Less is More: Use Cases for Lobe Aiming

Beamforming microphones combine an array of microphone elements with intelligent DSP to enable definable polar patterns. When the intelligent DSP includes beam tracking and/or multiple beams, these beamforming microphone arrays are excellent for capturing simultaneous talkers accurately, such as in an audio or video conferencing scenario. The beamforming aspect of these array microphones is that the “beams” (polar patterns) of the mics can be controlled and shaped through DSP, allowing the beams to be “aimed” at the appropriate areas in the room.

As with all microphones though, the room acoustics and ambient noise have a direct impact on the audio quality. If speech originating in the room is unintelligible to listeners in the same room, no microphone can successfully overcome this acoustical handicap. In this technical note, we’ll explore some of the design considerations and tradeoffs for lobe design in different use cases involving beamforming microphones. First, let’s recap the primary goal of any conferencing system: speech intelligibility.

Speech Intelligibility (Does the Room Sound Good?)

A conference room either sounds good or it doesn’t – there’s very little gray area - but that’s typically a subjective assessment by the listener. However, if you and others are regularly straining to understand the far end of calls, then the speech intelligibility in the room is low.

Speech intelligibility is the most important consideration for conferencing applications; if the listener(s) can’t understand the talker, the tele- or videoconference becomes ineffective. Proper lobe aiming plays a significant role in speech intelligibility, but other variables can include:

- the talker’s speech level (average is 65-67 dBA at 3 feet [1 meter])
- frequency response of the transducer
- background noise level (room noise floor)
- quality of the sound reproduction equipment
- echoes (reflections with delay > 100ms)
- reverberation time (RT60)[1]
- psychoacoustic (masking) effects

Quality of speech transmission is measured via the Speech Transmission Index (STI). STI predicts the likelihood of syllables, words, and sentences being successfully comprehended by the listener. STI is a numeric representation whose value varies from 0 (bad) to 1 (excellent), and an STI of at least .5 is desirable for most applications.

STI value	Quality according to IEC 60268-16	Intelligibility of syllables in %	Intelligibility of words in %	Intelligibility of sentences in %
0 – 0.3	bad	0 – 34	0 – 67	0 – 89
0.3 – 0.45	poor	34 – 48	67 – 78	89 – 92
0.45 – 0.6	fair	48 – 67	78 – 87	92 – 95
0.6 – 0.75	good	67 – 90	87 – 94	95 – 96
0.75 – 1	excellent	90 – 96	94 – 96	96 – 100

***Figure 1:** STI measurements for native speakers*

Currently, there aren't many scenarios where you'll find yourself measuring STI in a conference room. However, since effective communication is the purpose of a conference room, awareness and measurement of STI can help the designer/integrator provide an optimal system design. In some cases, STI measurements may provide the necessary data to allow a designer to advise against the use of poor acoustic environments for conferencing (or at least indicate the need for acoustical treatments).

Microphone Polar Patterns

A polar pattern describes a microphone's inherent directionality. In more specific terms, polar patterns refer to signal attenuation at any given angular direction from the microphone relative to its central axis. Polar patterns can be divided into two groups: omnidirectional and unidirectional. Unidirectional microphones only pick up sound from primarily one direction at full level, with significantly attenuated signal pickup from all other (off-axis) directions. Unidirectional polar patterns include cardioid, supercardioid or hypercardioid patterns, as well as figure-eight patterns and lobar (shotgun) patterns.

Polar patterns are a top-down view of a microphone, with zero degrees typically marking the front of the microphone, 180 the back, with the left and right edges found at 270 and 90 degrees respectively. The outer circle of the plot indicates 0dB of attenuation, while each smaller concentric ring represents a 5dB reduction in audio sensitivity, and attenuation increases as the center of the circle is approached. For example, the attenuation of the cardioid mic in the figure below approaches 30dB at 180 degrees.

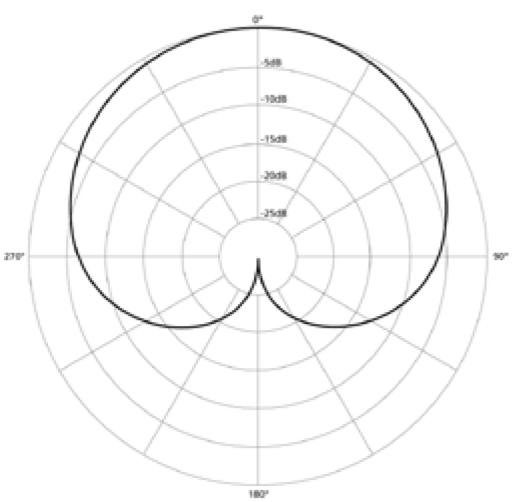


***Figure 2:** Mathematical representation of a cardioid polar pattern (top down view)*

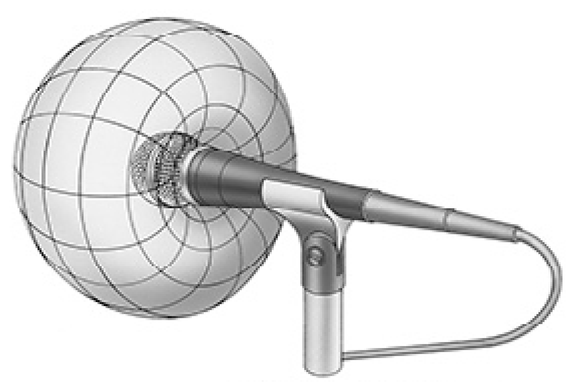


***Figure 3:** Cardioid polar pattern rendered in 3D*

Typically, as frequency decreases, unidirectional mics become more like omnidirectional mics, which is why unidirectional microphones are more prone to pick up low frequency noise from any angle[2]. The opposite is also true – as frequency increases, the patterns can become more directional[3].

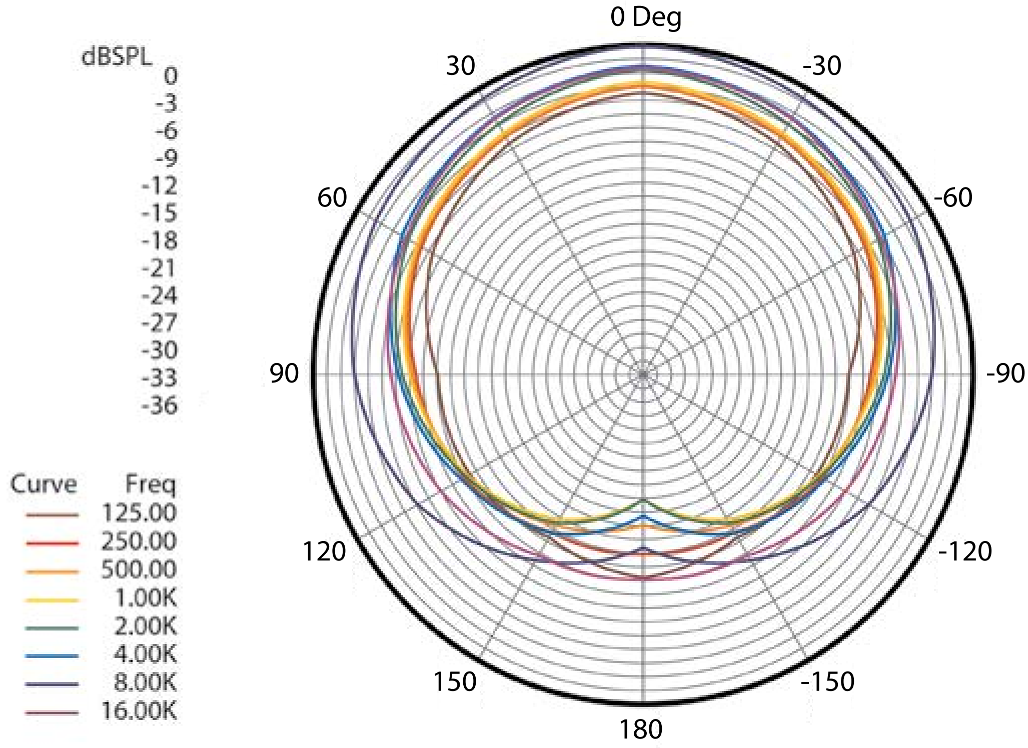
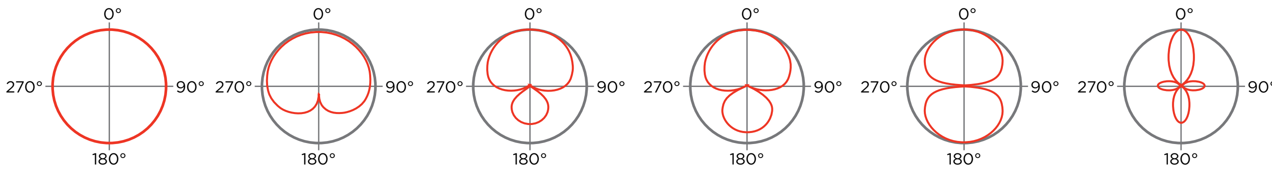


Figure 4: *Frequency response plot of Biamp's CM1-6W microphone*

It's important to note that there's no specific barrier or limit at which the audio pickup degrades or gates off (there is no measure of distance on the polar plots), although pickup can decrease rapidly when the talker moves off-axis. Also, polar patterns are actually three dimensional, so both the horizontal and vertical alignment of the mic with respect to the talker are important for allowing the talker to remain in the mic's "sweet spot".

As the polar patterns become increasingly directional (off-axis signals attenuate at increasing rates), starting with the cardioid and progressing to the supercardioid, hypercardioid, and finally the lobar (shotgun) pattern, the microphones must be aimed more precisely at the acoustic source. For example, microphones with a lobar polar pattern are excellent for isolating individual talkers, but are not suitable for applications in front of large sound sources such as choirs or orchestras due to their narrow pickup patterns.



Pattern	Omnidirectional	Cardioid	Supercardioid	Hypercardioid	Bidirectional	Lobar (Shotgun)
Acceptance Angle [4]	n/a	130°	115°	105°	90°	35°
Maximum Rejection	n/a	180°	125°	110°	90°	30°
Distance Factor [5]	1.0	1.7	1.9	2.0	1.7	3.0

Microphone Placement

Typically, designers consider speaker placement first for uniform coverage of a room (refer to the InfoComm and ANSI specifications). However, there's a strong argument that microphone or mic array placement is just as important, if not more so, for uniform mic coverage (although no standard exists yet).

Optimal beamforming microphone placement is guided by the typical seating arrangement within the room. Consider these suggestions when determining mic installation for the best possible results:

- In rooms with flexible furniture arrangements or multiple beamforming microphones, map out all of the possible seating scenarios to ensure that coverage will be adequate for each.
- Lobes should be pointed towards the front of each talker (think of a lobe as a spotlight on the face of a performer). Carefully consider placement in rooms where talkers may face a screen during a video conference.
- Identify the maximum recommended distance from the beamforming microphone to a talker before designing the room.
- Avoid installing the microphone directly next to unwanted sound sources, such as air vents or video projectors.
- Consider installing acoustic treatments to improve speech intelligibility in rooms that are too reverberant.
- The pickup pattern of a beamforming array can be narrower than a shotgun microphone, and therefore it can sometimes be placed farther from the source than other types of microphones.
- Keep in mind that there is no specific barrier at which the pickup audio degrades or gates off.
- Like all microphones, tonality changes as the distance from the source increases.
- The lobe coverage area increases as distance increases.

Aiming Lobes

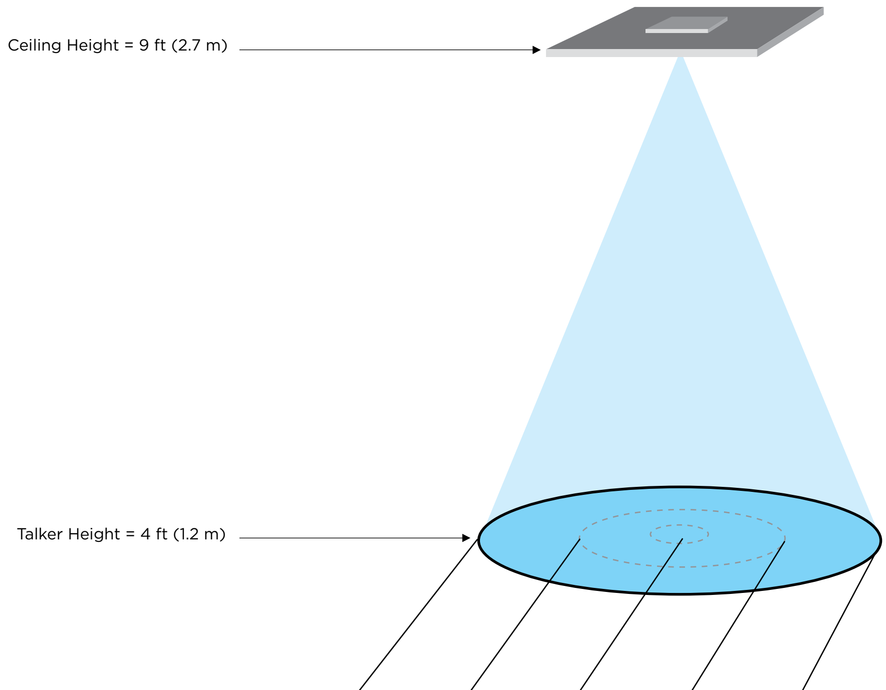
Think of each lobe as an individual virtual microphone. If there were eight microphones on the table, each one could be physically placed according to seating arrangements, with independent gain and channel controls. Independent width control makes it possible for some lobes to capture individual talkers (narrow), while other lobes cover multiple talkers (wide). A note of caution: even though the above analogy makes it appear that beamforming microphones can be used for voice lift/local reinforcement applications, it is not recommended.

Consider these suggestions when aiming lobes for the best possible results:

- Use speech as the sound source. Do not use test tones.
- Eliminate noise sources in the room if possible, such as projectors or fans.
- For the best acoustics, talkers should not be directly against a wall or in a corner, as acoustic reflections can reduce aiming accuracy.
- Speak from the exact location where the lobe should be positioned.
- Each lobe can be used to pick up one or more talkers, so the concept of “less is more” should be kept in mind.
- If a lobe needs to cover an area with multiple talkers, talk from the center of the region. Manual adjustments may be necessary after the lobe is positioned to ensure the entire area is covered.
- Any unnecessary lobes should be deleted.
- As the lobe width is set tighter, side lobes become more prominent. In some applications, it may be better to use medium or wide lobes depending on the desired results.
- Provide complete coverage in a space, either by adding lobes or changing the lobe width. This ensures the sensitivity is within 6 dB in all areas.
- Ensure that spacing and isolation are adequate to reduce noise and maximize automatic mixing performance. However, it is acceptable for lobes to slightly overlap.

Here are some additional reference measurements that may aid in aiming or confirming aiming of microphone lobes:

- The average conference room table is 30” Above Finished Floor (AFF).
- The average height of a talker’s mouth is 48” AFF when in the seated position.
- The average height of a talker’s mouth is 60-66” AFF when in the standing position



Narrow					
Centimeters	44	30	0	30	44
Inches	17	12	0	12	17
dB	**-6**	**-3**	0	**-3**	**-6**
Medium					
Centimeters	47	38	0	38	47
Inches	18	15	0	15	18
dB	**-6**	**-3**	0	**-3**	**-6**
Wide					
Centimeters	62	44	0	44	62
Inches	24	17	0	17	24
dB	**-6**	**-3**	0	**-3**	**-6**

Figure 6: *Approximate widths of narrow (35°), medium (45°), and wide lobes (55°). Note that down-firing lobes are not effective, as the talker will be speaking approximately 90° off axis. Most lobes will be more accurately represented as an ellipse as opposed to a circle.*

Use Cases for Lobe Aiming

For optimal sound pickup, aim lobes where talkers are likely to be, and exclude areas of high ambient noise or sources of reverberation. However, as with most things in AV, there are tradeoffs to consider. For instance, if there's a whiteboard in the room, should the lobe be aimed where a person's sitting, aim a second lobe at the whiteboard, or use a wider lobe to cover both areas?

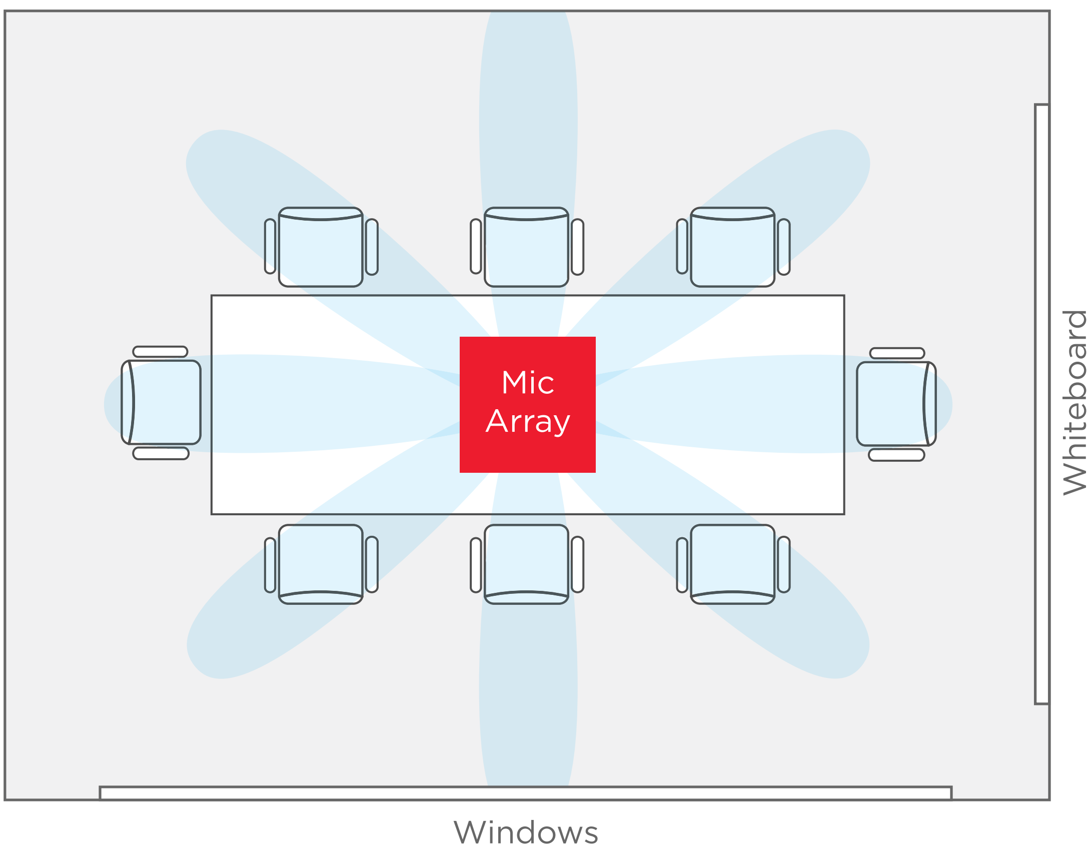
There's also a common misconception in the pro AV industry that you should always use as many lobes as possible. Yes, there are certainly scenarios where using all available lobes is desirable, but there are also instances where using fewer lobes may result in broader and more consistent coverage, as we'll see in the following use cases.

For these use cases, our example beamforming array will support up to 8 separate lobes.

Scenario 1: Conference room with centrally located table and mic array

Unimproved solution

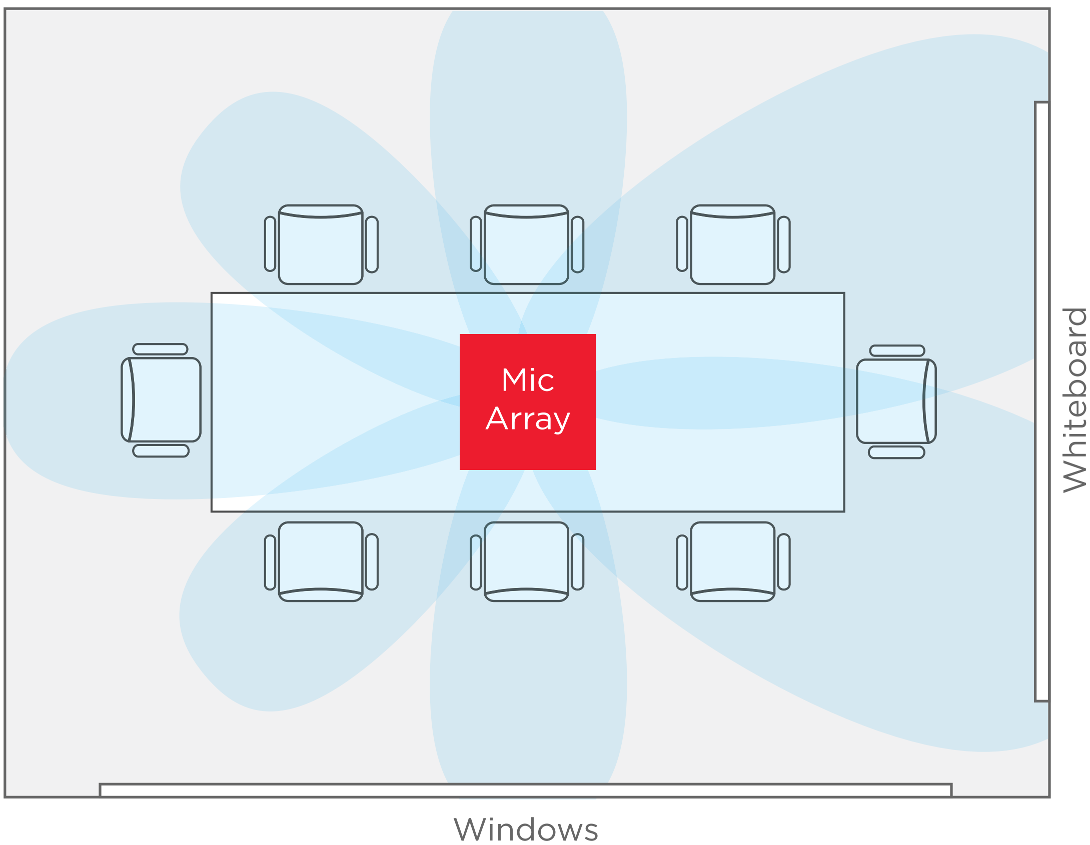
The center of attention is the center of the table – meeting participants will primarily be looking at one another when speaking. Using eight narrow lobes, one per chair, leaves a lot of space in the room without good microphone coverage. The interstices between the beams (the gray areas) are the parts of the room where speech will be attenuated by 12dB or more. This potentially poses a problem if the chairs are on wheels, or meeting attendees move back and forth from the whiteboard.



Improved solution

One possible solution is to vary the beam widths (many beamforming microphones offer a narrow, medium, or wide lobe pattern) to cover the gaps. One common solution is a three- or four-leaf clover pattern.

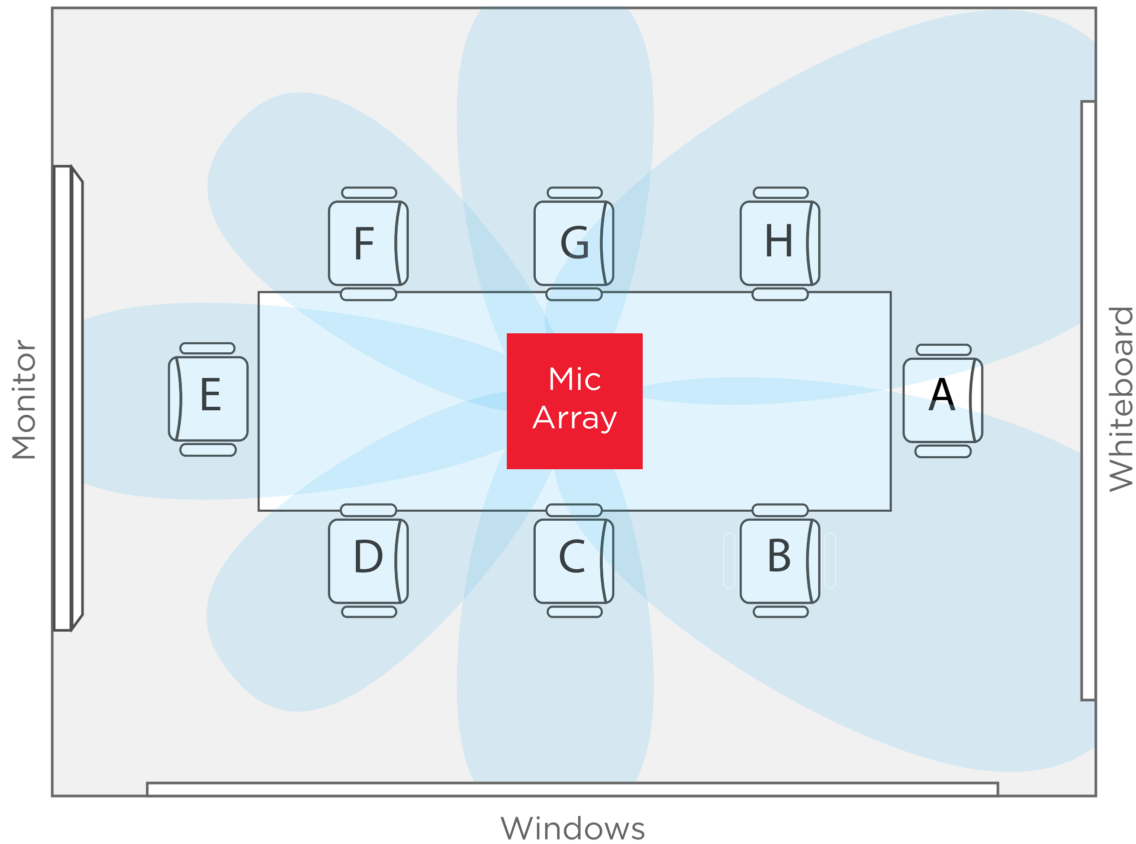
Another issue, though, is lobe overlap. Typically, the less off-axis noise or unwanted sound, the better. Some overlap is acceptable, but if there's too much overlap, you may hear switching between the lobes or the person's timbre change. Also, when the lobes are summed, the duplicate speech may not be eliminated entirely, and comb filtering may result.



Scenario 2: Conference room primarily used for videoconferencing

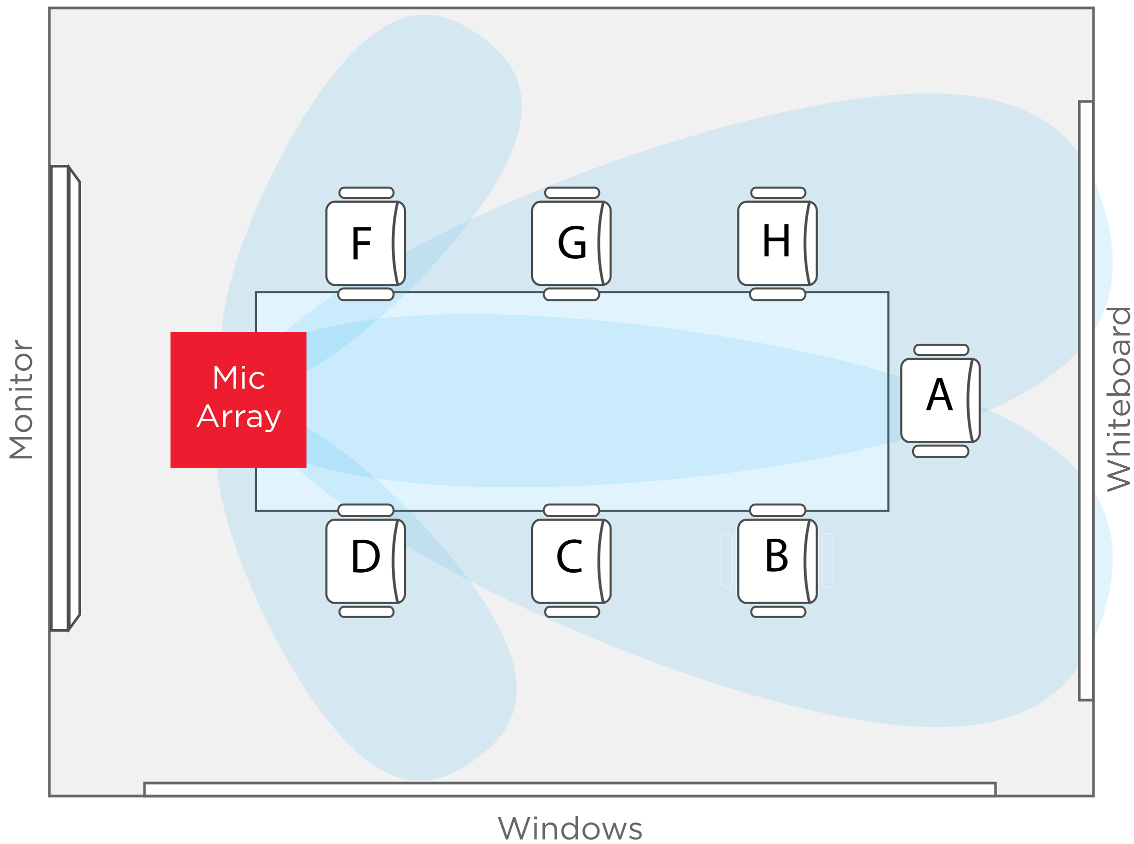
Unimproved solution

In this scenario, the center of attention is the monitor – meeting participants will primarily be looking at it when speaking. If a talker has their back to the beamforming mic, their speech will be greatly attenuated (participants C, D, F, and G will be 90° or more off-axis). No one is likely to sit in the chair closest to the monitor, so that particular lobe is not needed.



Improved solution

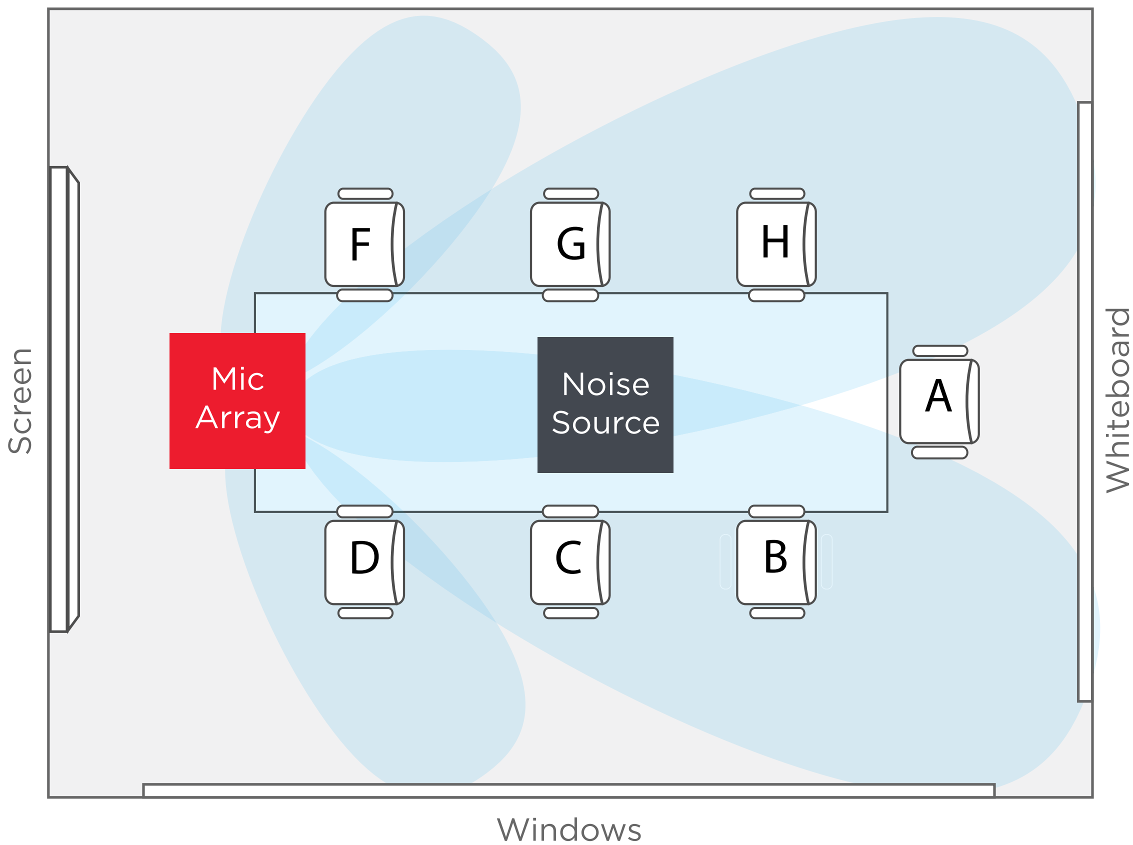
If possible, move the microphone towards the monitor (keep in mind the signal-to-noise ratio (SNR) increases the further the mic is from the talker). This allows the lobes to be pointed at all of the potential talkers. Most people remain seated during a video conference, so narrow lobes may be adequate, but use of the whiteboard needs to be considered.



Scenario 3: Conference room with a mechanical noise source

Unimproved solution

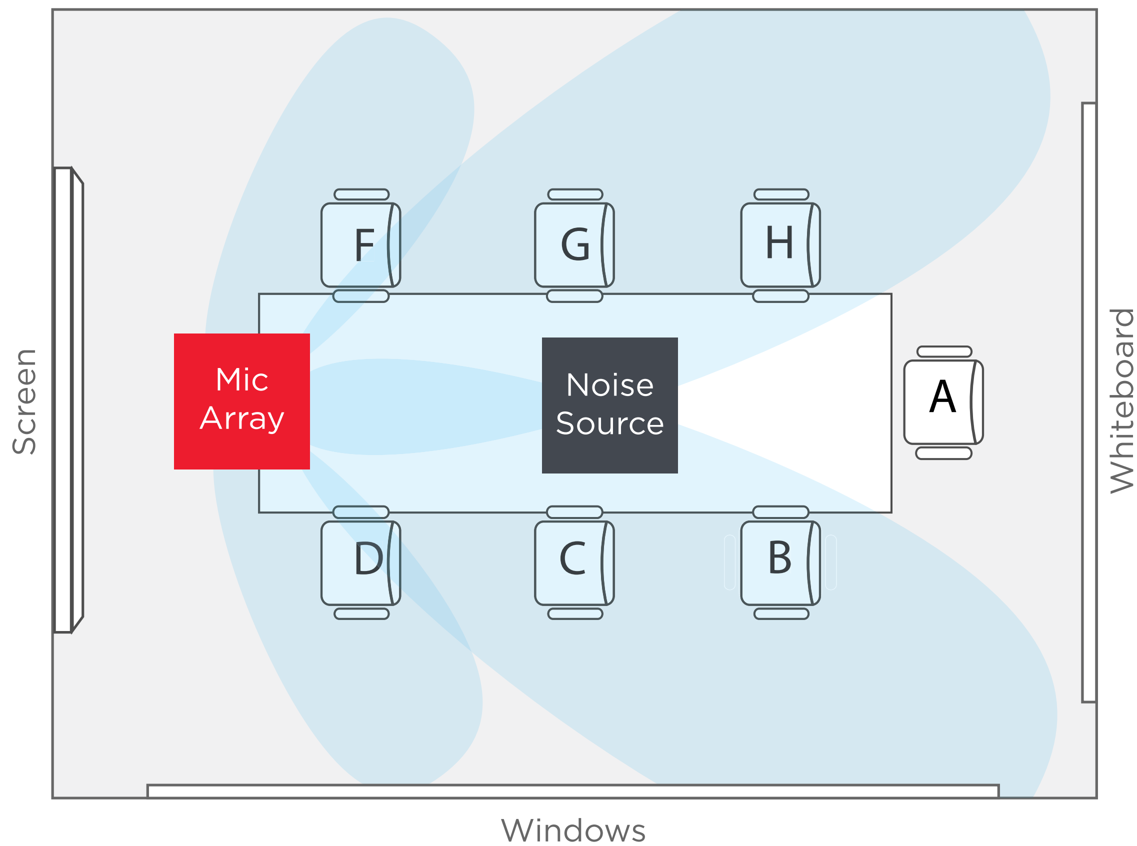
Electrical equipment in the room such as a PC or AV gear is likely to emit a fair amount of fan noise and other unwanted sounds. The constant nature of the noise means that at least one lobe will remain active regardless of who's talking, so anyone listening will hear both the talker and the noise, resulting in a suboptimal experience. Also, the narrower the shotgun polar pattern, the more pronounced the side and back lobes become. For ceiling microphones, you can potentially pick up mechanical noise (primarily HVAC) from above the ceiling tile.



Improved solution

Eliminating the noise source entirely is ideal. That's not always an option, though, such as in the case where the noise source is the HVAC system. Physically separating the microphone from the noise source is the next step. If the audio is still unacceptable, place the noise source off-axis between two lobes. That will greatly attenuate the unwanted sound. To minimize mechanical noise from above the ceiling, apply an acoustically absorbent material on top of the microphone. HVAC noise increases over time, so the best the system will sound is what it sounds like today – set your customer's expectations accordingly. HVAC also introduces vibration noise across the ceiling tiles, so applying material between the microphone and the drop ceiling to absorb and minimize the vibrations is also recommended.

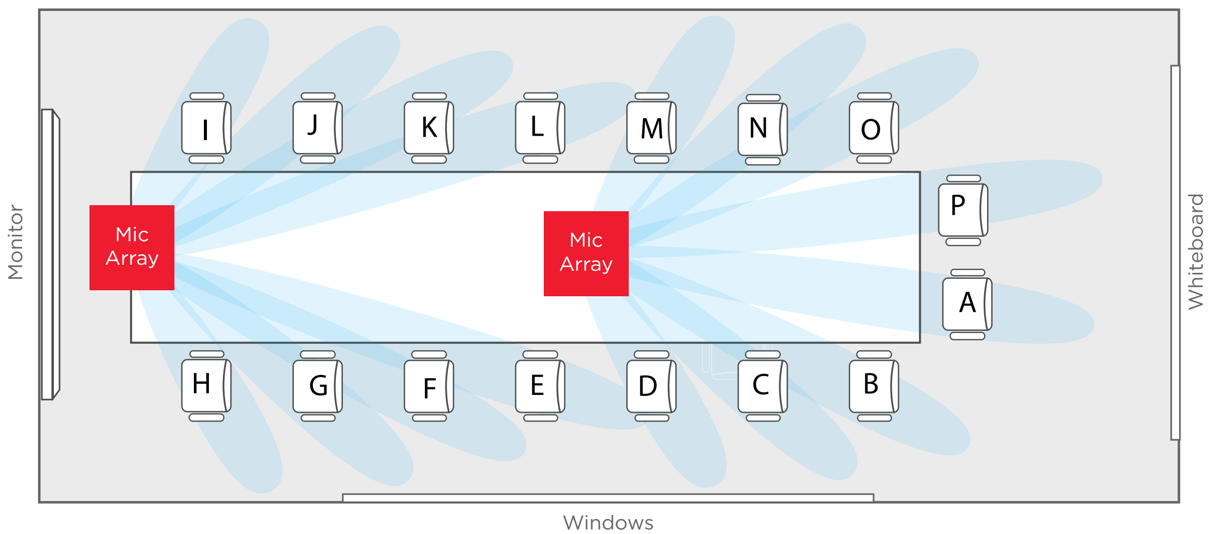
If the room is used for videoconferencing and a ceiling projector is being used, you may need to install a second microphone for Chair A since that seat may be outside of lobe coverage.



Scenario 4: Large multi-use space requiring two or more beamforming mics

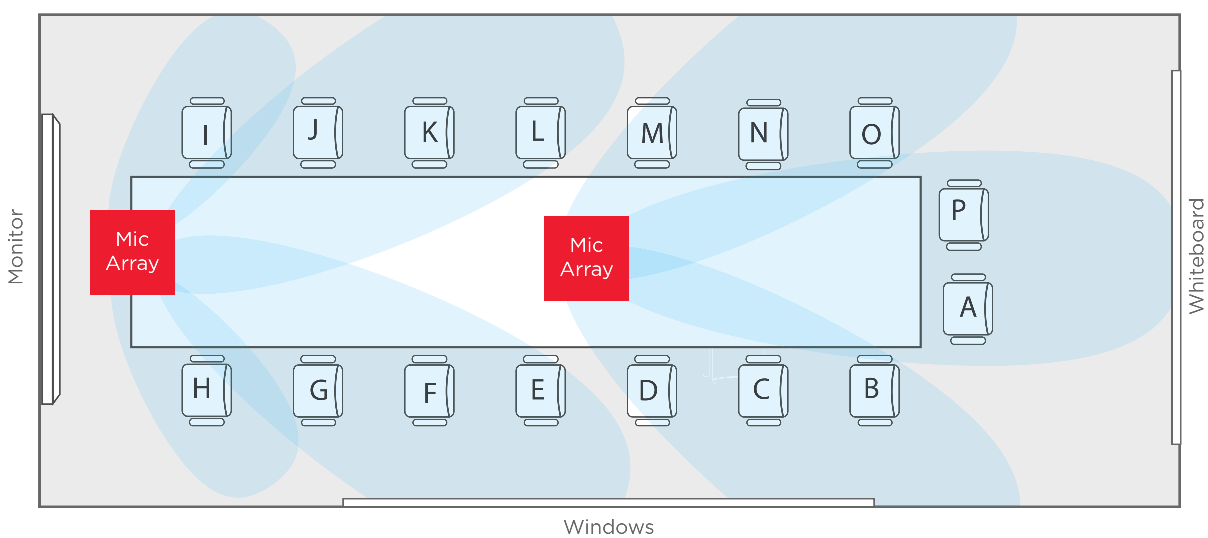
Unimproved solution

This scenario suffers from some of the same problems as the first scenario, only with more lobes. Using narrow lobes, one per chair, leaves a lot of space in the room without good microphone coverage. The interstices between the beams (the gray areas) are the parts of the room where, essentially, speech will be attenuated. The significant number of lobes also increases the chances of a comb filter occurring if a talker is picked up by more than one lobe.



Improved solution

Lobes should only be aimed where talkers are likely to be. In the case of an executive boardroom with a large table, there's no value in pointing lobes from each mic towards the other, since there will never be a talker in that area. Another consideration is whether the room will hold more participants than chairs (it will happen at some point). Narrow lobes will not pick up the additional attendees sitting/standing around the perimeter of the room.



Key Takeaways

Beamforming microphones are becoming increasingly popular for conferencing applications - beamforming ceiling microphones in particular– due in part to aesthetics and the trend of reducing/removing technology from the tabletop. However, using more lobes doesn't always equate to improved speech intelligibility or consistency of coverage. Using too few lobes isn't ideal either – you'll pick up a lot of unwanted signals and noise. In spaces where talkers move around a lot, then a handful of wide lobes may be more suitable than a bevy of narrow lobes.

Consider these suggestions when designing lobes for the best possible results:

- Aim lobes at the areas where the talkers are likely to be located
- Don't automatically assign each chair in the room its own lobe – sometimes that's ideal but sometimes it isn't
- Understand the primary use case for the room, then use the fewest possible number of lobes to cover the desired area while also maintaining an acceptable STI (typically 0.66 or higher)
- Don't be afraid to use Medium or Wide lobes – instead, start with them, and move to narrow lobes when needed
- Delete any unused lobes
- Use presets to reconfigure the lobes based on how the room is being used (size of the meeting, reconfigurable walls, etc.)
- Listen and adjust as needed to achieve optimal sound quality
- Consult with the microphone and DSP manufacturers' tech support during the design phase for tips and tricks to optimize the audio quality

Additional Resources

AP-112 Acoustical Ambient Noise (pages 18-23)

AV Design Reference Manual (chapter 11)

AV Systems Performance Verification Checklist

Comb Filters

Microphone Placement

Microphone Sensitivity

[1] Internal testing indicates that room architecture is the most significant factor affecting speech intelligibility, but your results may vary.

[2] In a multiple microphone scenario such as a beamforming microphone, low frequency noise tends to result in a cumulative effect – upwards of 4-5dB in some cases - which is undesirable.

[3] A challenge with high frequency noise in a multiple microphone situation is audio phase shifting between the pickup lobes, producing a comb filter (also undesirable).

[4] Acceptance Angle is the angle from the axis before reaching 3dB in attenuation

[5] Distance Factor refers to the relative distanced a unidirectional microphone can be placed relative to an omnidirectional microphone and produce the same audio quality.