How to Configure Your Church PA System

In our companion article, "*How to Mix and Record Church Services—and More*," we cover the critical nature of choosing the right mixer for your house of worship. In this article, we will discuss selecting and configuring loudspeakers for your congregation's needs.

Next to your mixer, your loudspeaker system is the most important part of your PA system.

Investing in a high-quality loudspeaker will make mixing easier and provide a more enjoyable listening experience for your congregation.

Selecting the best loudspeaker for your service

When a high-quality loudspeaker accurately reproduces audio, the speaker is sonically transparent, so that the audio simply fills the room. You hear the audio content, not the speaker.

When selecting a loudspeaker system for your sanctuary, there are a few things to consider:

- **Sanctuary size.** The size of your sanctuary will largely determine the size, wattage, and number of speakers that you will need to purchase. If your sanctuary is wider than it is deep, you may need to install one or two center-fill speakers in addition to the loudspeakers on either side of the pulpit. If the sanctuary is long and narrow, you may need to install delay systems to evenly distribute audio throughout the pews. (We'll discuss center-fills and delay systems shortly.)
- **Service needs.** The type of service you offer will also guide the type of loudspeaker system you need. If your praise team consists of a full band with a choir on the first Sunday of the month, you will need a much larger system than if your services utilize acoustic instruments like piano and guitar with one or two vocalists.
- **Budget.** When upgrading or investing in a new loudspeaker system, always purchase the best system available for the budget you have. Be forward thinking: Purchase a system that fits your congregation today with an eye toward how it will serve your congregation as it grows.

Speaker Basics

Let's take a moment to define a few common loudspeaker terms. These will help you to better understand what types of loudspeakers are available and how they will perform.

Driver. Drivers (also called transducers) are the part of the speaker that produce sound waves. Drivers come in all different sizes, but in general, the bigger the driver is, the lower the frequencies it will reproduce.

Some loudspeakers, such as the StudioLive 328AI, rely on two smaller drivers to produce the same low frequencies as one larger driver. Smaller drivers tend to be tighter than larger drivers so they are more responsive and can more accurately reproduce sound.

StudioLive 328AI

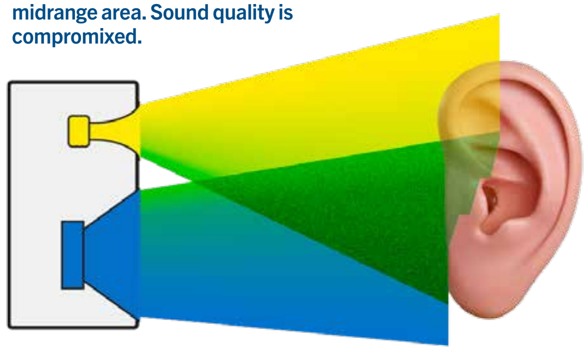
Larger drivers have more surface area and can move more air, creating more sound waves, resulting in more volume.

A dual-eight-inch configuration essentially doubles the surface area a single eight-inch driver provides. While this is still one-third the surface area of a single 15-inch driver, a dual-eight configuration makes up for the loss of volume and delivers a higher fidelity performance than a 15-inch driver, although it can't get quite as low.

Because of this, the StudioLive 328AI is ideal for situations that require high intelligibility and fidelity, but not as much low end. This makes them great as a center fill or for services that rely more on acoustic instruments and vocals.

Two-way versus three-way. In a two-way loudspeaker, the high- and low-frequency elements share the responsibility for mid-frequency repro-

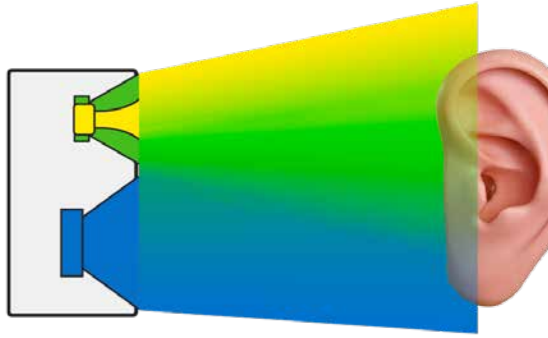
A two-way speaker doesn't have a dedicated driver for the critical midrange area. Sound quality is compromised.



duction. A three-way system provides a separate driver that is dedicated to mid-frequency reproduction.

Because a two-way system relies on a high-frequency driver and a low-frequency driver to cover a frequency range that they're not optimally suited to produce, there are sonic compromises in the midrange—and the midrange is pretty much where everything you need to hear "lives." The human voice, guitars, most instrument attacks, and snare drums are all examples of the important audio content that resides in the midrange.

In contrast, three-way loudspeak-



A three-way speaker has a dedicated driver for the critical midrange area, vastly improving sound quality.

ers, including all StudioLive AI-series full-range loudspeakers, offer a dedicated midrange driver that's just the right size to accurately reproduce midrange frequencies. This benefits the entire audio spectrum, not just the midrange, because the high- and low-frequency drivers are free to reproduce the frequencies they are best suited for.

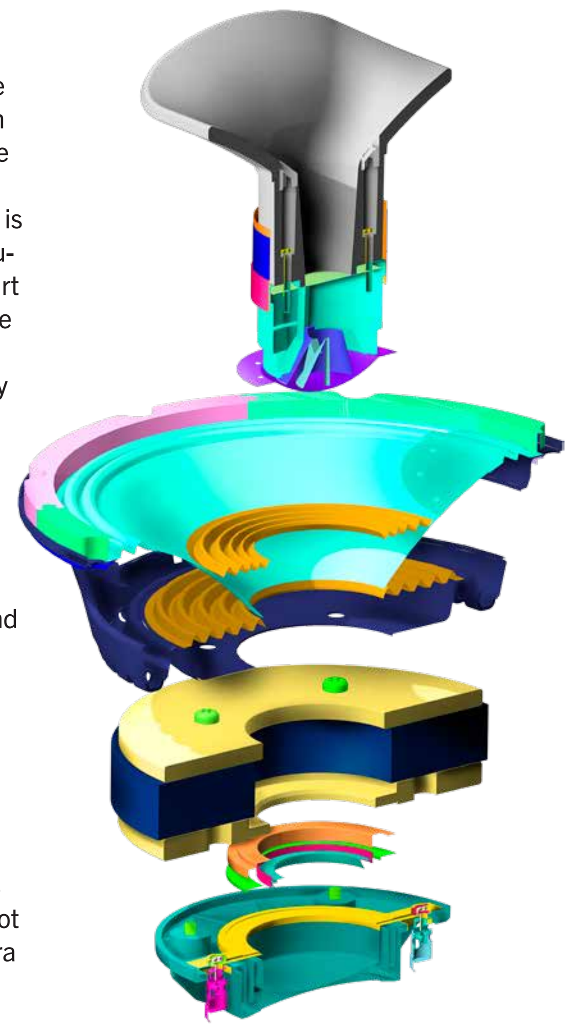
The net result is that the StudioLive AI series reproduces clean, clear, accurate audio across the audio spectrum.

Coaxial speaker. A coaxial speaker places the high-frequency driver in the center of, and on the same axis as, the low-frequency driver, which is similar to the way the human ear works. The heart of each of the StudioLive AI-series loudspeakers is PreSonus' proprietary CoActual™ design. This 8-inch coaxial speaker is responsible for the high- and mid-frequency reproduction for all three full-range models, and it provides amazing intelligibility, even at high SPLs, delivering clean, natural sound without distortion.

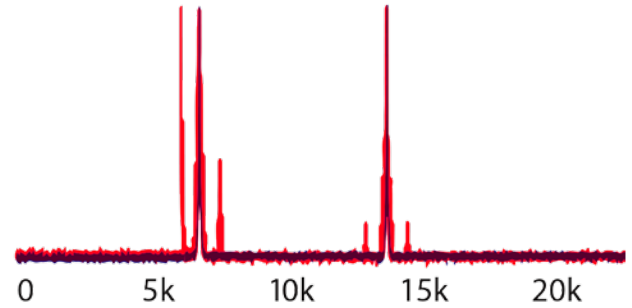
This is because the CoActual design can reproduce transients with very little change in the shape of the waveform. Consonant sounds in speech, percussion, and the attack and decay portions of musical instrument sounds are all examples of transients. When a waveform is not accurately reproduced, you get extra impulses that blur and distort the transient. This is called "intermodulation distortion," and it becomes increasingly uncomfortable at high volumes because artifacts that aren't supposed to be present blur the audio signal. This discomfort can increase the perceived loudness of a speaker.

Good news: By maintaining accurate transient reproduction, even at high volumes, CoActual speakers minimize distortion and produce clean, clear, consistent audio at any SPL. This means that even when the praise band is at its loudest, your congregants will not be distracted by uncomfortable distortion and can let the spirit move them.

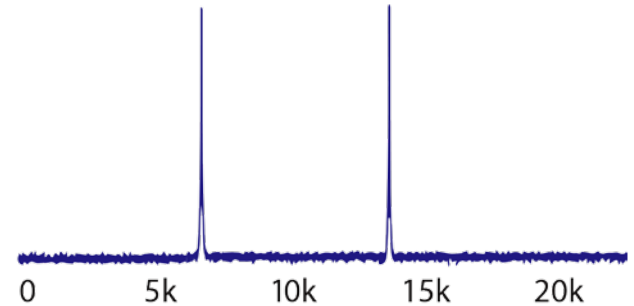
PreSonus' unique CoActual™ transducer superimposes treble and midrange drivers on the same axis, improving sound quality.



How to Configure Your Church PA System



The extra red spikes are *intermodulation distortion*. They cause a "smearing" of the sound impulse.



Free from *intermodulation distortion*., the sound impulse is clear and well-defined.

Active. Active loudspeakers have onboard power amplification that has been designed to optimally power the drivers. This provides a huge advantage in that it takes the guesswork out of choosing the right speaker/amp combination, because engineering teams have ensured an ideal match. The disadvantage is that active loudspeakers are heavier and will require AC power at their installed location. In general, the benefits of active loudspeakers like the StudioLive AI-series far outweigh the additional, well, weight. Every component has been carefully selected to complement the others.

Passive. Passive loudspeakers do not have an onboard amp and require an external power amplifier. The advantage of the passive loudspeaker system is that you only need to run speaker cable to the install location.

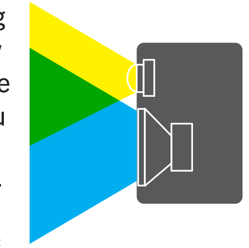
Passive loudspeakers often seem like a more affordable solution but they only sound as good as the amplifier that powers them.

If you cut costs by purchasing a lower-wattage power amplifier, you will shorten the life of your loudspeakers because underpowering them is as bad as overpowering them. Furthermore, even if the power level is sufficient, separate amplifiers and speakers combine in ways that can be difficult to predict; some speakers simply sound better with certain amplifiers. With active loudspeakers, this is not an issue, since the amplifier has been selected specifically for that speaker.

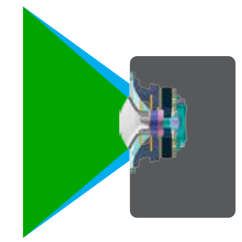
Crossover. A crossover separates the frequencies coming into a speaker and distributes them appropriately to the woofer and tweeter. This helps the speaker run more efficiently and reproduce the frequency spectrum more reliably. The crossover transition is the point in the audio spectrum where one driver stops and another takes over.

Coaxial speakers, like the one used in the StudioLive AI-series loudspeakers, offer a seamless crossover transition because of their symmetrical response both horizontally and vertically. This means a wider "sweet spot" that is more consistent throughout the sanctuary.

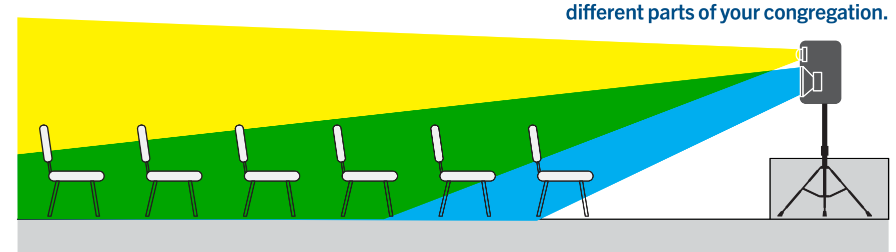
Conventional two-way designs that place the high-frequency driver directly above the low-frequency driver can't provide this symmetry. With conventional designs, where you are listening is critical to how well you hear the audio. While you are in the sweet spot, the speakers can sound great, but out of the sweet spot (shown in green), you will lose frequencies and imaging. This means that the quality of the audio will change depending on where you are sitting in the sanctuary.



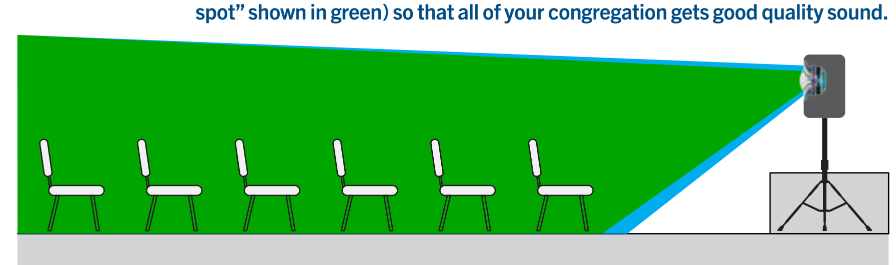
Good news: The PreSonus 8-inch CoActual design offers a symmetric response, so it provides a wide "sweet spot" that is consistent throughout the room. With no gap or overlap in the crossover transition between the high- and mid-frequency ranges, the sound affords a more pleasing, natural listening experience. This also makes mixing easier for church volunteers because they don't have to rely on equalizers to bring back audio information that was lost in the crossover transition of their front-of-house system.



Poor crossover transition inherent in 2-way systems means sound quality varies for different parts of your congregation.



Our 3-way CoActual design optimized crossover transition (and increases the "sweet spot" shown in green) so that all of your congregation gets good quality sound.



Bi / tri / quadamplification. This term relates to active loudspeakers .and refers to the number of amplifiers inside. For example, a biamped loudspeaker has two amplifiers inside: one dedicated to powering the high-frequency driver and the other dedicated to

powering the low-frequency driver. By separating the frequencies before they hit the amplifiers, a biamped system removes one of the major sources of intermodulation distortion. The resulting sound is more open, clear, and less fatiguing.

StudioLive AI-series loudspeakers provide one 500W amplifier for the high-frequency driver, another 500W amplifier for the mid-frequency driver, and two 500W amplifiers for the low-frequency driver. The StudioLive 328AI, with its dual eight-inch woofer, uses these two 500W amplifiers independently: one for each woofer. The two 500W amplifiers in the StudioLive 312AI and 315AI are bridged, delivering a total of 1,000 watts of power to the 12- or 15-inch woofer, respectively.

DSP. DSP is an abbreviation for "Digital Signal Processor." A DSP used in audio products is basically a computer that handles digital audio information in some fashion. Many modern loudspeakers provide some type of onboard DSP. Most use it to correct speaker anomalies ***after*** the speaker has been built.

PreSonus StudioLive AI-series loudspeakers use a different approach. StudioLive AI-series loudspeakers were designed from the ground up with the knowledge that they were going to be equipped with a powerful DSP. So when our engineers came across two competing benefits in the physical design – meaning that if they went with one benefit they would lose the other – they solved one in the physical design and the other with DSP.

This is the basis for Temporal Equalization, or TQ™, a patented technology that PreSonus licensed from its inventor, Fulcrum Acoustic. TQ relies on a special type of filter called "Finite Impulse Response," or FIR. FIR filters' ability to make more detailed frequency-response and time-domain adjustments is responsible for TQ's famed benefits: crisper stereo image, greater soundstage depth, more separation between components in a complex mix, increased resistance to feedback, and a less fatiguing listening experience at very high SPLs.

temporal equalization tq

Good news: StudioLive AI-series loudspeakers are easier to mix on because every audio source can be easily heard by the operator. This is especially useful when training novice volunteers. The pristine stereo imaging and three-dimensional soundstage make the listening experience more natural for the congregation because the audio can easily be made to sound like it's coming from the right spot.

The human brain is designed to hear in three dimensions, and when a speaker can deliver that experience, it "feels" like the pastor's voice and the praise band's instruments are projecting naturally, without any help from sound reinforcement. Thus, StudioLive AI loudspeakers sound transparent, which is the ultimate goal of any properly designed loudspeaker.

Recognizing Problem Rooms

A PA system consists of two main components: the mixer and the loudspeaker system. How well they perform together is very much influenced by the acoustics of the space in which they operate.

Difficult room acoustics, combined with improper loudspeaker placement, can interfere with achieving the fidelity of which your loudspeakers are capable and force you to use the equalization on your mixer to correct problems rather than using it to enhance the sound quality of your mix.

The goal of proper speaker placement and room tuning is to provide a blank slate for the mix engineer. The more time that is spent on setting up the loudspeaker system properly, the less time even the most experienced volunteer will need to spend dialing in a good mix.

Church sanctuaries, like most live-sound environments, are rarely designed to maximize the listening experience. More often then not, money is spent on aesthetic appeal rather than acoustic treatment.

While your sanctuary may be a great environment for prayer and contemplation, it's necessary to recognize and correct what that space does to the sound system in order to optimize the PA's performance in the room.

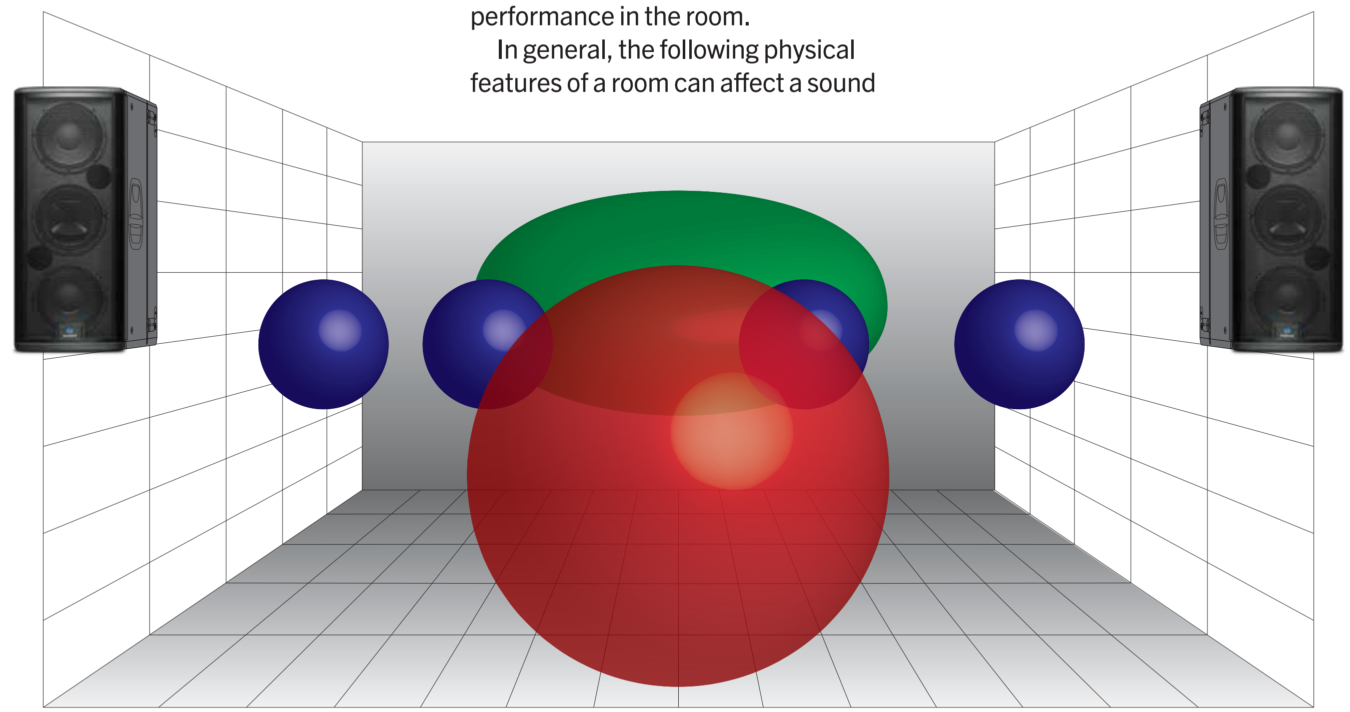
In general, the following physical features of a room can affect a sound system's performance:

- Room size
- Construction
- Reflectivity

The size of the sanctuary (or any other room) directly impacts how well certain frequencies will be reproduced. For example, if you measure the room diagonally, you will discover how well that room will be able to sustain low frequencies. This may seem odd until you think about the physical size of audio waves at various frequencies.

For example, a 50 Hz wave is about 22.6 feet long. (To calculate how big an audio wave is, divide the speed of sound—1,130 ft./second—by the frequency. For a 50 Hz wave, 1,130/50 = 22.6 ft.) So a sanctuary that is 45 feet on the diagonal is going to regenerate low frequencies more effectively than one that is 15 feet on the diagonal.

When a room's width or length correlates directly to the size of a waveform at a specific frequency, a standing wave can occur where the initial sound and the reflected sound begin to reinforce each other. Let's say we have a long, narrow sanctuary



Poor crossover transition inherent in 2-way systems means sound quality varies for different parts of your congregation.

where the distance from one side to the other is 22.6 feet. When a 50 Hz wave bounces off the wall, the reflective wave travels right back along the same path and bounces off the other wall, and the cycle repeats. In our example sanctuary, 50 Hz reproduces very well. So any mix will have a heavier low end. Whether or not that is a problem depends on the style of service you offer and the type of loudspeakers you are installing. In this example, it would be a point to keep in mind when configuring and installing a PA system.

Another way low-frequency waves can be impacted by your sanctuary is in its construction. Low-frequency waves are powerful enough to cause the walls, ceiling, and even the floor to flex and move. This is called "diaphragmatic action," and it dissipates energy and strips away the low-end definition.

So if you're in a newer church in which the walls and floor are made of thick concrete that don't vibrate much, the bass response is going to be much more powerful than if your church is older and was built before the invention of amplification.

Older churches were designed so that congregants could easily hear the pastor and the choir, so they reproduce mid and high-range frequencies much better. In addition, an older church typically has a long natural reverberation and will need some acoustic treatment to better facilitate the use of a modern PA system.

Good news: StudioLive AI-series mixers' included VSL-AI control software provides an easy-to-use Room Analysis Wizard that shows you exactly what your room is doing to the audio spectrum. Once you have generated a frequency-response trace, you can either use the StudioLive AI mixer's EQ to correct these issues, or use free PreSonus SL Room Control speaker-management software and the 8-band parametric EQ onboard the StudioLive AI-series loudspeakers to free up your mixer's EQ for aesthetics.

PreSonus' PRM1 Measurment microphone, used with the StudioLive AI mixer's VSL Smaart Room Analysis Wizard

The Smaart Room Analysis Wizard and a PRM1 measurement microphone let you smooth out the frequency response of a sanctuary. In real time, you see the current frequency distribution (white line). Then you can "pull" the Low, Low Mid, High Mid and High control points (colored circles) to create an equalization curve (blue line) that compensates for room acoustic problems. .

Maintain a High Direct-to-Reverberant Ratio

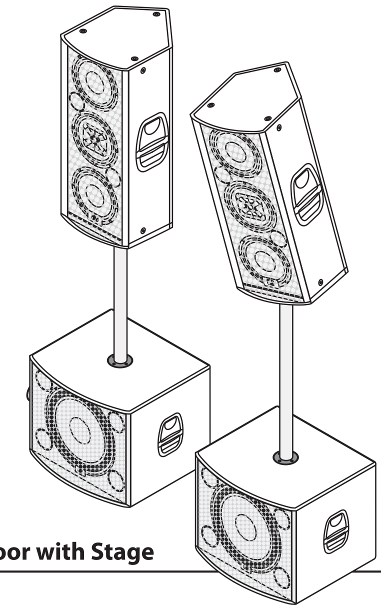
How well your sanctuary reproduces sound naturally will also determine how powerful a PA system you need. The less reverberant a room is, the more help your pastor and praise band are going to need from a loudspeaker system. The better the room amplifies sound, the smaller the system you should install to retain direct signal clarity.

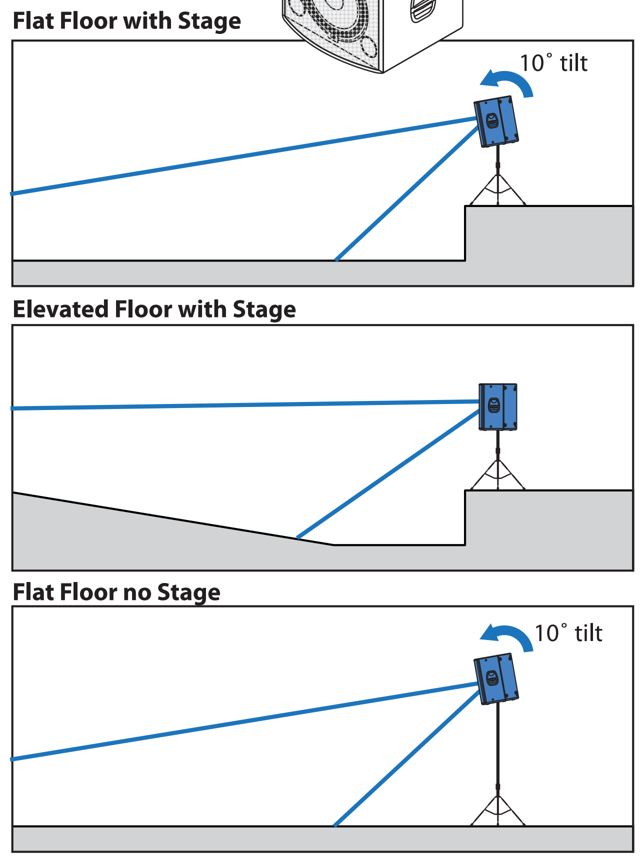
Like most room anomalies, reverberations can be good and bad. Consider the effect of a cathedral's reflections on a choir or a piano. This type of reverberation (reverb) is quite desirable. But not all reverb is good reverb. Reflections can also cause comb filtering. For example, if a speaker is placed near a reflective surface (such as a stained glass window), the direct sound coming from the speaker and the reflected sound coming from the window can arrive at the listener's ears out of phase with each other, causing cancellation and reinforcement. If they're 180 degrees out of phase with respect to each other, they will cancel each other out.

If you are using your loudspeakers in a reverberant environment, position them so that as much sound as possible is focused on the congregation area and steered away from reflective surfaces. When you do your placement and positioning, it's a good idea take some time to do a "walk around" of your loudspeakers, playing either pink noise or program material, so you get a feel for how the sounds translate in the room.

Vertical Coverage

It is important to keep in mind that vertical coverage is just as important as horizontal coverage. If you are using a ground-stack approach with pole mounts, make sure your coverage matches the listening plane. Suspension of speakers will provide even further control.





Important note: Always use a licensed and insured system integrator when suspending loudspeakers. These individuals are not only trained to know how to safely suspend heavy loudspeakers, they also know how to hang them so they sound their best.

Some loudspeakers, such as the StudioLive 312AI and 328AI, feature dual-position pole mounts that allow you to mount the speaker atop a stand at 90 degrees or at a 7- to 10-degree downward tilt. (The StudioLive 315AI does not offer a downward tilt.) Using the downward-tilt mount will focus the loudspeaker's energy onto the congregation and avoid destructive reflections. This is ideal for situations where the loudspeaker is mounted atop a tripod stand and placed on a stage or where the pole-mounted loudspeaker is on the floor and the coverage area is relatively shallow, as with a small youth room.

Wall and Corner Loading

Very low frequencies are not directional, so they radiate out of the sides and back of the loudspeaker, as well as out of the front. If you place a loudspeaker against a wall, the rear sound propagates back into the room. This can increase output of bass frequencies as much as 6 dB, and as much as 12 dB if you put the loudspeaker in a corner.

In order to have the most control over your sound, it's best to always start with the flattest response, so you normally should avoid wall and corner placement. On the other hand, if you need some extra bass boost, this technique may be worth a try. It is important to be aware of what's happening and be prepared to take advantage of it or compensate for it.

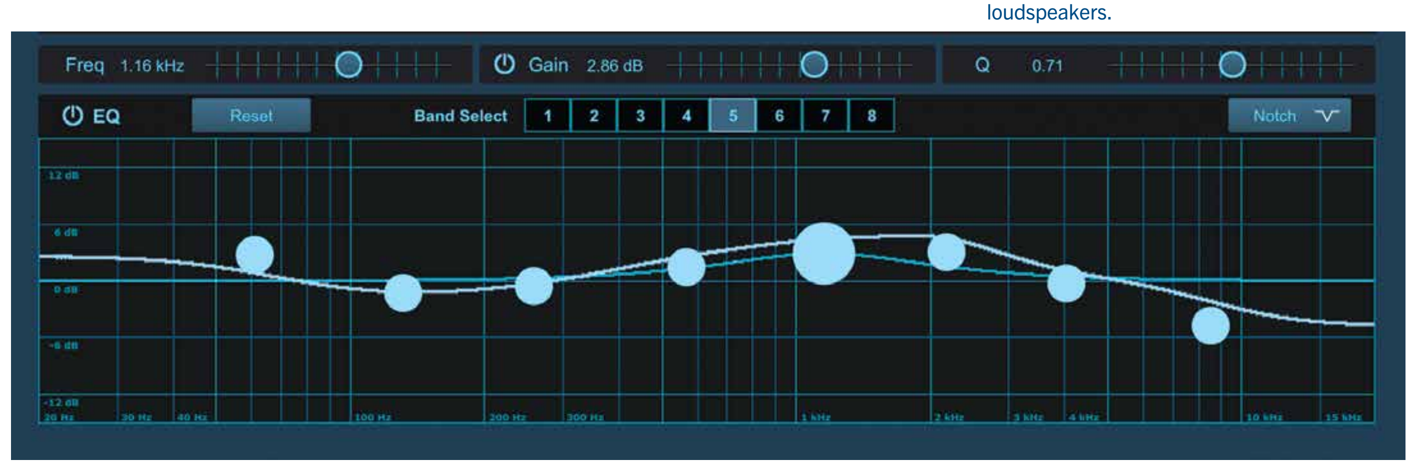
Good news: Each StudioLive AI-series loudspeaker provides an 8-band parametric EQ, accessible via SL Room Control. In the event that wall or corner loading can't be avoided, you can use this EQ to flatten out some of the bass energy.

Delay Systems

In most situations, a PA system relies on two main speaker systems positioned at the front of the room to reproduce audio for the entire space. As a result, the level of the system is considerably louder in the front rows of pews then it is at mix position. Unfortunately, the congregants sitting in front are generally the most sensitive to loud volumes.

With a point-source, horn-loaded loudspeaker (such as a single, powered StudioLive AI loudspeaker), sound intensity is lost at a rate of

8-band parametric EQ, part of SL Room Control software,free with StudioLive AI loudspeakers.



-6 dB per doubling of distance. This is true regardless of tuning, amplification, power rating, or any other speaker specifications. So if your signal level is 106 dB SPL at 1 foot, at 8 feet away it's down by 18 dB!

Here's a simple chart that illustrates the math:

DISTANCE	dB SPL
1 foot	106 dB
2 feet	100 dB
4 feet	94 dB
8 feet	88 dB
16 feet	82 dB
32 feet	76 dB

In situations where sound must be reproduced outside of the main system's optimum range, well-placed delay systems offer support by extending the intelligible range of the PA. Rather than relying on a pair of front-of-house speakers to fill the entire room, you can create listening zones throughout the sanctuary so that your front-of-house system only needs to be loud enough to cover the front of the room. This allows you to lower the level, letting congregants in the front pews listen at a comfortable level and getting better fidelity from your speakers. This type of configuration can be beneficial in just about any midsized to large room, as it will keep the overall level of the system at a much lower volume.

However, it's not as easy as just installing an extra pair of speakers. Since electricity travels much faster than sound, listeners in the rear of the sanctuary are likely to hear the sound coming from the nearest set of speakers before they hear the sound from stage, which can dampen the attack and intelligibility of the sound and create an unpleasant phasing effect.

To compensate, you need to delay the signal going to the additional speakers. For example, it takes about 55 ms for sound to travel 50 feet. So if you put your speakers 50 feet back,

When used with our PRM1 Measurement Microphone, the Smaart Delay Wizard automatically sets the correct delay time — a task that used to require an experienced, professiona sound engineer.

you need to delay the signal by 55 ms.

Good news: StudioLive AI-series mixers and VSL-AI provide an easy-to-use System Delay Wizard that calculates the exact delay time needed for you. Once it has been calculated, you can either connect your delay system to your mixer's subgroup outputs and use the StudioLive AI mixer's subgroup delay, or, better yet, use the 500 ms delay onboard each StudioLive AI-series loudspeaker, controlled with SL Room Control, and run your entire system off the Main outputs.

If neither your mixer nor your loudspeaker has a built-in delay, use a separate delay processor to achieve the same results.

Delay Basics

Delay speakers allow you to run the main speakers at a lower volume, as they relieve the mains of handling high- and mid-frequency content for part of the space. As a speaker is pushed harder, the edges of its frequency response begin to distort, so by easing the demands on the mains, delay systems increase fidelity sonically, as well as mechanically. This also means that the front row doesn't need to be blasted just so the people at the back can hear the praise band.

The goal of distributed sound is to extend the intelligible range of the system, without killing the front of the crowd with excessive level. As noted earlier, sound travels much slower than electricity, so the audio coming out of the delay system will arrive to the listeners before the audio coming out of the main system.

Without proper alignment, the multiple arrival times create confusion to the listener and sonic definition is lost. Speech and beat transients become less intelligible. In large rooms, this can actually create a flam or echo effect. By delaying the audio going to side and rear fills, you can create a cohesive listening environment for the entire congregation.

It should be noted that frequencies in the sub-bass range of a delay system do not require distribution. In fact, a delay system's highpass filter should be rolled up as high as 300 to 400 Hz to avoid sound going back toward the stage as low frequencies become omnidirectional.

Good news: Use the onboard 8-band parametric EQ on StudioLive AI-series loudspeakers to roll off low frequencies as needed. Because StudioLive loudspeakers are wirelessly remote controlled by SL Room Control, you can experiment with the best roll off frequency for your sanctuary and your system.

When placing delay systems, the main goal is to maintain intelligibility of the PA, especially in the vocal consonant range (2 to 4 kHz). However, this goal is achieved by overcoming different obstacles depending on whether you are indoors or outdoors. In both situations, the delay system should be set where the main system's intelligibility falls apart. As with the main system, the placement of the delay systems will determine how successfully you are able to achieve these goals.

Inside. Indoors, you are trying to overcome the direct-to-reverberant reflections. The location of the delay system is dependent on the critical listening area (typically just behind front-of-house). Your goal is to find where the direct signal-to-reverberation ratio has reached about 50/50. At this point, the reflections in the room are at an equal level to the direct sound of the PA, and vocal intelligibility is lost. Listen for a lack of intelligibility in the praise band's vocals and find the point at which the drums and rhythm section don't feel tight.

A great way to find the best position for your delay speakers is to set up and tune your main system and play audio through it. Play something similar to what you will be mixing later. Set the level so that it is comfortable from the front row. Walk backward away from the main system until you notice a lack of clarity. This is the beginning of the space that will need delay-system coverage.

Outside. Outdoors, you are trying to maintain level as the noise floor of the crowd begins to be at equal level to the PA in the intelligibility range. When working outside, the delay system is used to overcome outdoor noise, including (but not limited to) crowd murmur, generators, babies, etc. At this point, the main system needs more support in order to deliver the same perceived loudness as you get further from the source.

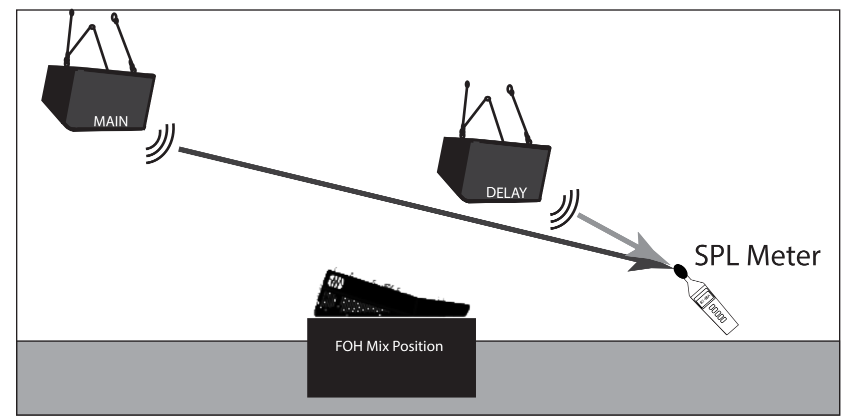
Once you have positioned and delayed your satellite system, use an SPL meter to match the output of the main and delay systems at the measurement point. If you are standing 20 feet from the left side of the main system and 30 feet from the left side of the delay system, and if the output of the main system is 85 dB, the output of the delay system should also be 85 dB.

Your goal is to create a seamless transition from your front-of-house system to your satellite system. When done properly, you should not be able to hear where your main system leaves off and your satellites take over.

Sub Alignment

Delaying subwoofers relative to their full-range counterparts compensates for the cancellation or reinforcement of low frequencies that occurs when the same frequencies are reproduced by two sound sources set some distance apart. Low frequencies in the crossover region between full-range and subwoofer have wavelengths that are several feet long—the wavelength of a 150 Hz wave is about 7.5 feet—which means that reinforcement and cancellation will occur as the waves interact in the room.

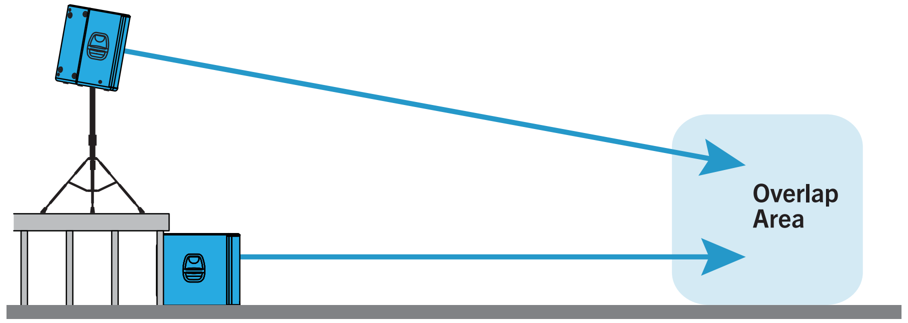
Delaying a subwoofer will compensate for this effect when the loudspeaker is about the same distance away from, or in front of, the subwoofer, as specified in the setting. As room acoustics will influence effectiveness, we recommend listening tests using different delay settings, in conjunction with alternate polarity settings, to determine the best results.



Good news: SL Room Control for StudioLive AI-series loudspeakers not only provides the alignment delay you need to set up a satellite system, you can also group your systems separately and remotely set the level of each speaker or group from SL Room Control. Again, because SL Room Control communicates wirelessly from your laptop or iPad to your loudspeakers, you can set the level right from the affected part of the sanctuary.

If you are aligning for a custom installation you will need to do some calculating:

Find the spot in the room where coverage from the main speakers and the subwoofers overlap. Measure the distance from the overlap area to each speaker location. Subtract the smaller distance from the larger. Divide that number by 1,100 and apply that delay value to the speaker that is closest. Keep in mind that the overlap area may be behind front-of-house.

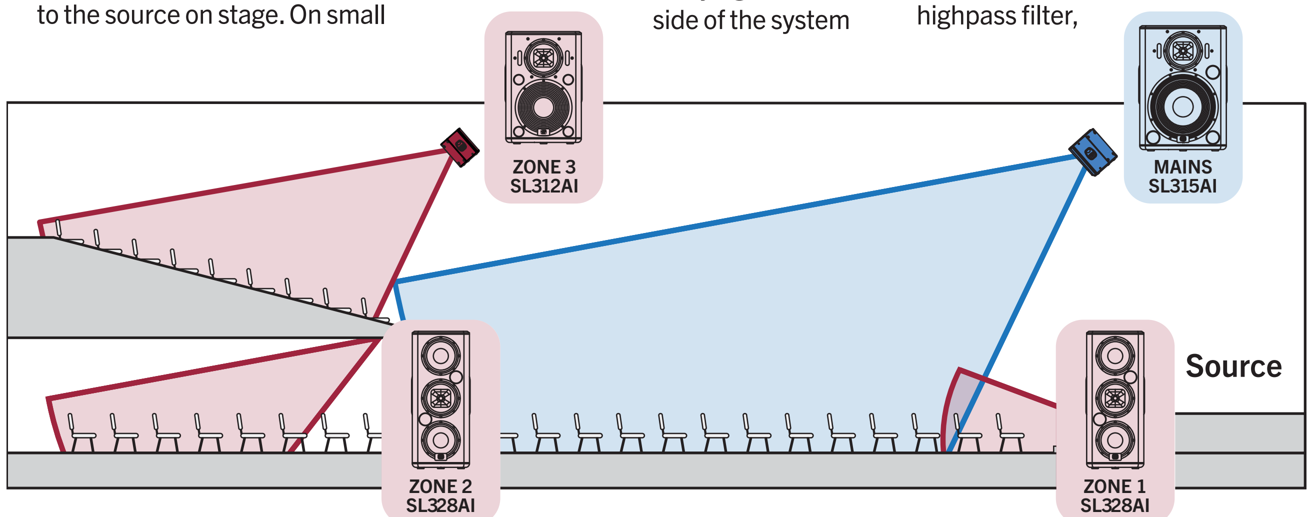


The goal in a complicated system with loudspeakers distributed throughout the sanctuary is to delay each satellite system relative to its counterpart in the main system (e.g., delay the left front fill relative to the left front-of-house loudspeaker).

- Delay the main system relative to the source on stage. On small stages where the guitar amp and drum kit can be clearly heard above the front-of-house (FOH) loudspeaker system, delaying the main system can "move up" the backline so that it aligns with these instruments and decreases blurring in the mix. This will tighten the overall mix and give it more punch.
- Delay the front fills relative to the main system by delaying each side of the system independently (e.g., delay the left front fill relative to the left FOH loudspeaker).
- Delay subwoofers relative to the main system. How you do this will depend on how your subwoofer system is positioned and configured. In general, you will want to delay each subwoofer relative to the full-range loudspeaker closest to it.
- Delay down-fill speakers (upper and under balcony and choir loft) relative to the main system, again delaying each side of the system independently.

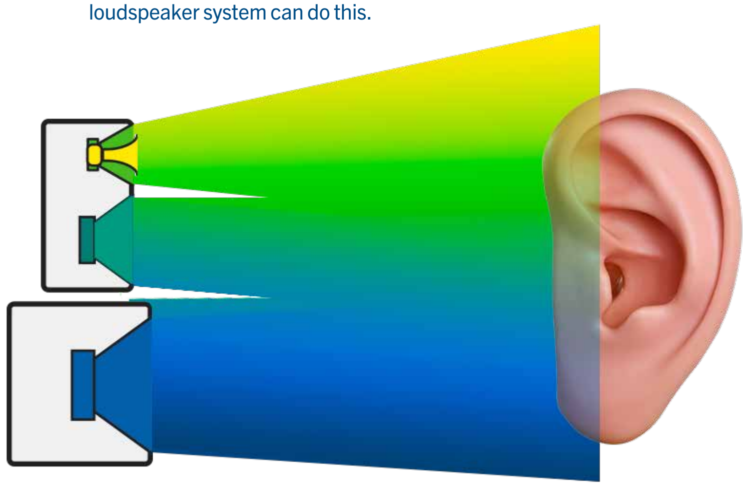
Good news: The full-range StudioLive AI-series loudspeakers are phase- and time-aligned to create a true 4-way system with the companion 18sAI subwoofer, with or without their 100 Hz highpass filter engaged.

With most 4-way systems, leaving frequency content below 100 Hz in the full-range loudspeaker can introduce destructive cancellations between the lowest part of the full-range frequency response and the highest part of the subwoofer's response. Most systems solve this by either requiring the use of the onboard highpass filter, or connecting an external crossover (see illustrations at the top of the next page).



Above: To deliver the same sound simultaneously to different parts of a church sanctuary, you need to create separated "zones", each with its own delay calculation.

Below: The powerful processor in StudioLive AI loudspeaker systems can be used to create a true, 4-way system where the output of the 18sAI subwoofer blends seamlessly with the 3-way main speakers. No other compact active loudspeaker system can do this.



StudioLive AI-series loudspeakers are designed to avoid this problem when combined with an 18sAI subwoofer. No crossover or external speaker processor is required to keep them phase and time aligned.

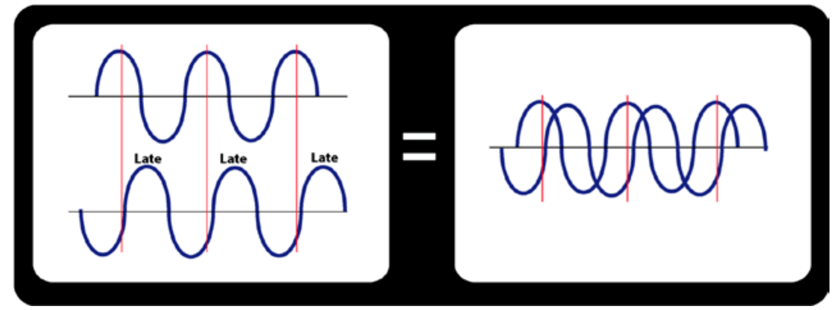
When all the components of a 4-way system are phase and time aligned, they work together, instead of against each other, to deliver the tightest bass response possible.

An aligned frequency overlap at 100 Hz will let you achieve a bigger bass sound without any extra effort and will deliver tighter, punchier bass.

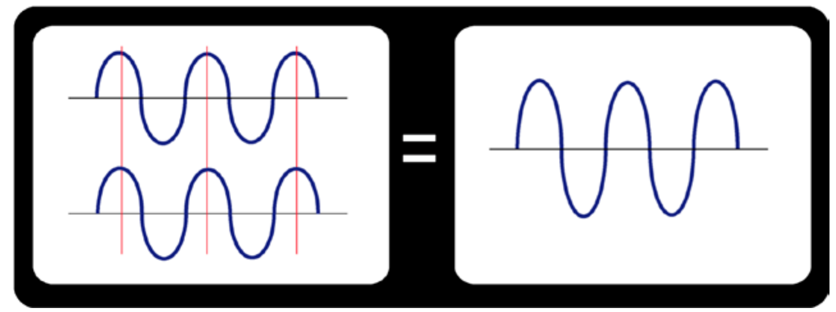
For applications that require a more linear frequency response, just engage the highpass filter on the full-range AI-series loudspeakers. This will retune the speaker and keep it in phase with the 18sAI, so you still get tight bass response, just without the bump at 100 Hz.

This flexibility allows you to create the best four-way system for your application. We've also included alignment-delay presets on the 18sAI for the three most common 4-way configurations (below). This ensures that your 4-way system will remain aligned even if the subwoofer is not directly beneath the full-range system.

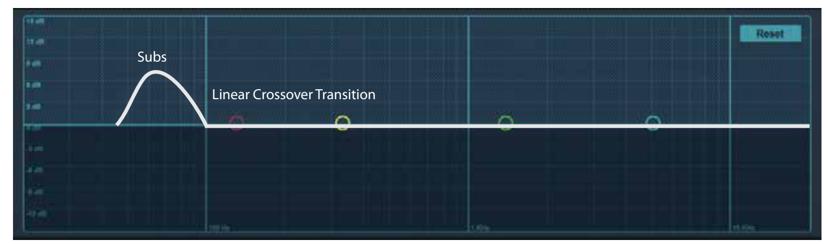
For systems that require a longer delay time, SL Room Control provides access to each loudspeaker's 500 ms alignment delay.



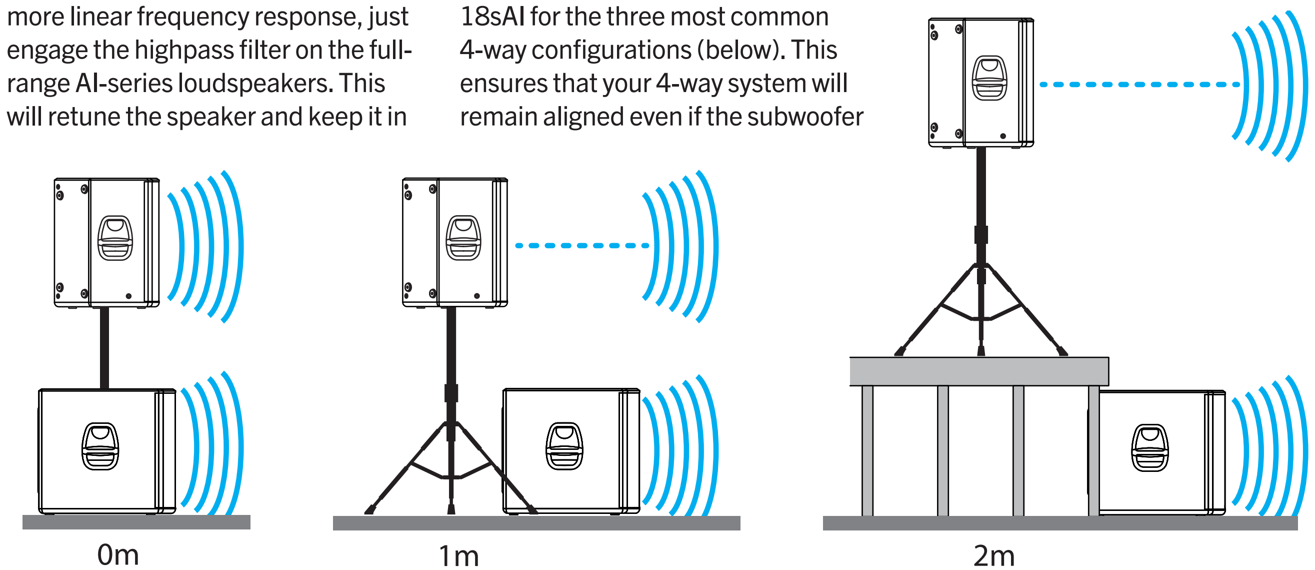
Two sound sources *out of phase*, create a smeared, garbled sound.



When two sound sources are *in phase*, the combined sound is clear.



The full range system handles frequencies down to 100Hz; then the subwoofers kick in.



System Management Tools

System-management devices allow you to create EQ settings that compensate for room anomalies, set delay times for satellite systems, and dial in crossover transitions between full-range systems and subwoofers. There are many outboard system management devices on the market.

This section will discuss the system management tools built into StudioLive AI-series loudspeakers and accessible via SL Room Control software. Whether onboard StudioLive AI loudspeakers or inside a rackmount unit, these tools have the same benefits. The largest benefit of having them built into your loudspeaker is that there is no extra cabling involved!

Speaker Grouping. As its name implies, this feature allows you to create a group of speakers and control them individually and as a single unit. This allows you, for example, to adjust the overall level of your entire front-of-house system, including delay satellites, while maintaining the relative level between each loudspeaker. SL Room Control also provides a 31-

SL Room Control 31-band Group Speaker EQ

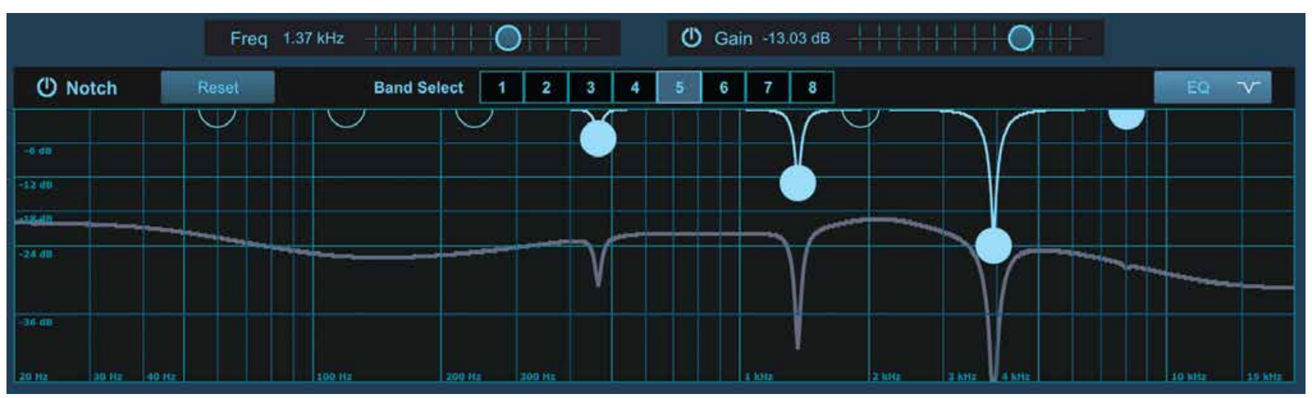
band graphic EQ so you can make frequency adjustments to your entire system as needed (shown above).

Notch Filter. A notch filter is a very narrow EQ band that only provides attenuation; that is, you can only lower the level of a specific frequency band. SL Room Control provides eight notch filters per speaker, each with a Q of 24—that's less than a half-step. This allows you to surgically remove unwanted frequencies from floor monitors and to quickly remove feedback from the source. Ringing your floor monitors is an extra step before services that will save you major headaches once the congregation shows up on Sunday.

Good news: VSL-AI for StudioLive AI-series mixers provides a visual tool to quickly identify feedback problems. Using the Spectrograph (see the colorful screenshot at the top of the next page), you can easily see which frequency is causing the problem and remove it using either of your mixer's EQ options (parametric or Graphic) or switch over to SL Room Control and notch it out permanently.

Parametric EQ. Parametric EQs offer continuous control over the audio signal's frequency content, which is divided into several bands of

SL Room Control Notch Filter



VSL Smaart Spectrogram and 4-band parametric equalizer.

frequencies. SL Room Control provides eight bands per loudspeaker (screen shot at the bottom of this page). Each band offers continuous control of the gain (boost/cut) for each frequency band, the bandwidth, and the center frequency for the band. Use SL Room Control's parametric EQ to help mitigate room anomalies without using your mixer's EQ. Another advantage is that you can EQ each speaker separately so that you are adjusting it for its surrounding environment rather than applying an EQ to the entire system.

Level. SL Room Control allows you to remotely adjust the level of an individual speaker or an entire group. This not only makes setting up delay systems much easier, it allows you to set a "brick wall" so a volunteer can't crank up the system too much and damage your loudspeakers.

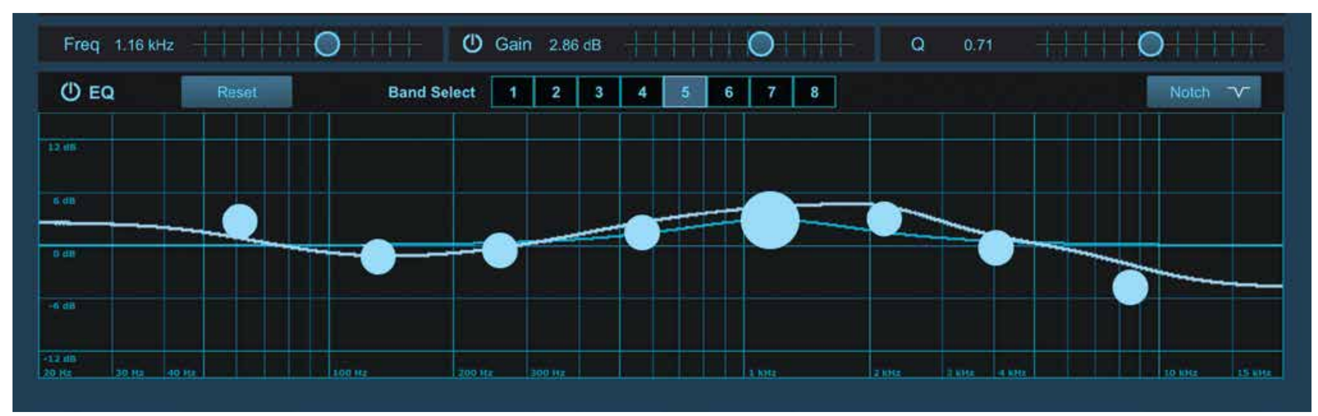
SL Room Control Speaker Level adjustment

Limiter. As its name implies, a limiter sets a sound-level threshold that the audio signal can't exceed. This is a great way to protect your system, especially your floor monitor system, from feedback. SL Room Control provides a limiter for each loudspeaker.

SL Room Control Limiter adjustment

Alignment Delay. As we discussed in previous sections, an alignment delay lets you align speakers that are placed some distance apart to compensate for sound traveling slower than electricity. With a properly aligned system, the sound will naturally fill the room and the transition

SL Room Control 8-band equalizer.



SL Room Control Alignment Delay adjustment

from one system to another will be undetectable.

SL Room Control provides 500 ms of delay per speaker and allows you to make adjustments in 0.1 ms increments for precise control.

Performance Monitoring. Being able to check in on how your loudspeakers are performing is critical to keeping a well-maintained PA system. This can be tricky once your loudspeakers are installed and the rear panel, where performance indicators are usually placed, is hidden from view. SL Room Control provides extensive monitoring for every loudspeaker in your system from your iPad or laptop. In SL Room Control, you can view input metering, clipping, a protection limiter, speaker excursion, and real-time temperature. These let you head off problems before they become dangerous to your loudspeaker, and they are especially important when novice operators are running the system.

Good news: SL Room Control is a free, wireless remote-control application for StudioLive AI-series loudspeakers that runs on Mac, Windows, or iPad and provides an easy-to-use network setup wizard. You have the option of hardwiring your loudspeakers to a wireless router with Ethernet cables or connecting them wirelessly using the including Wi-Fi LAN adapter.

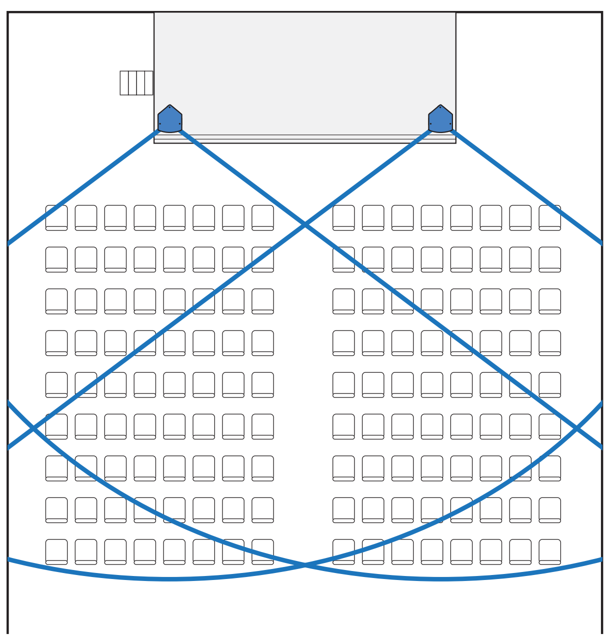
System Configuration Suggestions

The following discussion and graphics will demonstrate some system configurations for common rooms. The size and shape of your sanctuary and the application for which it will be used determine, to a large extent, how many speakers you will need and where they should be placed. In every situation, keep in mind two important design factors: your loudspeaker's coverage pattern and half-space loading.

Every full-range StudioLive AI loudspeaker offers a 90˚ horizontal x 60˚ vertical coverage pattern. If you are using StudioLive AI-series speakers, be sure to pay close attention to these angles when using your speakers. Rotating the cabinet changes the horizontal and vertical coverage. If you are using conventional (non-coaxial) loudspeakers, find out what their coverage pattern is and figure accordingly.

When configured for stereo use, make sure the cabinets are not placed too wide for the room or too far back into the corners. Too wide of a placement will direct too much energy onto the walls and can potentially add destructive interference to the room. Adjust the left and right speakers, as well as the toe-in angle, to produce the best stereo image. If a room is very narrow, a mono cluster might be a better choice than stereo.

Wherever you place your loudspeakers, you should be aware of half-space loading. Half-space loading occurs when a speaker comes in close contact with, or touches, a hard surface like a floor or wall. As its name indicates, this type of summation happens when the circular radiation of the speaker is blocked by a hard surface and forced to radiate in a crescent shape. Depending on the proximity and position, there may be a boost in low-frequency energy.



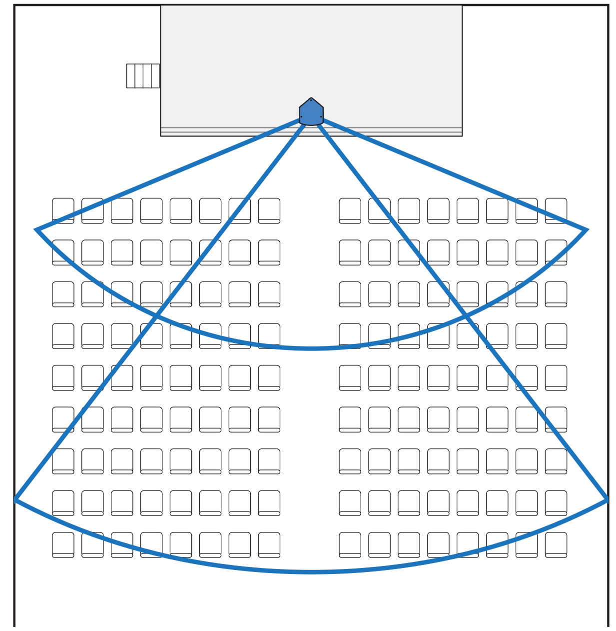
Testing your speaker placement and doing some critical-listening tests will help determine the best final location for your loudspeaker system.

If your speakers are sitting on the floor, you can expect a certain amount of half-space loading. If you are using your speakers as floor wedges, you might want to experiment with using a highpass filter, like the one built into all full-range StudioLive AI loudspeakers, to reduce low energy. In some cases, this might improve intelligibility.

Good news: A floor monitor placed on the stage is unavoidably subject to half-space loading, so the Floor Monitor DSP contour in a StudioLive AI-series full-range loudspeaker is designed to compensate for bass buildup and maintain a tight mid-bass response. Using the Floor Monitor DSP contour will also help you get the best use out of StudioLive AI loudspeakers in this position.

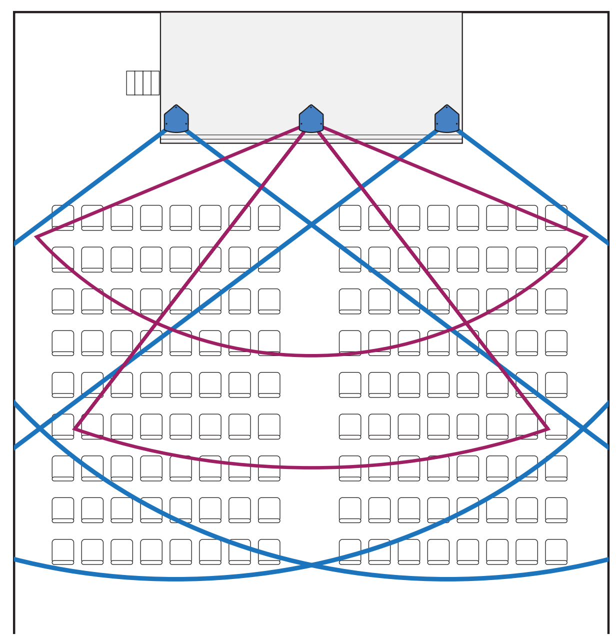
Stereo System. A stereo system (shown below) allows panning and adds depth to the acoustic image. This is good for speech reinforcement and greatly enhances live or pre-recorded music. Locate speakers to give the best horizontal coverage. Ensure that the listeners are well covered by the pattern.

Mono Cluster with Down Fill. Center or mono systems can provide a simple, economical solution for churches where speech intelligibility is the priority, rather than music. As with a stereo system, make sure the coverage pattern of the speaker focuses the energy on the congregation.



The graphic above actually shows two speakers. The upper speaker is for thro wing to the back of the room, and the lower speaker covers the space in the front of the room, closest to the stage.

Typical LCR church system.



LCR Systems. An LCR system (shown below) is a stereo system with a center speaker added. This configuration allows panning and adds depth to the acoustic image. This type of system will provide more control than a basic stereo system and is ideal in situations where music and speech intelligibility are equally important.

If you've read this far…

…you've learned a lot about how to get good sound in a church environment.

Some of what we've imparted here can be applied to use with any conventional loudspeaker system. But a lot of it — the parts that help you truly optimize a PA system to your sanctuary's acoustics, size and shape — would only be possible if you choose our StudioLive AI mixer and CoActual loudspeakers. Otherwise, you'd need a whole lot of expensive processing equipment — and considerable sound engineering experience.

Our software makes it easy.

The most important thing that sets PreSonus live sound products aprt from our competition is the remarkable, integrated suite of software programs we've seamlessly integrated into them.

Volunteers can now perform sophisticated adjustments that used to require extra equipment and considerable expertise.

You — yes *you* — can work wonders, dramatically enhancing the quality of the sound your congregation hears at every service.

And since we have a bit of space left on this page…

…we're going to add a shameless plug for some of the other benefits of that integrated suite of software that comes with our mixers.

One-click recording. Just press the RECORD icon in Capture 2 and the whole service is recorded in multi-track *and* stereo.

Capture 2 comes free with our mixers.

Open the stereo version in Studio One Artist DAW, cut out that pause during the Offering and the pastor's

Studio One Artist DAW comes free with our mixers.

cough, and post the service online before some of the congregation can even get back home.

Play back the multitrack version to do a virtual soundcheck early next Sunday morning before the praise band even gets there. Or use it to help train volunteers on how to mix.

You can't do this with any other brand of compact digital mixer.

Visit your PreSonus dealer soon. And God bless.

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