



## HOW TO USE THE SHURE MXA910 CEILING ARRAY MICROPHONE FOR VOICE LIFT

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The Shure MXA910 Ceiling Array Microphone offers the unique advantage of eight independently steerable lobes, with very narrow pick up patterns. In AV conferencing applications, this allows the microphone array to be placed as much as four times farther from the sound source than an omnidirectional microphone, while maintaining a similar ratio of speech vs. ambient sound performance.



Many customers ask if the MXA910 can also be used in AV conferencing systems that require voice lift. The answer is “yes.” When compared to an omnidirectional or cardioid microphone, the MXA910 can provide from 7 to 11 dB additional level of voice lift before feedback starts to occur in a typical conference room.

For those who have successfully installed and configured voice lift systems in conference rooms in the past using omnidirectional microphones on the ceiling, or cardioid mics hanging from the ceiling, you may already know what it takes to accomplish voice lift with the MXA910 in similar rooms, and achieve even better results.

The implementation of voice lift in an AV conference system is complex. Special considerations range from separate signal paths for the “transmit audio” and voice lift system, to configuring virtual references for the acoustic echo cancellers. In addition, there are some specific considerations when using the MXA910 for voice lift that can help get the most from the system.



It is important to differentiate voice lift from sound reinforcement (SR) and public address (PA). A SR or PA system is designed to amplify the sound, including voice, to such levels that it can be understood in very noisy environments and at great distances from the loudspeaker. A voice lift system, by contrast, is intended to merely restore the natural speech level from a talker to the farthest listeners in a room. In this case, the reason the farthest listener cannot hear is not because of loud room noise, but rather because of the natural acoustical loss in level of speech that occurs when traveling a distance of 25 feet or more to reach the listener. In a properly tuned voice lift system, you can hear the farthest

talkers clearly without even realizing that the system is turned on. Voice lift systems provide the minimum amount of gain to establish intelligibility.

This document highlights some considerations for voice lift systems that are specific to the MXA910, as well as the voice lift application for conference rooms in a broad sense. We discuss techniques that help the system designer and the installer properly design, install, and configure the system for maximum gain-before-feedback.



## MXA910 Special Considerations:

- 1- Polar Pattern and EQ
- 2- Automatic Mixing
- 3- PAG/NAG

## Additional Techniques for Voice Lift systems:

- 4- Room size and acoustics
- 5- Zoned Loudspeaker System
- 6- Mix-Minus and AEC Reference

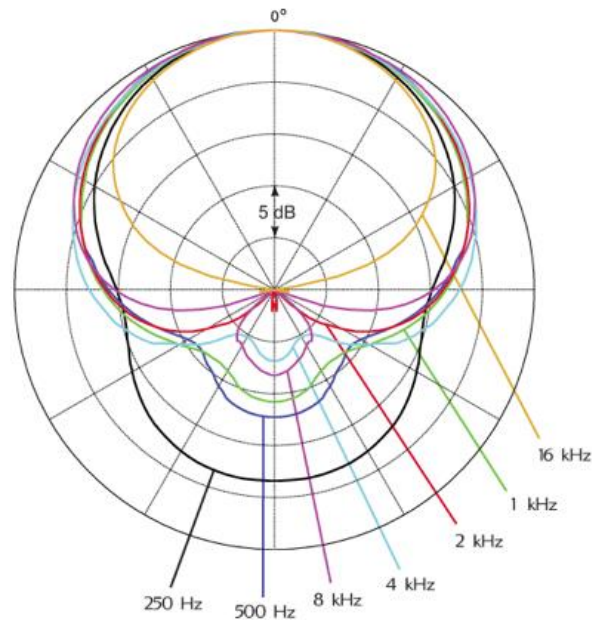
### 1-POLAR PATTERN AND EQ

*All microphones are less directional (more “omni”) at lower frequencies and, while the MXA910 remains more directional than even a hyper-cardioid microphone at low frequencies, it is no exception to the rule. Shure testing has shown that the best performance on voice lift systems is achieved when using lobes set to Medium and Wide instead of the narrow setting.*

*Additionally, all the MXA910 channels are coherent which, as explained further in this document, also increases low frequency response. The following 2 EQ filters are recommended:*

- 1- *Hi-pass (Low Cut) EQ for normal speech systems: Filter setting of 150 to 280 Hz and a slope of 18 dB/octave. The exact filter frequency will depend on room noise and acoustic characteristics. Listening to the channels on headphones helps determine the ideal frequency.*
- 2- *In MXA910 units with firmware v1.3.4 or later, keep the “Low-shelf Filter” button engaged. This filter deploys a Low-shelf EQ at 960 Hz / -10 dB to prevent the buildup of low frequencies due to the higher sensitivity of the microphone at lower frequencies.*

A properly aimed unidirectional microphone helps a voice lift system by rejecting sound from the loudspeaker and allowing the system to get louder before feedback starts to occur. However, all microphones are less directional at low frequencies. A typical cardioid microphone has a coverage angle of 131 degrees at 1 kHz, and a supercardioid microphone is 115 degrees. But both types are much wider at lower frequencies, and become practically omnidirectional below 150 Hz. This effect reduces the microphones ability to reject sound from the loudspeaker at lower frequencies.

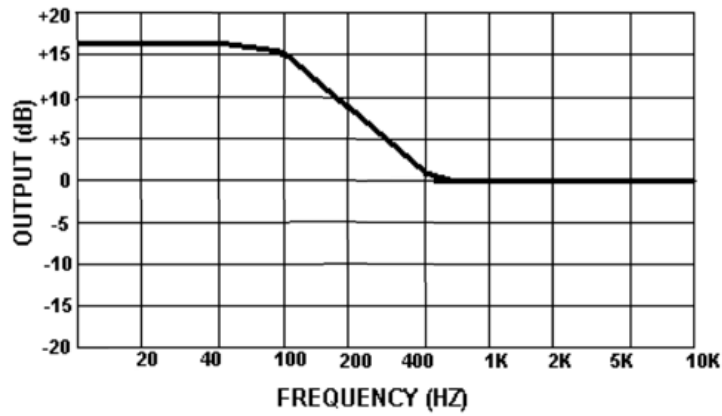


**EXAMPLE OF SUPER-CARDIOID POLAR PATTERNS  
MEASURED AT DIFFERENT FREQUENCIES**

Each MXA910 lobe is much more directional than any unidirectional microphone, and the coverage pattern of each lobe is adjustable to 35, 45, or 55 degrees. This high degree of directionality from the MXA910 allows greater voice lift levels from farther away than other microphones before feedback starts to occur. However, it is not immune to the laws of physics, so it also is less directional at low frequencies.

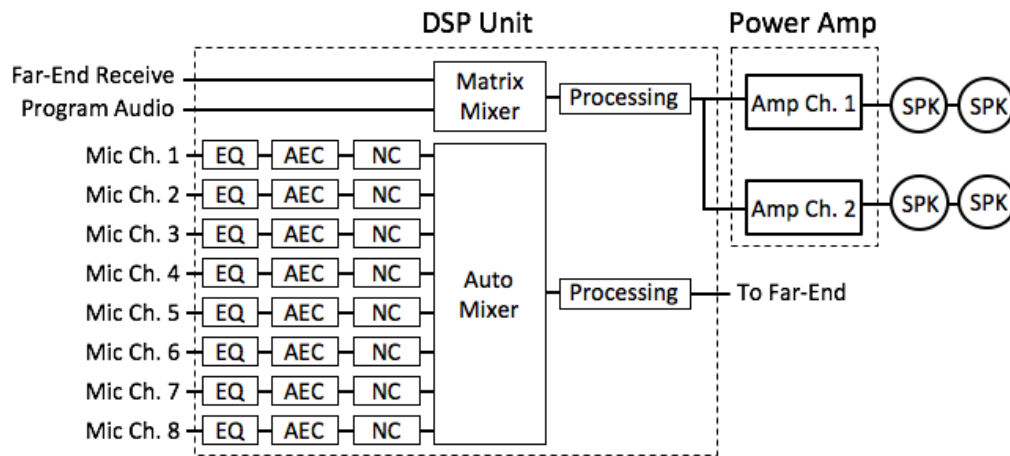
Like any other microphone, in order to achieve maximum voice lift levels without feedback, apply a hi-pass (or low-cut) EQ filter to each microphone channel set between 150 and 280 Hz, depending on room noise and acoustics characteristics. Additionally, unlike other systems with multiple microphones, all channels of the MXA910 are coherent, since the eight microphone lobes all originate from the same physical location. Because of this, the resulting cumulative low frequency response from multiple channels is significantly higher, even when an automatic mixer is in use. To correct this increase in low-end response, Shure recommends using a Low-Shelf EQ set at approximately 960 Hz and with an attenuation of -10 dB. On microphones with firmware version 1.3.4 or later, the EQ is engaged by default via the “Low-shelf Filter” button on the top left of the UI.

Note that additional parametric EQ filters should also be applied on the output driving the loudspeakers to reduce level at specific frequencies that are naturally amplified in the environment due to room dimensions and surface acoustic characteristics.



EXAMPLE OF CUMULATIVE RESPONSE FROM MULTIPLE CO-RELATED DIRECTIONAL MIC ELEMENTS

There are several places in the signal path where EQ filters can be applied. Shure recommends EQ be applied before the Auto-mixer by using the built-in EQ on the MXA910 or by using input processing on the DSP. This also benefits the Automatic mixer which, as discussed later in this document, is a key component of both the Voice Lift and AV Conference system.



SIGNAL DIAGRAM OF SYSTEM WITH EQ PRE-AUTO MIXER (WITHOUT VOICE LIFT)

**2-AUTOMATIC MIXING**

*Automatic mixers are a key component of any Voice Lift and AV Conference System because they reduce the number of active microphones. This significantly improves intelligibility as well as the amount of gain-before-feedback the system can reproduce when compared to having many microphones active simultaneously. But not all automatic mixers are designed for voice lift systems. A gating automatic mixer with an off-attenuation of -20 dB or more is recommended for voice lift, as opposed to a gain sharing automatic mixer. The MXA910 has a separate Dante audio output with built-in IntelliMix automatic mixing that operates as a “gating” automixer when set to “Classic” mode. Off attenuation is set to -20 dB by default, but can easily be changed to any other level.*



In any situation where multiple microphone channels are used, automatic mixers help the voice lift system provide more gain-before-feedback by activating channels only for active talkers while keeping the rest of the channels turned down. A sound system with four simultaneously open microphones would have 6 dB less acoustic gain-before-feedback than a system with one microphone, or with an automatic mixer in use. Also, automatic mixers often include a function called Number Of Open Microphones Attenuator (NOMA), which reduces the output of the mixer to prevent feedback as more talkers become active and engage multiple microphones simultaneously.

However, not all automatic mixers are the same, and not all are appropriate in every application. While a gain sharing automatic mixer (which Shure refers to as “Smooth Mode” in Shure Audio Processors) is often preferred for many applications, Shure recommends a gating automatic mixer (which Shure refers to as “Classic Mode”) for the MXA910 to provides more gain before feedback in a voice lift application and also better speech-to-room-noise ratio in a AV Conference system.

A gain sharing automatic mixer gives more gain to the mic of the loudest talker and then dynamically takes away gain from all other mics. Having the mixer automatically turn a channel up too much for a loud talker can often drive the system into ringing or even feedback. Therefore, more margin of stability needs to be considered up front when designing the system. This can penalize the voice lift level for all other talkers that are not as loud.

Additionally, and perhaps more importantly, recall that all channels in the MXA910 are coherent. This means that summing two channels together, whether they are fully “on” or if they are attenuated, results in a higher level than it would when using any other type of microphone. Depending on frequency and where the channels are aimed, or the amount of overlap between channel position, the summing of two channels results in as much as a 6 dB gain increase instead of the normal 3 dB bump.

This behavior makes it easy to underestimate the contribution that the *attenuated, or “off,”* channels make to system gain. Although a gain sharing automatic mixer dynamically allocates gain to each channel, a mixer with eight microphones would only attenuate the *OFF* channels by approximately 9 dB when just one person is talking. But for the MXA910, the needed attenuation may be closer to the -18 dB range. This difference of approximately 9 dB is high enough that it cannot be ignored, and it may make the system unusable if it does not have enough feedback stability margin to absorb the additional 9 dB of gain.

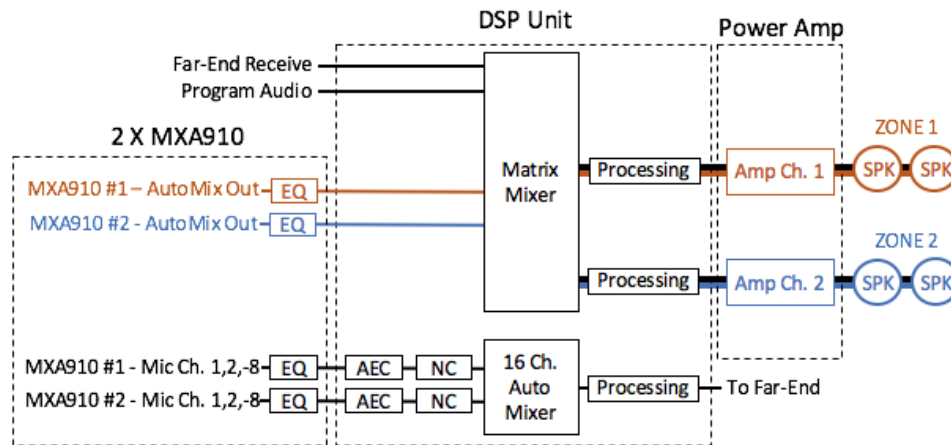
A gating-style automatic mixer has a user setting for off-attenuation level. All mics that are not activated by an active talker are attenuated equally to the level set by the user. In this case, it is very easy to dial the off-attenuation to -20 dB or more and allow the system to remain stable (i.e. without feedback) at all times. This allows the voice lift system to be louder for active talkers that have their channel gated ON.

Since the voice lift system should only be providing enough amplification for talkers to be heard at a natural speech level, any pumping of background noise caused by the activation, or *gating*, of mics should be masked by the background noise of the room itself.

In addition to the direct output from each microphone channel, the MXA910 has a ninth output for the built-in IntelliMix® Automatic Mixer. When the *Classic* preset is selected in the user interface, the algorithm used is a Gating Automatic Mixer. The default Off Attenuation is -20 dB and is adjustable up or down to minus infinity.



A straightforward application for voice lift in a system with two MXA910s and two loudspeaker zones would be to use the built-in automatic mixer output of each MXA910 to drive each loudspeaker zone separately for voice lift. Use the direct outputs from both microphones into a separate Gating automixer for the video conference feed, with all direct output channels mixed. In this system with over 8 MXA910 channels, a good starting level for the Off-Attenuation setting would be -24 dB.



SIGNAL DIAGRAM OF VOICE LIFT SYSTEM USING THE PROCESSING IN THE MXA910

### 3-PAG-NAG

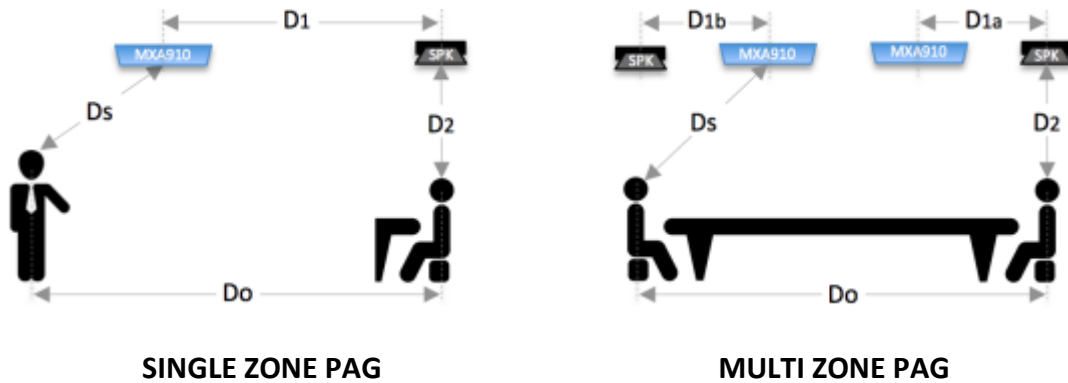
*The MXA910 can provide 7 to 11 more dB of additional gain in voice lift systems when compared to an omnidirectional or a cardioid microphone used in ceiling applications. The Potential Acoustic Gain (PAG) and Needed Acoustic Gain (NAG) equations allow a system designer to calculate whether the system will be able to provide enough gain-before-feedback before the system is built or modified. We have created web calculator as well as an Excel spreadsheet to help easily calculate PAG/NAG for omni, cardioid, and MXA910 microphones. The spreadsheet also includes calculations for multi-zone systems. In a system with multiple zones, both single and multi-zone sides should be filled out and the correct result is the lower of the two.*

Even after observing all the considerations listed in this document, it is possible that the system would start to feedback prior to reproducing a loud enough voice level to be useful. This would typically indicate that the room is too small for voice lift, the loudspeaker zones are not far enough apart, or the microphone is placed too far from the talker.

Using a combination of PAG (Potential Acoustic Gain) and NAG (Needed Acoustic Gain) equations, you can calculate fairly accurately whether the system will provide enough voice lift without feedback.

The standard PAG equation, in use for decades, assumes omnidirectional components. However, as we've discussed so far, the highly-directional lobes of the MXA910 mean that under the right conditions, it can allow

a voice lift system to outperform the predictions of the standard equation. To assist in determining whether the MXA910 will provide enough gain-before-feedback, we have developed this “modified” PAG calculator, which accounts for the directionality of the MXA910 lobes by adding a 9 dB “bonus” to the PAG result. In order to achieve the results shown by this version of PAG, the guidelines detailed in this document must be strictly adhered to. Please pay special attention to the following section, “Room Type and Environment,” as room acoustics play a primary role in determining the real-world application of this equation. Note that results of this equation are not a guarantee of system performance, but serve as a useful tool for system design and estimation of system performance. In a system with multiple zones, both single and multi-zone sides should be filled out and the correct result is the lower of the two.



VOICE LIFT CALCULATOR		
Fill out YELLOW cells		
	SINGLE ZONE	MULTI ZONE
Ds - Distance from Talker To Mic	10 Ft/M	10 Ft/M
Do - Distance from Talker to Listener	25 Ft/M	25 Ft/M
D1 - Distance from Mic to nearest Loudspeaker	16 Ft/M	
D1a - Distance from Mic to nearest Loudspeaker 1		8 Ft/M
D1b - Distance from Mic to nearest Loudspeaker 2		8 Ft/M
D2 - Distance from farthest Listener to nearest Loudspeaker	10 Ft/M	10 Ft/M
NOM - Number of open Mics or Lobes. 1 if Automixer is used	1 Mics	1 Mics
PAG (Potential Acoustic Gain)	6.04 dB	6.04 dB
NAG (Needed Acoustic Gain)	12.40 dB	12.40 dB
If results below is 0 dB or greater, the system should provide adequate Voice Lift level		
MXA910	2.65 dB	11.65 dB
CARDIOID Mic	-5.35 dB	-5.35 dB
OMNI Mic	-6.35 dB	-6.35 dB

## 4-ROOM TYPE AND ENVIRONMENT

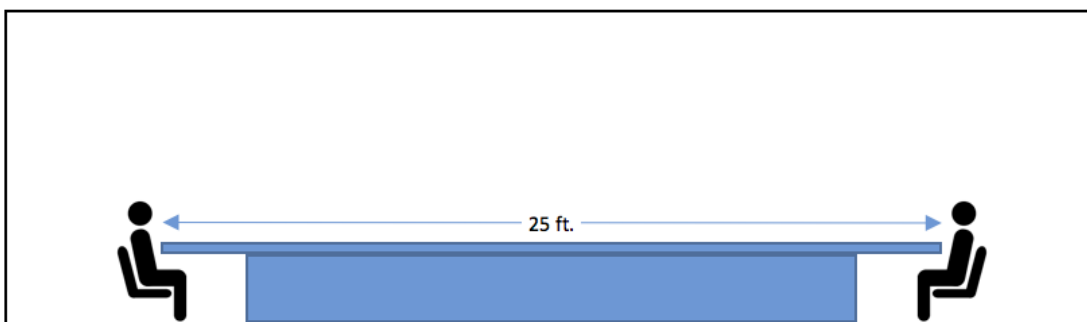
A voice lift system typically cannot be implemented in a room that is so small that it doesn't actually require voice lift, or in a room that requires voice lift due to loud background noise. The background noise level of the room should be less than 45 dBA (NC30-35). It should also be noted that voice lift will not be beneficial in a room where speech cannot be understood due to excessive reverberation (long decay time).



The first rule of voice lift systems is do not attempt to implement a voice lift system in a room that is too small to actually require it. In other words, if a listener on the one side of the room can clearly understand a talker on the other side of the room, voice lift is not required.

Excessive noise from air handling, hallway traffic, electronic equipment or other sources, sometimes combined with reflective surfaces like glass walls, can create conditions where relatively small rooms may appear to need voice lift. However, the solution in those rooms would be to control the noise issues and apply acoustic treatment. Trying to implement a voice lift system in rooms under these conditions will likely not work.

A room would need to have a dimension of at least 30 feet, with a talker-to-listener distance of at least 25 feet, before a voice lift system should be considered. Such a room should have a background noise level of no more than 45 dBA (NC30-35).



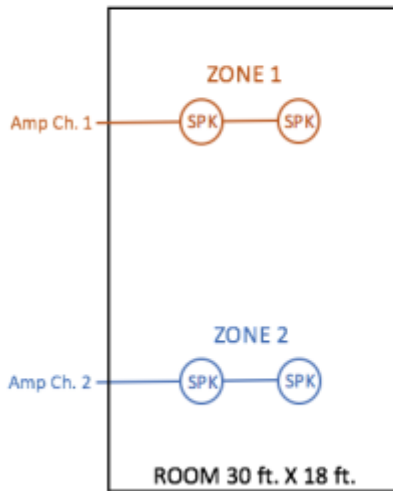
EXAMPLE OF MINIMUM DIMENSIONS  
FOR A VOICE LIFT SYSTEM

## 5-ZONED LOUDSPEAKER SYSTEM

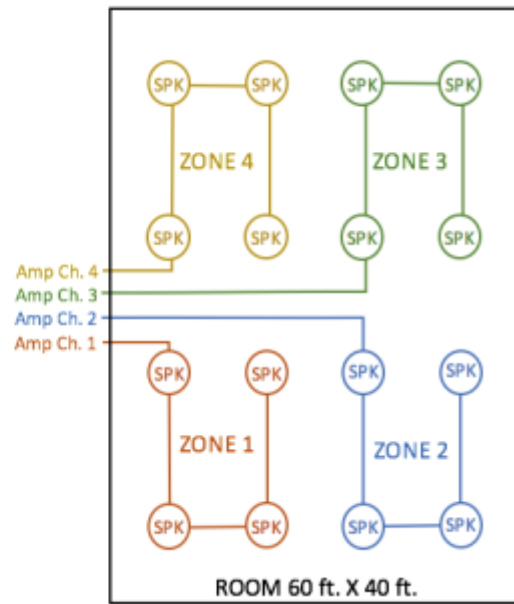
*To achieve adequate voice lift levels with any ceiling microphone, the signal from a microphone must be reproduced only by those loudspeakers that are farthest from that microphone.*

In order to achieve sufficient voice lift level in a room using any type of ceiling mounted or overhead microphone, there must be at least 2 zones of loudspeakers. Loudspeaker “zones” typically refers to one or more loudspeakers that are located in an area of the room and are electrically isolated and separately driven from loudspeakers physically located in a different area of the room. This will allow microphones located in one side of the room to drive only the farthest away loudspeaker zone located in the opposite side of the room.

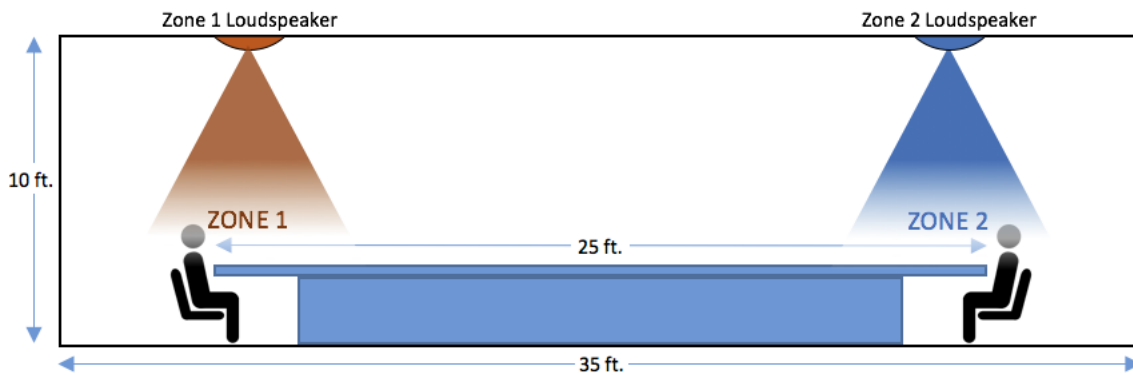




SYSTEM WITH 2 LOUDSPEAKER ZONES



SYSTEM WITH 4 LOUDSPEAKER ZONES



ROOM WITH TWO LOUDSPEAKER ZONES

## 6-MIX-MINUS AND AEC REFERENCE

*Similar to loudspeaker zones, mix-minus is used to remove a microphone channel from the mix on the output that drives the loudspeaker that is physically closest to it.*

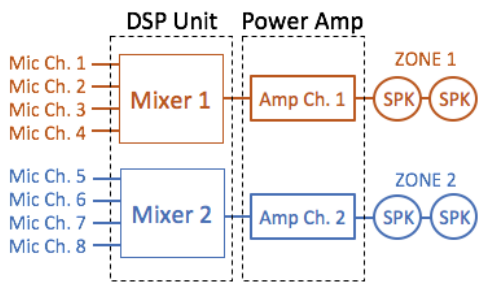
*When using voice lift in a system that is also used for AV Conferencing, multiple different references will need to be setup for the AECs to prevent undesired audible artifacts.*

Mix-minus means creating microphone mixes where some microphone channels have been purposely removed from the mix. This sub-mix of microphones, or mix-minus, can then be routed to the loudspeaker

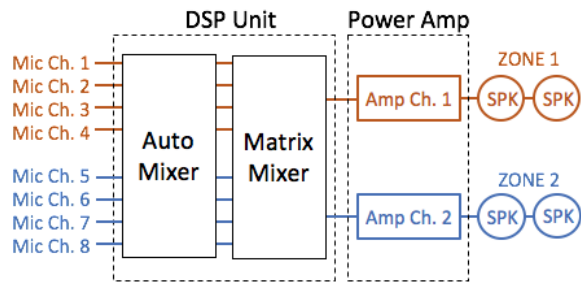
zone that is closest to the microphones that were removed from the mix. This implementation makes voice lift possible in the room where the system otherwise would not be able to produce enough gain before feedback.

Mix-minus can be created using different options or processors inside a DSP unit. One way is to employ multiple small mixers and route only the correct microphones to each mixer to create a sub mix for each loudspeaker zone.

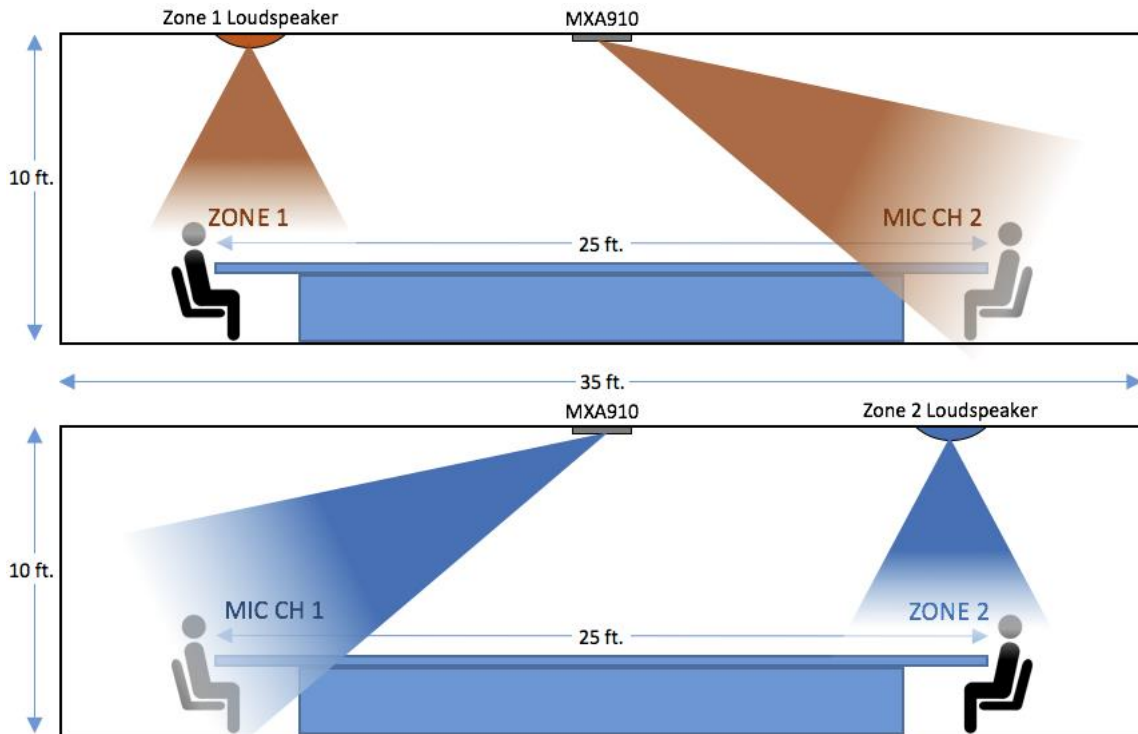
For voice lift applications, always use automatic mixers to obtain maximum gain-before-feedback. In this case, an automatic mixer with direct outputs is desired. Each direct output of the automatic mixer can then be routed to a matrix mixer to create the desired sub-mix for each loudspeaker zone.



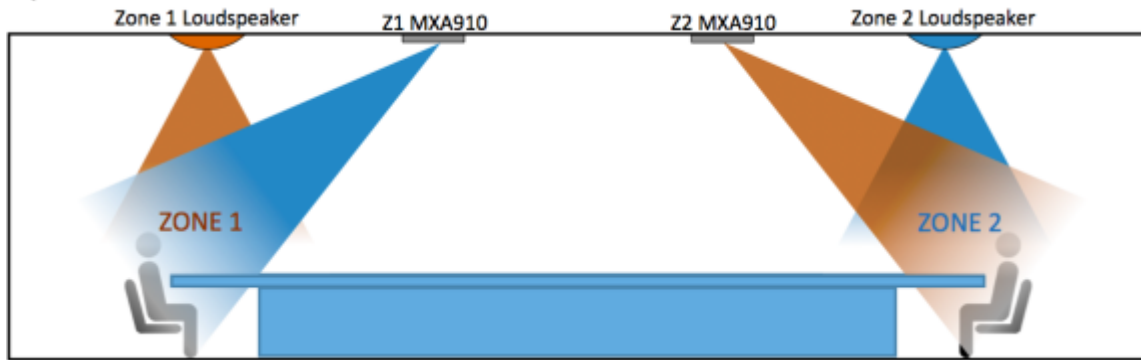
MIX-MINUS SYSTEM WITH 2 MIXERS



MIX MINUS SYSTEM WITH AUTO MIXER AND MATRIX MIXER



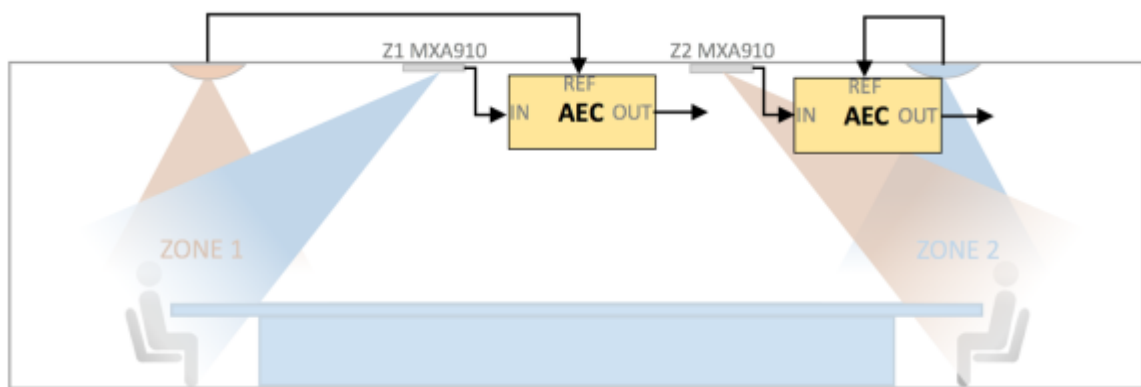
MIX MINUS SYSTEM WITH 2 ZONES



VOICE LIFT SYSTEM WITH TWO ACTIVE ZONES

One last important factor to consider in a system that employs a both Mix Minus for voice lift, and is simultaneously used for AV Conferencing, is the reference for the acoustic echo canceller (AEC). Just adding voice lift to the traditional AV Conference design where all AECs use the same reference will not work properly. If the AEC reference DOES NOT include the local voice lift signal, the far end will hear the local room with increased reverberation (not echo), as if talkers were much further from the microphone or room acoustics had gotten worse. However, if the AEC reference DOES include the local voice lift signal, then the AECs will malfunction, causing major audio loss and artifacts. The reference signal assigned to an AEC cannot contain its own input.

The proper setup is to use an AEC block that provides a different reference input for each AEC channel. The best signal to route to each AEC reference input is the output that drives the loudspeaker that is closest to that input channel. Because of the Mix Minus setup, the closest loudspeaker to any microphone input channel would not contain that microphone's signal, but would contain a Mix of all other signals the AEC needs to cancel to prevent echo and other artifacts.



EXAMPLE OF AEC REFERENCE FOR VOICE LIFT + AV CONFERENCE SYSTEM

For additional information or for special applications beyond what is covered by this document, please contact our Shure Applications Engineering: [support@shure.com](mailto:support@shure.com)

Answers for all Shure products can be found at: <http://shure.custhelp.com/app/answers/list>

MXA910 Best Practices and specific answers can be found at: <http://shure.custhelp.com/app/answers/list/st/5/kw/MXA910/>