AVTECHNOLOGY MANAGER'S

Workbook

Delivering The Best Live Sound Experience for Indoor Spaces

LEARNING OBJECTIVES

- → The Connection Between the Room, the Audio System, and the Audience
- → Understanding the Audio Spectrum as Applied to the Intended Room Use
- → Designing the Audio System
- → Selecting Loudspeakers, Placement, and Understanding Pattern Control
- → Additional Audio Equipment Choices
- → Selecting the Correct Microphones for the Venue
- → Acoustical Treatment and Acoustical Replacement Systems

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Listen Up. Audio is as much of an art as it is a science. In the world of integrated audio systems, there is no "one-size-fits-all." And yet, AV and IT managers are expected to magically appear at a corporate theater, boardroom, or college campus, clip mics on presenters, adjust goosenecks, and ensure a solid sound performance. A good technician needs to know how sound systems work and understand parametric equalization in order to eliminate unwanted ringing. Loudspeaker system design, room acoustics, and the interplay among them are highly complex, but there are essentials every manager must understand. Think of this Workbook as your "sound reference" for distributed audio applications, live sound best practices, and every decibel in between.

Margot Douaihy,
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HARMAN

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Content

AVTECHNOLOGY MANAGER'S Workbook

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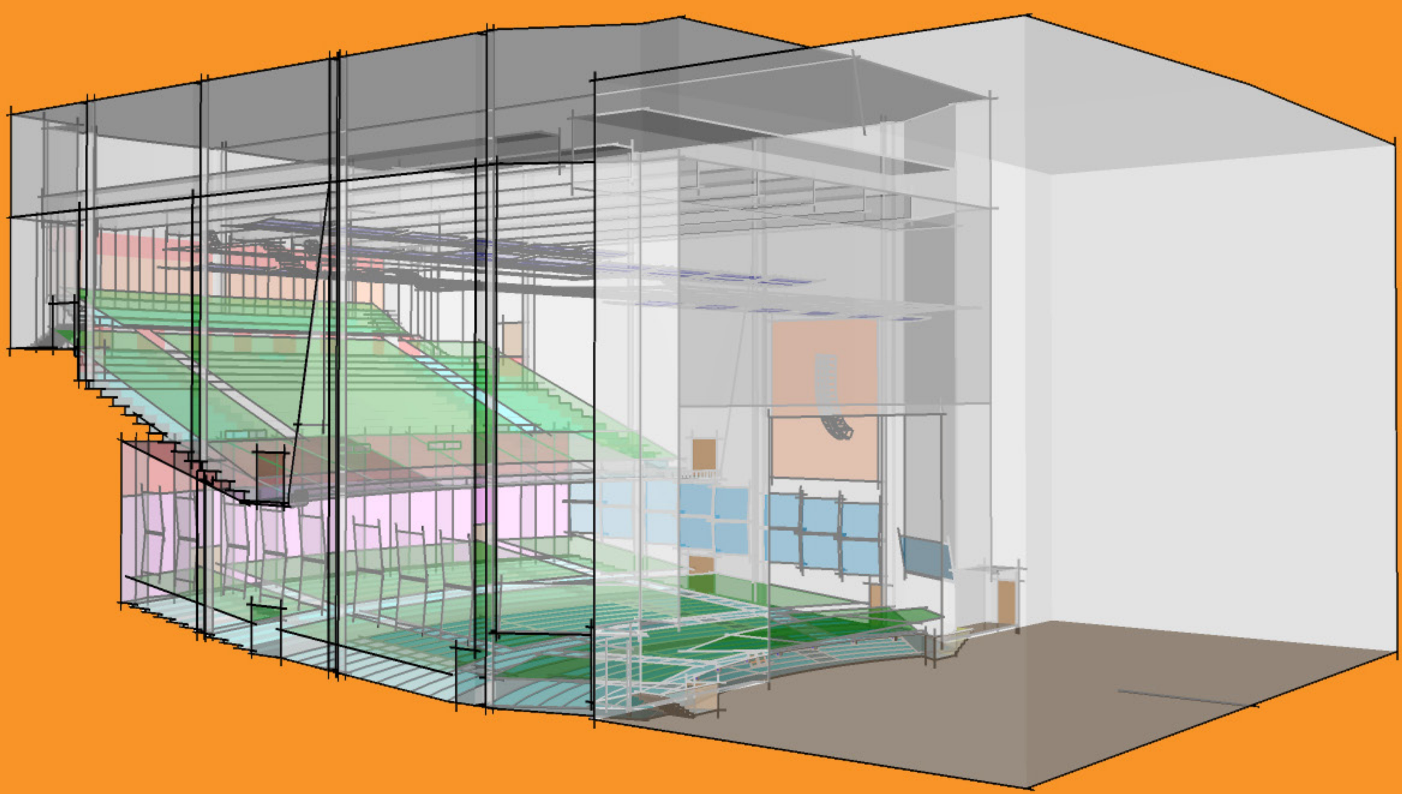
As a key interface to the contracting and consultant markets for HARMAN Professional Solutions, Fullmer has extensive experience in audio system design for an enormous variety of designs and installations. As an experienced electro/acoustical system designer and problem solver, he supports the Installed Sound, Tour Sound, and Cinema Markets, and has been involved in product development as well.

With more than 30 years in the audio industry he has held many positions, including system design and virtual modeling specialist, live audio production designer and engineer for theatrical, music, and corporate events, and system installer.

Fullmer enjoys teaching training classes for HARMAN Professional Solutions, and is a past AES presenter. As a reformed semi-professional musician, he still finds time to play live music and thus retain a connection to what got him started in this field in the first place.

Creating the Best **EXPERIENCE**

The audience experience for public spaces has been on an upwardly mobile path. The how and why can be debated, but the fact remains that the level of acceptability for what is considered “good audio” in public spaces has experienced an upward trajectory for several years. As designers, specifiers, installers and technicians, we must also have a similar path … upward! Keeping up with current technologies demands that the rest of our “game” is solid. This workbook offers points for consideration as you deal with new challenges and the diverse variety of projects that present themselves to the AV professional. Moving upward from a **Good Experience** to the **Best Experience** is the goal.



It is often said that history repeats itself, and as we look at several trends in new audio technologies, we can see glimmers of the past, but with a modern alignment. The column loudspeaker from the 1960s, for instance, was prevalent in both houses of worship to a variety of multi-use spaces. That same basic geometry—slim, tall, and even considered handsome by some interior designers—has returned, but with electronic additions only imagined several decades ago. We now have the ability to not only control aspects, such as vertical directivity, and to steer the beam, but also to shape the beam to the audience plane and easily and accurately predict this behavior.

Even with the advent of affordable signal processing and built-in feature sets that would leave our AV pioneers speechless, there are still situations that require mighty planning and a well thought out strategy. These scenarios can often be found at the intersection of old and new. As in a really old building, and a seemingly non-conforming space used by the client—and one that may not have even existed when the building in question was erected! Having designed dance club systems in historic bank buildings, aerobic spinning studios in converted glass greenhouses, and high-impact contemporary church music systems in spaces that were designed for longer reverberation times—meant to enhanced choral and organ music—nothing should surprise us. But occasionally, The Laws of Physics as tested by a demanding client can do a pretty good job.

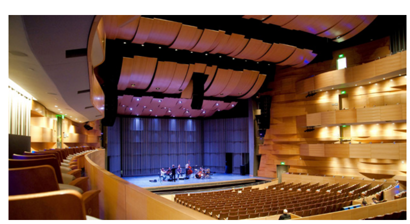
The delicate balancing act of managing several key parameters is where we proceed, from:

» The **expectations** of the client or end user have to be fully considered, but first they have to fully reveal themselves.

» The **performance goals** need to be clearly established prior to commencing with a design.

» The **budget guidelines** have to be accounted for, even with a client initially stating that no fixed budget exists.

With these three factors actively managed, we can proceed forth …



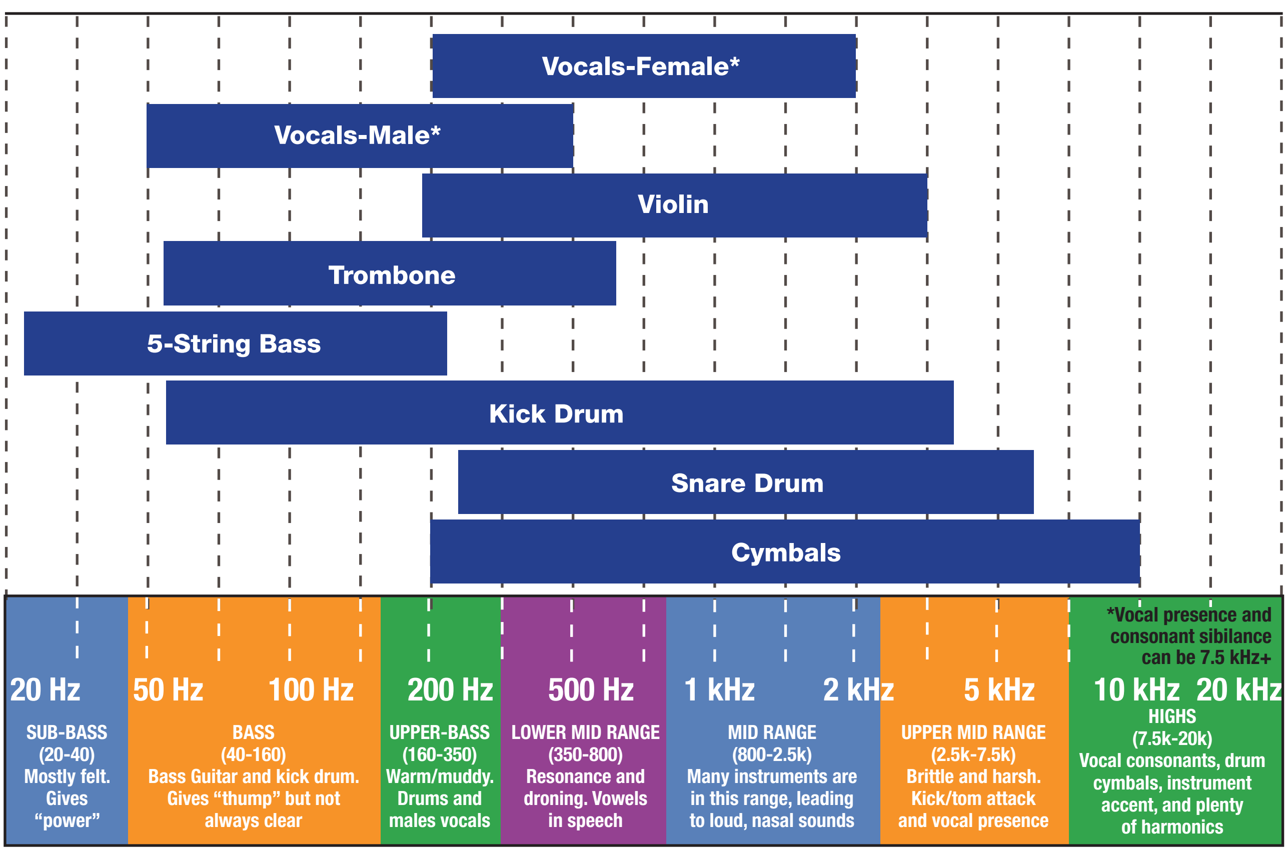
Understanding the Audio Spectrum as Applied to the **Intended Room Use**

'HIGH FIDELITY REQUIRED!' OR 'HOW MUCH FIDELITY REQUIRED?'

If we have a small lecture hall with the primary purpose of spoken word to be delivered in an intelligible manner, we should concentrate on the range from 125 Hz to 6 kHz. Many of the systems we would typically specify will meet that criterion with extended response above and below those figures. If audio playback with attached music tracks is required, additional low frequency elements can be added. For low frequency extension, plan on adding one 18-inch device for every 25,000 cubic feet as a baseline. The table below illustrates several frequency ranges of instruments and vocals.

Audio wavelengths are measured in cycles per second, or hertz (Hz). So, for example, 440 Hz, the frequency of the A note above middle C on a piano, equals 440 cycles per second. The audible

The Audio Spectrum



spectrum for humans is 20 Hz to 20 kHz (20,000 Hz). For live or recorded music, the entire audible spectrum needs to be considered. For speech intelligibility, sound system designers need to pay particular attention to the middle range of frequencies, generally from 500 Hz to 4 kHz. However, even for speech-only systems, the frequencies below and above this band are important for imparting a natural sound character to voices, and for avoiding boomy, or thin-speech character.

Each portion of the audio spectrum imparts its own challenges due to differences in wavelengths. High frequencies have short wavelengths. A 20 kHz wave is 0.50 inches (12 millimeters). Lower frequencies have longer wavelengths. A 20 Hz wave is 50 feet (12.7 meters). These wavelengths behave very differently, and react different when they encounter walls, floors, seats, or sounds in other wavelengths.

Whereas with outdoor stadiums and systems for larger areas and coverage distances, we may be less concerned with frequencies above 10 kHz, largely due to air absorption loss, but in the indoor environment, we need still need to support these upper octaves.

UNDERSTANDING THE ROOM'S ABILITY TO HANDLE LOW FREQUENCY INFORMATION

If the source material of the room in question requires more low frequency reinforcement, how do we proceed? A threshold commonly referred to as, "the point of diminishing returns," which is typically used in reference to economics but also applies to audio, states that an acoustic space will accept only "x amount" of low frequency information. Beyond that volume, additional low frequency is not only impractical, it won't make a difference.

WHY FULL BANDWIDTH ACOUSTICAL ABSORPTION ISN'T ALWAYS AN IDEAL SOLUTION

Quite often when the room under investigation is too "live," the response is often to add absorption. This is an understood reaction if the case study is a large reverberant church that now supports more contemporary services. The addition of standard absorptive panels that feature that extend absorption into high frequency range will achieve most of the goals, but may be detrimental to music playback, as the offending range of reflective energy may exist only in the mid and low frequency range.

Best Practices

- ➔ **Develop a complete understanding of the room's characteristics and intended uses.**
- ➔ **Perform acoustical investigations and assessment prior to modeling (existing construction only).**

Every space has different reflections (that is, they have different points where sound bounces back) that can alter the way audio sounds at different spots in the room. The goal of sound reinforcement design, then, is to account for these reflections and put together a system that will provide the perfect listening experience for the event that is consistent throughout the space. Whether a live event is hosted in a ballroom, an auditorium, a gymnasium, a New York City concert hall or a theater, the acoustics can make or break the experience, and every seat in the house needs to be the "sweet spot."

The goal of sound reinforcement systems is to reinforce the existing sound experience. It is generally speaking, not intended to add anything, but rather to amplify and distribute the sound that is already there. When things are added to the original sound, this usually provides less than desirable results, because what is typically "added" is echo, feedback, and noise. There are a lot of things that can cause these unwanted additions, but many of them go back to the way the audio system interacts with the acoustics of the space.

WHEN TO ENGAGE THE ACOUSTICAL CONSULTANT IN A PROJECT

Given the differing nature of the acoustic spaces one encounters, a cookie cutter approach to sound control will yield mixed results. Broadband products that can be effective on severely problematic rooms are not usable across the board. Budgetary considerations for this addition are best included in the initial estimates for the client.

The expertise of an acoustical consultant can be a required element to allow us to achieve that elusive "best experience." The types of services offered by a firm or individual can range widely in scope and project involvement, but can be an overlooked aspect to the success of the venue. Often times it makes sense to evaluate the project's overall budget and gauge the potential end results with and without acoustic treatments. AVT

DESIGNING THE BEST Audio System

The ideal audio system is one that is unobtrusive yet allows the performer, or orator, in the words of Cicero, "To prove your thesis to the audience, to delight the audience, and to emotionally move the audience." While we don't necessarily want to provoke a 'Kumbayah' experience in the boardroom or corporate theater system we've recently completed, we ideally want the audio system to function in a near invisible manner. With a correctly designed audio or media system, the client's needs are met effortlessly and without frustration.

This process must start with a comprehensive needs assessment. The client's current methodologies and routines must be examined, types of current and future functions notated, and their expectations understood. It was once said that, "the main cause of anger and frustration is unfulfilled expectations." This is certainly true when a key aspect to a finished system was overlooked.

Different methodologies are often undertaken when a new design is commenced. Some designers tend to start with the loudspeakers, the last line in the audio chain, and work backwards. Conversely, others start at (or near) the beginning of the signal chain and work forward, defining the distribution network and infrastructure as needs develop.

Depending on the complexity of the design at hand, the distribution infrastructure is important to establish immediately at the front end of the design process. In many instances, a digital network will be a requirement for connecting various zones, or control points within the system. Switching topology and cabling decisions should be flexible enough to handle future growth and expansion. Many proprietary routing and control platforms exist these days (and will not be specifically addressed here), but Ethernet or fiber, and in many cases both, will handle the bulk of transport and control. Mixing surfaces and routing DSP platforms will need to be decided upon, with a nod to remote and wireless control surfaces. Once again, the needs analysis will help determine what control functions need to be placed where in the facility. Cabling and remote powering needs for touch panels, etc., should be addressed at this time as well.

Another variable that needs to be considered is the skill and/or technical abilities of the system operators. Many times, an alternate turnkey mode will be a requirement, as is often found in House of Worship environments. The Wednesday night women's group, which requires a single microphone, probably has no interest in joining either the media team, or mixing Front of House. They just need to hit a "go" button.

As the front end of the system starts to develop, it's probably time to start considering the physical movement of sound via amplifiers to loudspeakers. Availability of real estate for amplifier rooms and their air circulation requirements need to be considered both with new construction and in renovated spaces. No available amp rack room? The wider availability of powered loudspeakers may be a required alternative to traditional unpowered units. Provided that electrical power service can be delivered to the loudspeaker locations, along with signal and control cabling, the timeframe for getting a powered system up and running can be much less than their passive equivalent. This can be a potentially deciding factor for a renovation project on a restricted timeframe, for instance.

Best Practices

- ➔ **Fully comprehensive and completed needs assessment**
- ➔ **Full buy-in from interior design and architectural team prior to commencing of design**

After the order of magnitude of the loudspeaker system has been outlined, the structural implications of the loudspeakers' weight must be factored in for surface-mount or flown systems, or on a small-scale system, the positions of in-ceiling speakers must be decided. Review of potential rigging weights need to be cleared with the associated structural or professional engineering firm attached to the project, or subcontracted by the integrator.

It is also during this time that other trades may be outlining their plans and physical space requirements for lighting, video, interior design elements, etc.—and the real estate battles commence. To the interior designer, their wall surface designs and textures may not accommodate unsightly large loudspeakers. The lighting consultant will state their needs for specific placement of fixtures to complement the interior designer's vision. And then the AV specialist has to offer their drawing sets, with all layers turned on, adding video projectors and loudspeakers.

The case must be made for requested speaker locations with more than just verbal justification. Electro-acoustic modeling results, showing estimated frequency coverage and boundary interactions, should be prepared and presented to offer empirical data for review and consideration. If the design-build firm or contractor isn't able to produce these frequency plots, they can be produced by a third party, or even by a manufacturers technical specialist.

Loudspeaker zoning and discreet area control should follow usage as outlined from the needs analysis. This process will need to start during the definitions for audio routing and transport, but will be equally relevant at this point in time, as the loudspeakers selected for specific zones or coverage areas will need to have specific attributes and high-frequency pattern control characteristics. In many cases, high frequency horn patterns will dictate the enclosure selection, but there are other factors that may dictate that the frequency where vertical pattern coverage starts is the decision-making factor.

As an example, let's consider an upper balcony seating area in a performing arts theater that supports louder pop and country artists. A left-right main line array system with ample vertical coverage can address these upper seats, but potentially create a reflection that would carry into the upper areas of the theater, add to the reverberant field, and potentially affect speech intelligibility for these upper seats. By adding a delay group of 12-inch, 2-way enclosures, the direct-to-reverberant ratio increases, but below the crossover frequency the vertical pattern control starts to widen. When compared to a smaller three-element line array, alternately used in the delay area, the increased vertical pattern control of the three vertically oriented woofers extends the pattern control to a lower frequency, further reducing energy outside of the intended coverage area. These changes can also greatly affect speech intelligibility. AVT

SELECTING **LOUDSPEAKERS**, Placement, and Understanding Pattern Control

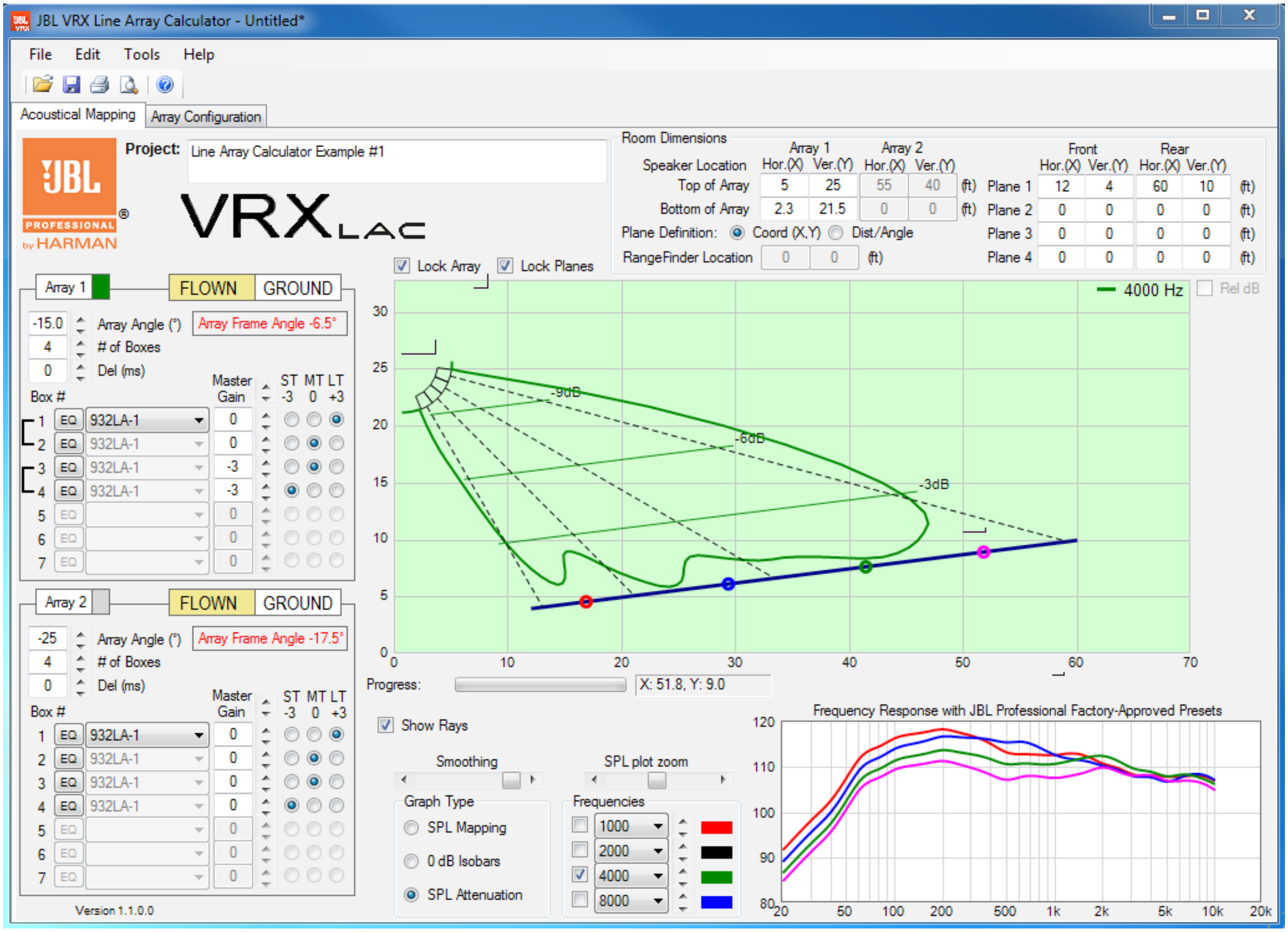
With consideration to the intended program material and required SPL and low frequency impact goals understood (as developed during the comprehensive needs analysis), we now embark on what is typically the largest single equipment expense of the audio system: the loudspeakers.

Boardrooms, lecture halls and corporate theaters often have speaker systems to address basic needs, such as voice lift/reinforcement and playback for video sources with attached audio. In many cases the monaural voice reinforcement system will function separately from the stereo audio playback system, which locates speakers adjacent to the video screen. In smaller sized rooms, with a lower ceiling elevation, flush mounted in-ceiling speakers will often be used for voice lift and remote mix-minus conferencing systems, using AEC (acoustic echo cancelling) circuitry. Speakers flanking the screen can often be incorporated into front wall recesses and may include hidden surround channel speakers as well. Subwoofer augmentation may be added to recessed front wall locations as well, or with multiple locations to counteract room modes and offer a more consistent subwoofer experience.

DISTRIBUTED AUDIO SYSTEMS

Areas such as office areas, hallways, and lobbies often implement distributed audio systems, rather than using audio systems where the attention of the sound is focused on a single point, such as a stage. Distributed sound systems consist of multiple speakers that are distributed throughout the location. The most common type of distributed system uses speakers projecting down from the ceiling, each covering a specific area.

In distributed audio systems, several loudspeakers are typically powered by a single amplifier. The location of the speakers within the room could require each to be set to different volume levels to provide even volume coverage, and are often powered at different levels. The calculations involved in determining the actual load impedance at the amplifier's output can be quite tedious, which is why constant voltage (70V/100V) distribution systems are popular. Constant voltage systems make calculations more simple and straightforward, as they provide a fairly high voltage to the speakers. Many loudspeakers can be placed across a single 70V speaker line, using distribution transformers at each speaker to convert this high



Supplied by most loudspeaker manufacturers, a line array calculator can offer accurate estimates of SPL coverage and will output array parameters, such as physical size, weight, and rigging requirements, for the structural design team to review. The figure above shows acoustical mapping and frequency response.

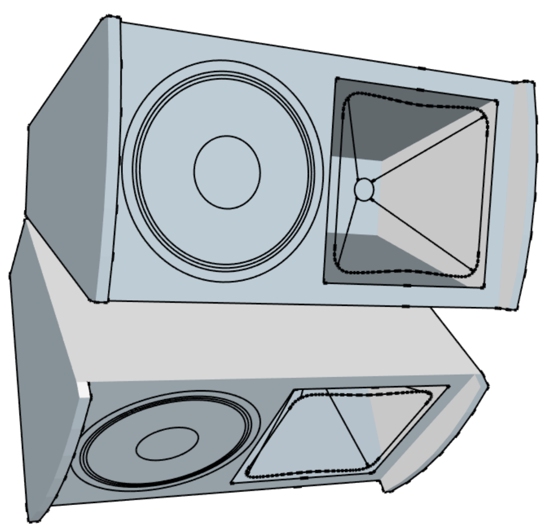
voltage back down to the proper level for each speaker.

Small- to midsize-lecture halls, and corporate theater settings often have a raked (sloped) floor with decreasing ceiling height toward the rear of the room. In these types of spaces, a single location voice reinforcement speaker system is ideal, as speech intelligibility tends to be higher with a single point source. If the distance from the speaker location to the near listener (D1) is less than a 1:3 ratio to the far listener (D2), then a single speaker may work fine in a room with a shorter reverberation time. With a greater distance ratio, multiple enclosures may be arrayed in a near/far orientation.

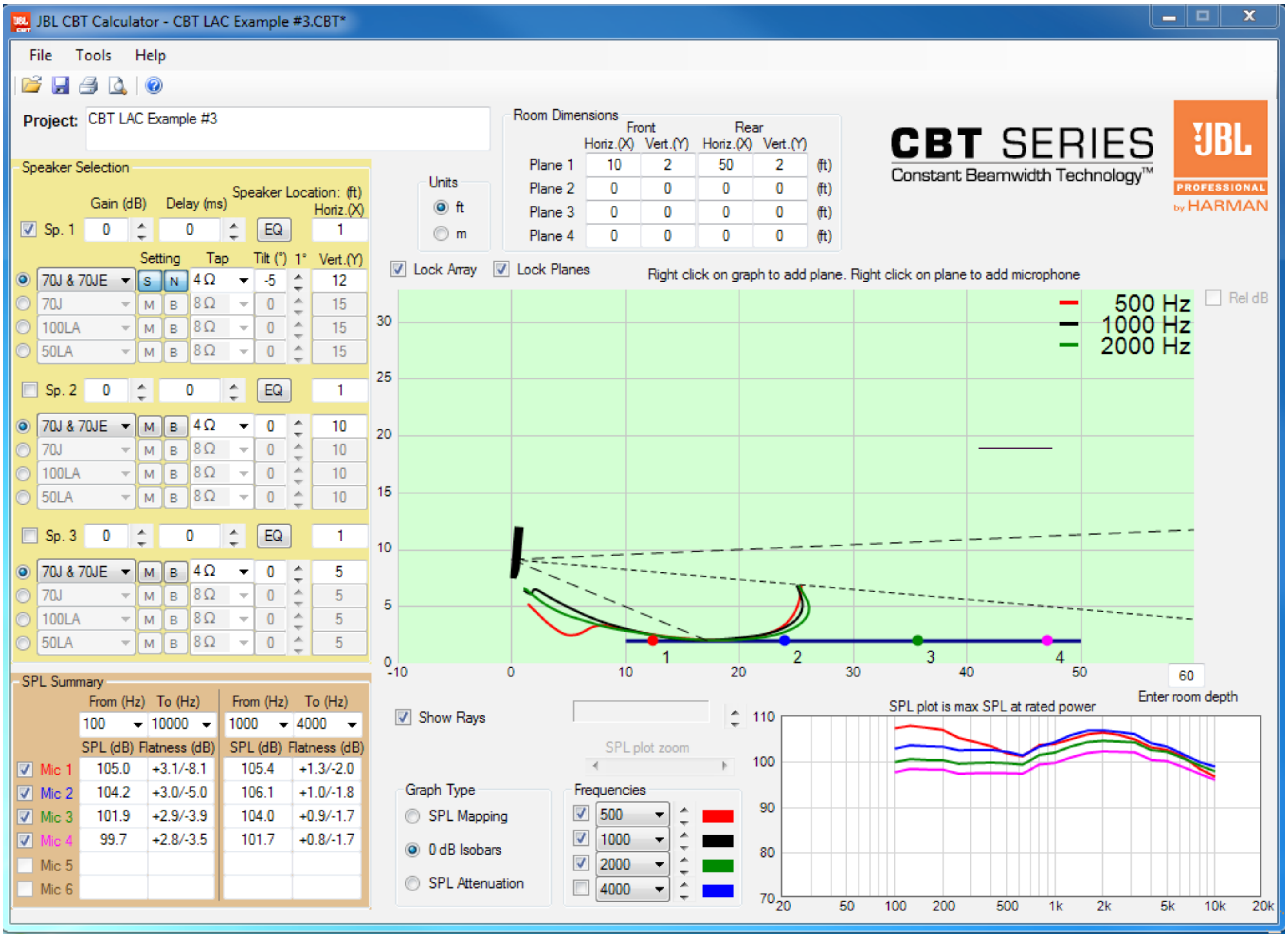
LINE ARRAY

Smaller line array-type systems can be very effective in these cases with smaller speakers used left and right of the array to fill the front outer seats with mid- and high-frequencies. These outer "wrap" speakers are typically high-passed in the 250 Hz range. In a narrower room, these outer fill speakers may not be required.

When estimating coverage using line array or column loudspeak-



Example of a two-box array



The above example demonstrates how selecting a speaker from the menu at the top left, and adding microphones, the line array calculator will create an SPL summary.

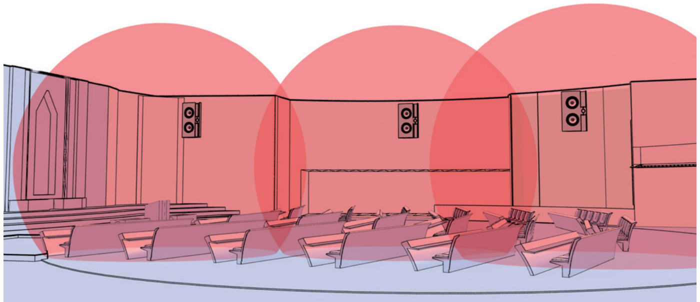
ers, a variety of tools are available to assist in developing a rough order of magnitude.

CALCULATORS AND MODELING TOOLS

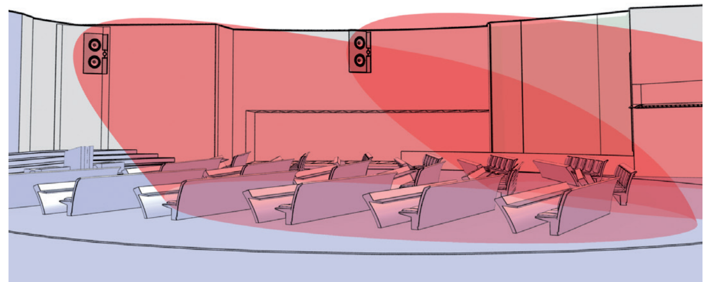
If a line array system is called for, based on the requirement of uniform front to rear SPL coverage and improved low frequency impact, then a quick check with a Line Array Calculator can be useful. These are supplied by most loudspeaker manufacturers and offer accurate estimates of SPL coverage and will output array parameters, such as physical size, weight, and rigging requirements for the structural design team to review. The advantage of this type of system is the ability to contour the main lobe, or wave-front, to the audience seating geometry. The two types of multiple enclosure line arrays are constant curvature arrays, which have an equal adjacent splay angle between box centers, and articulated arrays, which allow adjustment of the enclosures angles and thereby allow the shape of the array to cast greater energy to the far seating areas and less energy to the near audience seats.

For smaller rooms that would benefit from the aesthetic advantages of a column loudspeaker, there are calculators for these types of products as well.

Another set of tools that are especially useful in highly reverberant spaces allow both beam steering and beam shaping to closely match the vertical pattern to the existing seating geometry while minimizing wall interaction. With this technology, an acoustic model of the space is generated, using proprietary software, and then exports the complex Finite-duration Impulse Response (FIR) coefficients to the loudspeaker. With the pattern control offered by these types of devices, the direct to reverberant ratio is much higher than by using conventional solutions in these challenging venues. Overcoming difficult acoustic spaces while retaining high Speech Transmission Index (STI) levels is key to our "Best Experience" system.



Low-directivity loudspeakers distribute the sound evenly within a room and generate a lot of reverberation.



Directional loudspeakers focus the sound more effectively onto audience areas, creating less reverberant energy in the room and adding to improved intelligibility.

Best Practices

- ➔ **Structural steel and mounting surface analysis (including above ceilings) completed**
- ➔ **Speaker modeling for audience area coverage verification**

SPEAKER PLACEMENT

Once you understand the effects of the architecture, you can start designing your audio system. This often begins with the selection of speakers and their placement. Speaker selection and placement has a big interdependence with room acoustics, so having the right speaker in the right place is key to having an effective and clear sound.

One initial part of the process is determining where there are obstructions that will adversely affect the sound. Whereas the long wavelengths of low frequencies can diffract (or bend) around objects like columns, the shorter wavelengths of high frequencies are blocked by objects in their path, leading to portions of the audio spectrum disappearing for listeners seated behind an obstruction.

For listeners to hear midrange and high frequencies clearly in a theater or auditorium, loudspeakers need to be placed in a straight line-of-sight position to all members of the audience. Naturally, this means that the loudspeakers tend to be visible to the audience and thus have the potential to interfere with the aesthetics of a room's design. However, the visual impact can be minimized by a number of methods, including covering the speakers with an acoustically transparent façade. Any material that is being considered should be tested for sound transmission prior to installation.

Then again, hiding the system is not always best, nor is it popular anymore. Some loudspeakers, such as vertical line arrays and other cleanly rigged systems may be acceptably exposed. In fact, the loudspeaker design and color can be effectively incorporated into the overall room design to complement other elements and overall form. Whether speakers are hidden or exposed, identifying any visibility issues early in the design process can enhance the total ambience of the facility, both visually and acoustically.

PATTERN CONTROL AND LOUDSPEAKER SIZE

Loudspeakers are predominantly directional rather than omnidirectional, meaning the sound is designed to reach a specific coverage pattern rather than projecting sound uniformly in all directions. They focus the sound in one direction, so they can be aimed exactly at the desired coverage area. The benefit is twofold. On the one hand, persons sitting along the axis of the loudspeaker feel they are closer to the loudspeaker. On the other hand, the sound energy is focused on the listeners, so there is less sound energy left for reflections and reverberation buildup. The latter is a particularly important point in locations like houses of worship, where hard walls, high ceilings, and low sound absorption factors lead to critical reverberation times and levels.

Sound system designers choose loudspeakers with coverage patterns that match the listening area. Poor pattern control results in uneven sound coverage, feedback, reflections, tonal irregularities, and poor intelligibility. It typically takes large speakers to control coverage over a wide spectrum, down to a low enough frequency. Loudspeaker engineers generally achieve controlled coverage via horns or loudspeaker driver-spacing interaction. Both require the speaker to be large in order to achieve good pattern control. AVT

VERTICAL LINE ARRAYS become part of the of the visual experience, as well as delivers an outstanding audio experience.

A well-designed system makes the connection between the room, the audio system, and the audience.

Additional Audio Equipment CHOICES

When specifying a mixing console for our "Best Experience" system, the availability of reasonably priced digital consoles makes the most sense. Having in-line dynamics processing available on every input channel, for instance, makes outboard gear unneeded in most cases. Flexible output options allow for a variety of proprietary and other available Ethernet-based audio-over-IP platforms. With the ability to transport audio over a network, the wide distribution of multi-channel audio has been greatly simplified.

Given that many equipment scenarios are possible in a project referred to as a "live sound" space, we won't be able to address many of them on a case-by-case basis, but the knowledge level of these newer generation products will be key to understanding how they work together.

The digital console market has expanded in the last several years to offer either built-in or optional output cards to accommodate many of the current digital formats, including Dante, MADI, AVB, and BLU-Link, among others. Using this digital buss architecture, the majority of the entire signal chain remains in the digital format.

The amplifier choices available have also adopted many of these same digital transport protocols and, along with onboard digital signal processing, create a very efficient method of securely routing signals throughout a facility. The current trend toward multi-channel amplifiers of this variety allows for greater power density within the amplifier rack. AVT

Best Practices

- ➔ **Removing the analog wiring from the main audio distribution chain**
- ➔ **Use multi-channel amplifiers, where possible, to lesson amp rack space and increase power density of the available amplifier room square footage**

Choosing **THE RIGHT MICROPHONES** for the Application

Once you have selected the proper speakers and their placement, you must then determine the types of microphones to use in order to ensure the best intelligibility for the space.

UNIDIRECTIONAL VS. OMNIDIRECTIONAL MICROPHONES

We're moving to microphones next, because microphones are similar to loudspeakers in several ways, except that rather than looking at the direction of outward-going sound, we are looking at the directions of inbound sound. For directionality in microphones, we're talking about unidirectional vs. omnidirectional.

Omnidirectional microphones capture sounds from all directions indiscriminately, including reflections and ambient noise. This means that ambient noise is amplified by the same amount as the wanted sound, making the speaker's words much less intelligible. Even worse, the microphones can pick up the amplified reflections and loudspeaker sounds. This can create a loop, where the amplified sound is picked up by the mic, re-amplified, and then picked up by the mic again, louder this time. The resulting squelch is what is known as feedback.

This is why many microphone applications use unidirectional mics. Using a unidirectional microphone actually provides two main benefits. First, it attenuates (or lessens) sounds from other directions rather than its preferred direction. In other words, when you point the mic at the source of the sound, it will pick up the sound while reducing unwanted ambient noise coming from other directions (such as behind the mic). The second benefit of unidirectional microphones is that they deliver a stronger output signal at the same working distance than its omnidirectional counterparts. This means you get the same loudness of the thing you are wanting to mic at a lower sound system volume setting and with less background noise. Because the sound system is at a lower volume setting with the same loudness, the microphone will pick up less loudspeaker sound

POLAR RESPONSE PATTERN

Characteristic	Omni-directional	Cardioid	Super-cardioid	Hyper-cardioid
Polar response pattern				
Angle of max. rejection (null angle)	—	180°	120°	110°
Rear rejection (relative to front)	0	25 dB	12 dB	6 dB
Ambient sound sensitivity (relative to omni)	100%	33%	27%	25%
Distance factor (relative to omni)	1	1.7	1.9	2

This diagram of a cardioid microphone illustrates in which direction its sensitivity is highest (0 degrees), and in which direction the microphone pattern is attenuated (180 degrees).

and there will be less sound energy in the space for reflections. All this adds up to a much better intelligibility. In other words, using unidirectional microphones is a good way to avoid creating a feedback loop.

UNDERSTANDING POLAR RESPONSE

As we said previously, unidirectional microphones appear to sit closer to the talker than they actually do. They prefer the wanted signal (the talker's voice) coming from one direction and attenuate unwanted noise from around the talker. (This is similar to the effect of placing a hand behind your ear: sounds from certain directions are attenuated, while sounds from other directions seem louder). This minimizes the risk of feedback and improves intelligibility.

However, there are different types of unidirectional microphones, identified by their polar response. The polar response of a microphone determines how much sound the microphone picks up 360 degrees around the microphone. The two most common types of polar response for unidirectional microphones is cardioid and hypercardioid. There is also supercardioid, figure eight, and halfcardioid, among others.

The polar diagram of a cardioid microphone illustrates in which direction its sensitivity is highest (0 degrees) and in which direction the microphone pattern is attenuated (180 degrees). Cardioid microphones should therefore be your first choice in situations with a noise source or loudspeaker directly opposite the talker.

The hypercardioid provides its highest attenuation at approx. 125 degrees off-axis, so it is very efficient at rejecting reflections from the ground or tabletop. The hypercardioid and "shotgun" are the polar patterns with the highest off-axis rejection. Compared to an omnidirectional microphone, a hypercardioid picks up four times more wanted sound than ambient noise, so these microphones should be your preferred choice for spaces with high noise levels.

The overall performance of a sound system often depends on the polar patterns of the microphones used. Real-life polar patterns are not the same at all frequencies. The directivity is usually higher at mid frequencies than it is at the low end. This has its implications for designing a system, because most ambient noise is in the low frequency band, so using a high pass filter on the input channel is strongly recommended.

DYNAMIC VS. CONDENSER MICROPHONES

Microphones are what are known as electro-acoustic transducers. These transducers convert variations in air pressure (sound waves) into electrical signals. There are two main types of transducers (and thus, two main types of microphones): dynamic and condenser. Condenser microphones have a higher sensitivity, but need a polarization voltage to work. In most cases, this is fed from the mixer to the microphone via the cable, using a technique called "phantom powering." Because of their higher sensitivity, they sound louder than dynamic microphones at the same gain setting. The higher a microphone's sensitivity, the less you have to amplify the signal, and the less background hiss you will get. This is why condenser microphones are often used for quiet sound sources. The condenser microphones are also usually much smaller and lighter than dynamic designs, which is why most head-worn and lavalier microphones are (electret) condenser types. Because of their small dimensions, condenser transducers are also used in gooseneck microphones.

However, dynamic microphones are much more mechanically rugged than condenser transducers. As such, microphones designed for high mechanical stress and high sound pressure levels generally use dynamic transducers, which is why most dynamic mics are handheld microphones. Dynamic microphones do not require phantom powering, but are less sensitive than condenser mics.

TYPES OF MICROPHONES

Gooseneck Microphones

Besides having different technical and sonic properties, microphones come in different sizes and shapes. Gooseneck microphones are particularly suited for permanent installation (e.g., on a lectern), for it is obvious into which end the user has to talk, and the microphone is usually close enough to the talker's mouth to ensure good intelligibility. Choosing a microphone with a polar pattern optimally suited for the application will further improve intelligibility and gain before feedback.

Gooseneck microphones are available in various lengths and can be bent so that the talker will stand or sit in front of the microphone in a comfortable position and at the ideal distance. They are available in various colors to blend in with the interior decoration. Various types of installation accessories allow you to minimize mechanical noise (e.g., if a microphone is mounted on a piece of furniture).

Boundary Microphones

In some cases, no microphones must be seen. In these halls, boundary microphones can be an alternative to gooseneck types, as they can be mounted almost invisibly, for instance, on a lectern, or in a tabletop. Boundary microphones pick up all signals arriving from above the boundary (in this example, the tabletop). In other words, they have a pickup angle of about 180 degrees. When installing a boundary microphone, make sure to install it in a surface as large as the wavelength of the lower frequency limit.

Boundary microphones differ from gooseneck mics in that they are more sensitive to unwanted noise from the tabletop (users knocking on it, shuffling papers, etc.) and place the transducer further away from the talker. Also, many users put papers on top of the boundary microphone, so it cannot function properly anymore. In areas near loudspeakers, boundary microphones may not be the best solution, because they increase the risk of feedback. Sound systems with boundary microphones need to be designed extra carefully, and it may be a good idea to use special designs with a preferred pickup direction.

Overhead Microphones

Even nearly invisible boundary microphones may sometimes be perceived as too intrusive, or it may be impossible for architectural reasons to run cables to the lectern or pulpit. In such cases, you can use special overhead microphones suspended from the ceiling or boundary microphones flush-mounted in the ceiling.

Note, however, that this solution has inherent acoustical drawbacks. Loudspeakers are often installed in the walls or ceiling and many of them may be closer to the microphone than the talker. Such a system would wreak havoc on the sound in terms of gain before feedback and intelligibility.

Lavalier and Head-Worn Microphones

For talkers roaming around, the best solution is to use a wireless microphone. These microphones use a radio transmitter and receiver instead of a cable. Handheld radio microphones have a built-in transmitter, which comes in handy if a single microphone needs to be passed from talker to talker. Head-worn and lavalier microphones are connected to a separate "body-pack" transmitter.

The closer a microphone sits to the talker's mouth, the higher is the system's gain-before-feedback. Head-worn microphones also minimize handling noise and maintain a constant output level as the distance between the microphone and sound source hardly ever varies.

Microphones for Musical Instruments

Just like musical instruments differ widely in size, shape, and range, a wide variety of microphones for picking up their sounds can be found on the market today. Some microphones were specifically developed for certain instruments, such as the D112 microphone for capturing low-frequency sounds inside kick drums or in front of bass amplifiers. On a different note, many clip-on microphones were designed to look good on specific instruments.

Still other microphones are suited for a range of different applications. Some large-diaphragm condenser microphones, originally designed for studio use, are also excellent for miking up instruments in live sound situations. Large-diaphragm microphones use transducers with a diameter of 1-inch or larger. The sensitivity of a condenser microphone being directly proportional to the surface area of its transducer, these microphones are more sensitive and less noisy than condenser microphones with smaller diaphragms.

Vocal Microphones

Vocal microphones can be either dynamic or condenser microphones. Which type you will actually use is a matter of taste and depends on the cantor's style and timbre. As mentioned, dynamic microphones are usually more mechanically rugged, so less experienced users are less likely to overload the microphone.

It's important to use a microphone designed for voice. The essential information of a speech signal is found in a range from 300 Hz to 3.2 kHz. Most microphones have a flat frequency response within this band. Consonants, which are essential for keeping a speech signal intelligible, are sounds at frequencies between 2.5 kHz and 15 kHz. Therefore, speech-optimized microphones boost these frequencies, whereas applications such as instrument miking, require a flat frequency response.

To reduce the risk of feedback, use a unidirectional microphone, i.e., a cardioid, hypercardioid, or supercardioid. Microphones of this type are available in both hardwire and wireless versions. Which version to use will depend primarily on technical and visual considerations. If the cable does not intrude visually and no staff is available to operate the sound system, always use a hardwire microphone.

IMPROVING INTELLIGIBILITY

Microphones are the ears of your sound system. The more ears there are, the more they will hear, and unfortunately, this goes for unwanted reverberation, too. Therefore, to optimize intelligibility and avoid feedback, as few microphones as possible should be open at any time, at best only the one that is being talked into at the moment. If we reduce the number of open microphones, say, from four to two, the proportion of ambient noise and loudspeaker sound in the master output signal of the mixing desk or automixer decreases by 3 dB.

In most cases, there is no sound engineer available to operate the mixing desk. Therefore, automatic microphone mixers provide a convenient solution for managing open microphone channels, as they automatically activate only as many microphones as actually needed. This also dramatically reduces the risk of feedback in any room. So, no matter if your system will use a manual or automatic mixer, always remember to keep the number of open microphones as low as possible.

In most applications other than recording studios, microphones are intended to pick up only that part of the sound spectrum that is relevant for speech and to ensure maximum intelligibility. In other words, they should only capture the talker's voice and reject reverberation (even though it is audible, too) as well as any frequencies, particularly bass sounds that may reduce intelligibility. But beware of cutting out the low frequencies too radically by using an equalizer, narrow-band loudspeakers, and extremely speech-optimized microphones: your system may end up sounding like a telephone receiver! Select components that allow a good sound engineer to achieve a well-balanced, and inspiring sound. AVT

Best Practices

- ➔ **Correctly match the type of microphone to the sound source. Not sure? Consult with the microphone manufacturer for technical assistance.**
- ➔ **If the end user has untrained personnel, or timid vocalists, be sure to schedule a training session after the system's installation that includes proper microphone technique and use.**

ACOUSTICAL Treatment, and Replacement Systems

Defining the goals of a space relative to the target reverberation times for performance, lecture hall, or even a House of Worship has become more challenging because many live sound environments are being used for multiple; and sometimes very different purposes.

Best Practices

- ➔ **Utilize Electro-acoustic modeling programs when possible for loudspeaker design confirmation (Ease, Catt Acoustic)**
- ➔ **Retain the services of an acoustical consultant if the project poses any level of potential disaster!**

CASE STUDY

Using a 1700-seat performing arts center for moderate-intensity pop artists seems like a natural fit, except the room was designed primarily as a concert hall for orchestral use. Even the inclusion of a variable acoustic system does not support the occasional current use of a modern touring system as well as it should. The new and future direction will have more pop/rock use as well as the visiting orchestra.

Question 1: Is it correct to modify the space with additional treatment to achieve a 1.4s reverb time? How would this be achieved? The answer probably lies in the addition of suspended panels or banners that do not interfere with the installed surfaces and can also be retracted after use. The issue is obtaining funding for this addition.

Question 2: Had this higher percentage of mixed use been better projected at the time of design could this have been avoided? Quite possibly!

REFLECTED ENERGY VS. DIRECT SOUND: WHEN ARE REFLECTIONS NOT AS PROBLEMATIC?

The use of the earlier mentioned, and increasingly popular, digitally steered column loudspeakers have an attached question that is commonly asked, "Does the wide horizontal pattern cause problematic reflections into narrower audience areas?" The answer depends on several factors.

If the placement of the columns is directly against the side wall surface, no. There is a reflected path that would theoretically cause cancellation at a certain frequency, depending on this distance, but given these are primarily used for speech purposes, this is mostly unnoticed. When the columns are placed away from the sidewall boundary surface, the first order reflection that has a close timbre response to the main propagation wave tends to integrate and be a non-issue. The energy that is reflected is also contained in a very narrow vertical pattern which, assuming a relatively flat vertical sidewall surface, does not excite the reverberant field as would a standard enclosure with a wider vertical pattern.

CREATING A RECALLABLE ACOUSTIC ENVIRONMENT

In reference to target reverberation times, mentioned above, the need to have a greater ability to accommodate a wider range of events can be addressed through the use of an Acoustic Replacement System. With this type of system, smaller loudspeakers are installed throughout the venue, and in conjunction with the associated processing, recreate the acoustic characteristics of a wide variety of rooms and halls. Early reflection information can also be managed separately and added for 'voice lift' to assist with intelligibility throughout the room. AVT

Now, Create THE BEST EXPERIENCE

It's true that professionals must consider many factors in order to make audio within a space successful. However, with the proper planning and the right equipment, a better experience for facility owners, operators, managers, and their guests can be assured to be a **Best Experience**.

Additional Resources

Click on live links