</td><td>4 inches</td><td>Top Key 'C'</td></tr>
<tr><td>2000</td><td>6 inches</td><td></td></tr>
<tr><td>1000</td><td>14 inches</td><td>'C' 2-octaves above 'Middle C'</td></tr>
<tr><td>500</td><td>27 inches</td><td>'B' above 'Middle C'</td></tr>
<tr><td>250</td><td>5 feet</td><td>'B' one step below 'Middle C'</td></tr>
<tr><td>100</td><td>10 feet</td><td>'G' below 'C' below 'Middle C'</td><td rowspan="2">Bass Vocal in 50-100 Hz range</td></tr>
<tr><td>50</td><td>20 feet</td><td></td></tr>
</table>



The table may not mean much to many people, but it's important when we discuss acoustics and audio. And I think it brings clarity (pun intended) to the whole direct sound and intelligibility discussion. Paying attention here eliminates a myriad of "gotcha's" before they can happen.

The longer the wavelength, the stronger it is and the more difficult it is to deal with. As you read the following points, think of wavelength simplistically as a beam of light with finite edges projecting out from a source.

Example A. The lower we go in frequency, the wider and stronger the beam and the more difficult it is to block or absorb those frequencies. The obstruction required to fully block the frequency must be about equal to the size of the wavelength.

Acoustically speaking, the material thickness and density required to absorb lower frequencies must be substantial. A felt banner or thin drapery will be invisible to a sound at 1000 Hz and below.

A pillar of three feet in diameter will not block a sound at about 400 Hz or below. On the other hand, that three-foot diameter pillar will cast a "shadow" and block sound from 500 Hz and up.

"Gotcha": The installer just mounted thin column speakers on the front pillar in the church. People seated behind the subsequent two-foot diameter pillars say they cannot hear. The reason? The frequencies in the upper vowel range and consonant and 's' ranges are blocked by the pillars. People seated behind those pillars will not hear clearly. The installer needed to locate support speakers at the subsequent pillars.

Using the table as a guide, we can see that an obstruction of as little as six inches can damage intelligibility. We need to keep that in mind as we go about achieving our design goals.

> *An obstruction of as little as six inches can damage intelligibility.*

Example B. In audio systems, the lower we go in frequency, the more difficult it becomes to control, aim or "steer" those frequencies. Conversely, it is easier to control higher frequencies. In simple terms, we need "acoustic buckets" large enough to contain the frequency range we want to control.

Typically we use horns or other highly directional array type loudspeakers in a space, no matter the size. The horn or array dimension will dictate to what extent we can control where sound is aimed. We can use a small horn or device with mouth dimension of about 8 inches to control from about 1,500 Hz and up. That's good for consonants and 's' sounds. In order to control the projection of a 500 Hz sound (the vowel range), we need a device (horn, array height, etc.) of 28 inches; for 100 Hz we need a 10-foot horn!

"Gotcha": The installer just installed one or two loudspeakers, suspended from the ceiling. The total dimension of each enclosure is about 26 inches. The installer tells you that the speaker will deliver very good clarity. He is right—to a degree. What he failed to tell you is that the enclosure contains a high-frequency horn that measures about six or seven inches. It will control the consonants well. There is a 15-inch diameter bass speaker in the enclosure as well. The midrange and bass frequencies will be well-supported but not controlled. At 15 inches in diameter, that woofer cannot contain the wavelengths below about 1000 Hz, meaning that the vowel sounds are allowed to bounce around the space. Those vowels now mask the consonants and develop excessive reverberant sound. In a reverberant space, Aunt Tillie and Uncle Charlie won't be happy.

> *Aunt Tillie and Uncle Charlie won't be happy.*

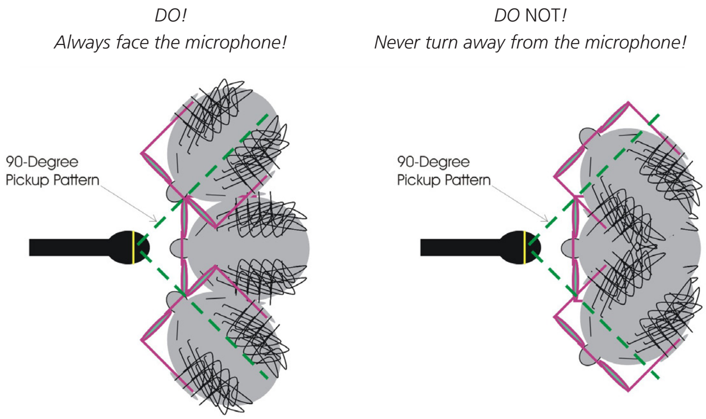
How do we make them happy? We need to utilize loudspeakers that are directional not just in the high frequencies, but also in the mid- and low frequencies. If we can control from about 500 Hz and up and deliver even coverage, then we have a very good chance at making every worshiper happy.

Okay, we've spent a lot of time on loudspeakers and direct sound. We needed to because if we don't get that part right, what we do with the rest of the sound system will not matter. But now let's move to microphones.

> *There is a difference between microphones. You will get what you pay for.*

There are a lot of brands and types of microphones available at virtually every price point imaginable. I need to encourage you up-front that there *is* a difference between microphones. You *will* get what you pay for. Many microphone issues are caused by one or more of three basic "gotcha's":

1. **Very inexpensive microphones were purchased.** Usually, the very-inexpensive microphone is not articulate or clear in the consonants. It may not sound natural. And it may not pick up well for what you need it to do.
2. **The wrong microphone type was chosen for the application.** A typical handheld microphone will not work well at the lectern, pulpit or ambo, or for the choir. They are designed for up-close solo work. Unless your talker can position their mouth within about three inches of the microphone, this is not the right mic.



Use a good long gooseneck microphone for the lectern, ambo, or pulpit. Use a similar type on a stand for the choir when needed as they are more "forgiving" in their pickup pattern and more sensitive and so will pick up from a greater distance—about eight inches at the lectern, pulpit, or ambo; and with greater gain (volume) applied, about two feet from the front row of the choir.

Use a good discrete headset/ear mic (definitely not inexpensive) that fits well for the pastor's wireless. These are designed for live sound, as opposed to lapel mics which were designed for use in TV news studios many years ago.

3. **The talker is not positioned properly.** The talker must be within the clear operating range of the microphone. This means within about three inches of a handheld solo mic, no more than eight inches from a gooseneck microphone, and two feet from the front row of the choir. Within these distance windows the sound pickup will be full and articulate. The sound level will also be strong for feeds to recordings and distributed speakers in cry rooms and such. Outside these windows the sound will become thin, volume will be low, feedback may occur when trying to increase the volume, and Aunt Tillie and Uncle Charlie will wonder why the sound is bad.

4. **The talker is not talking to the microphone.** This relates to the former point, but deserves its own block. You must address the microphone. In other words, the mouth must be pointed toward the microphone head. Just like we need direct sound from speaker to listener, we need direct sound from talker's mouth to microphone. If we talk "away" from the microphone, or physically turn away from the microphone to make eye contact somewhere, we will turn out of the microphone's pickup pattern and sound level and clarity will be lost.

   This is why the headset/ear mic has become popular and even better than a lavalier/lapel mic. But if you have a mounted mic, maintain a distance and position relationship with the microphone. When turning to make eye contact, turn your body about the microphone so that you can look the other way without looking away from the microphone. As was stated in the former point, this will help everything: sound levels, clarity from the loudspeakers, recording feeds, and even hearing assistance systems.

5. **The microphones are placed ahead of the loudspeakers.** Feedback occurs when sound from the loudspeaker reflects around the room and back into the microphone. It can occur when the microphone is placed in front of the loudspeaker. If that speaker is too close to the microphone, or if the overhead speaker is turned up and microphone is out front, feedback will be an issue. Proper system tuning (next installment) can help somewhat, but the needed clarity and volume levels will not be realized. And feeds to recordings will suffer as well.

My design philosophy has always been simple: use the right microphone and loudspeaker, set them up properly, and the sound will be right at all destinations. I won't need to play tricks to try to get the system to work right. If you follow the same philosophy, you will be successful as well.

> *Use the right microphone and loudspeaker, set them up properly, and the sound will be right at all destinations.*

Next up: system setup—good stuff and "gotcha's." Also, the hearing assistance system: why do we need it, and which type is best?

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