Optimize your conferences with microphone array beamforming

EXPAND 80 & EXPAND 80 Mic

Introducing the EXPAND 80 Series

Quality conferencing
Conference calls are a more sustainable approach to collaborating with colleagues across geographic locations. This means that the quality and reliability of conference solutions is vital to support productive virtual meetings. EPOS is at the forefront of creating audio technologies that release the potential of today's workforce. Our advances in microphone pickup technology enable EPOS conferencing tools such as the EXPAND 80 Series to boost the quality and clarity of communication.

EXPAND 80 beamforming microphone arrays
EXPAND 80 is a speakerphone series for medium to large-sized meeting rooms*. The series is composed of the EXPAND 80 speakerphone and the EXPAND 80 Mic extension allowing the speakerphone to be used in even larger meeting rooms. Both are designed with beamforming microphone arrays. These arrays use multiple microphones and algorithmic signal processing in order to capture and convey speech to remote participants with the greatest intelligibility.

What do ideal speakerphones do?
A meeting room speakerphone has two primary functions. It must broadcast remote speech through a loudspeaker, and capture speech in the meeting room while conveying it to remote participants. Ideally, the speakerphone must convey speech as intelligibly as possible. Such speakerphones use "echo cancellation" to prevent sound from the loudspeaker, broadcast in the meeting room, from returning to the remote listener through the speakerphone's own microphone. Quality speakerphones also use microphone array beamforming to further isolate voices from both ambient noise and reverberation, resulting in enhanced speech intelligibility for remote participants.

*Small meeting room – 2-6 | Medium meeting room – 6-12 | Large meeting rooms – 12+

Noise and reverberation

In many meeting room scenarios, the presence of background noise and reverberation can interfere with the sound of people talking, which leads to a reduction in speech intelligibility.

Noise

Speech becomes more difficult to understand in a noisy environment, as represented in the waveform in figure 1.0 (a). So, it is essential to find a way to ensure that the ratio of speech to noise received by any remote listener is weighted in favor of speech, as in the waveform in figure 1.0 (b).

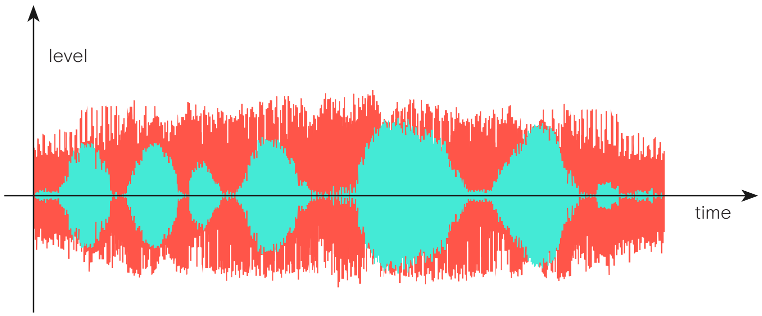


Figure 1.0 (a): unintelligible speech. The speech signal ( ) drowned in noise ( ).

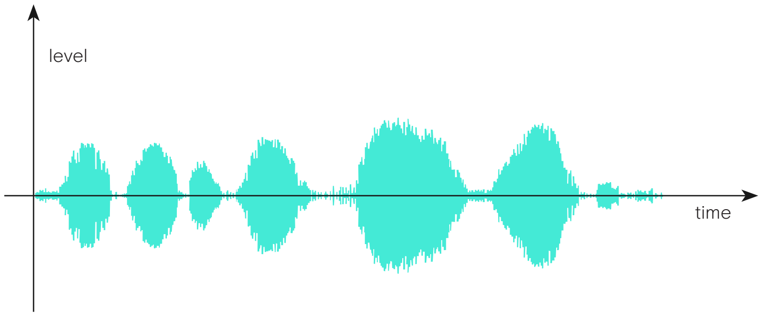


Figure 1.0 (b): clean speech.

Noise and reverberation

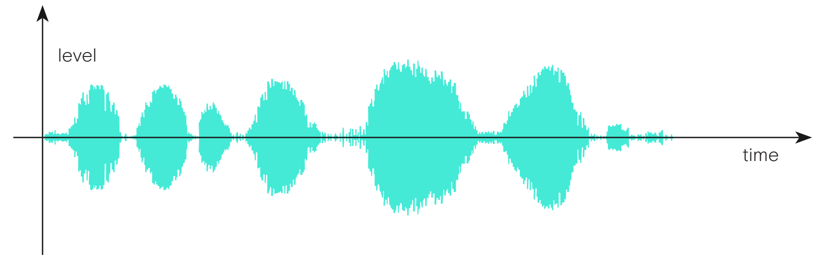


Figure 1.2 (a): Direct signal.

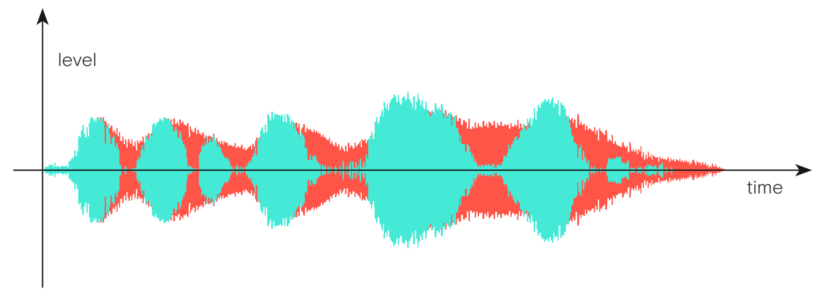


Figure 1.2 (b): Direct signal ( ) & reverberation signals ( ).

Reverberation
A microphone picking up speech in a room will first pick up the voice – a sound which initially arrives at that microphone directly from the person speaking. This microphone will then pick up reverberation as a series of additional sounds from the voice being reflected from the room's walls, ceiling and floor.

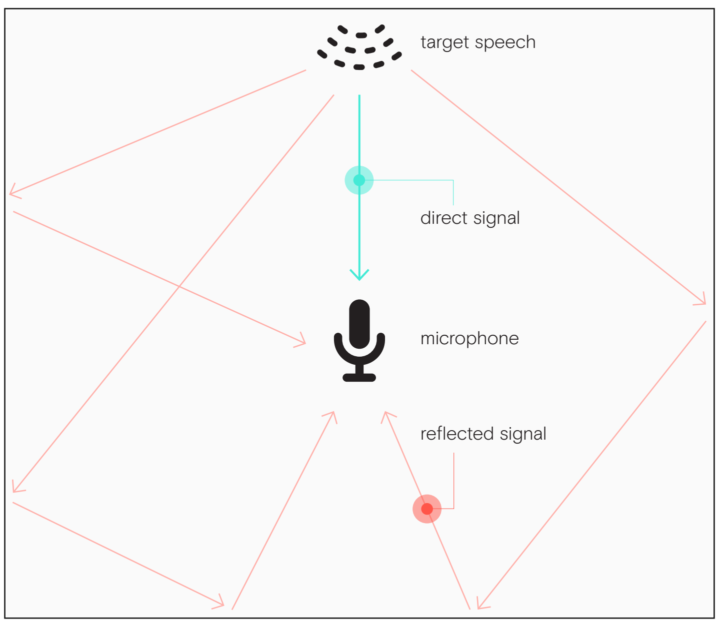


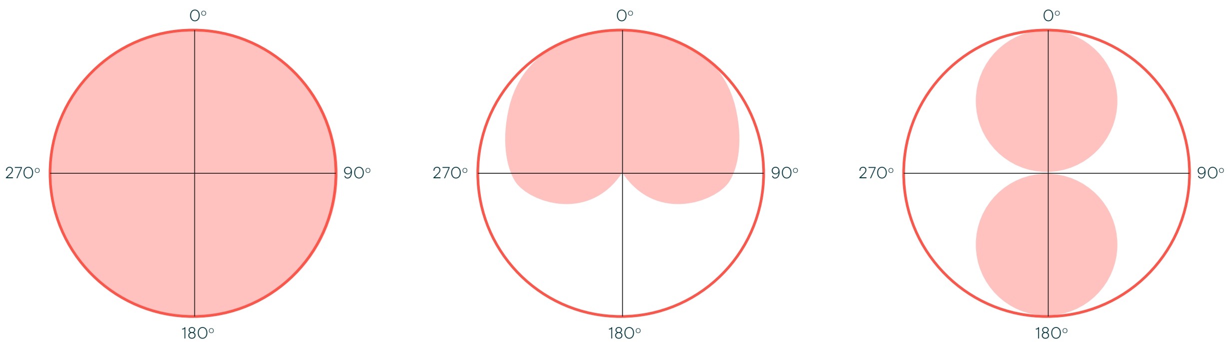
Figure 1.1: Direct & reflected signals arriving at microphone.

The differences between direct and reflected signals arriving at a microphone are a question of time and strength – reflected sounds arrive later than direct sounds (at a different 'phase'), and they arrive with less energy (or 'amplitude'). When a microphone receives direct and reflected sound at a similar level, the effect is a 'blurring' of the signal conveyed to the remote listener. This affects the intelligibility of speech. For many meeting rooms, this is perceived by the remote listener as if the speaker were standing in a bathroom, for example. Figure 1.2(a) represents a signal arriving directly at the microphone from the source, figure 1.2(b) represents the same signal and a comparison of how it is conveyed by the microphone with the source sound reverberating in the room.

Directional microphone systems

An overview of directivity
To overcome noise and reverberation in a typical meeting room, the ideal speakerphone would be more sensitive to the direction of speech than to the direction of the sources of noise and of reverberation. Such a microphone system can be described as a **directional microphone system**.

In general, all microphones are designed with a certain 'pickup pattern', determining how sensitive a microphone is to sound arriving from any specific direction. These vary between omnidirectional patterns (equally sensitive to sound from any direction) to bi-directional (sensitive to sounds from two directions), as shown in figure 2.0. Of these pickup patterns, the simplest and most commonly available is the omnidirectional microphone.



1. Omnidirectional (circular polar pattern)

2. Unidirectional (cardioid polar pattern)

3. Bidirectional (figure-of-eight polar pattern)

Figure 2.0: Omnidirectional, unidirectional (cardioid) & bidirectional (figure of eight) microphone pickup patterns.

Directional microphone systems

Producing directional pickup from omnidirectional microphones
It is possible to produce a focused and directional pickup of sound from a series of omnidirectional microphones. This can be achieved by taking advantage of differences in both the level of sound and the time at which it arrives at the different microphones. A directional microphone system such as this is known as a microphone array beamformer. To illustrate this concept, figure 2.1 shows a typical example known as a 'Delay and Sum beamformer'.

Typical 'Delay and Sum' beamformer
In figure 2.1 sound arrives at the microphone array from an angle. On account of its angled arrival, the sound reaches the array's microphones at different times. These differences in time are determined by the amount of distance between the microphones.
By introducing specific delays to each microphone, it is possible to align the signals in such a way so as to synchronize them for a certain direction of sound arrival. Subsequent summation of these signals increases the output level of the microphone array for a certain direction, while decreasing the output level for other directions, through a process known as 'interference'. By adjusting these delays, it is even possible to virtually 'steer' the array to 'focus' on sound arriving from any specific direction. The geometry of the array and the precise amount of delay must be carefully designed if the system is to function accurately and flexibly.

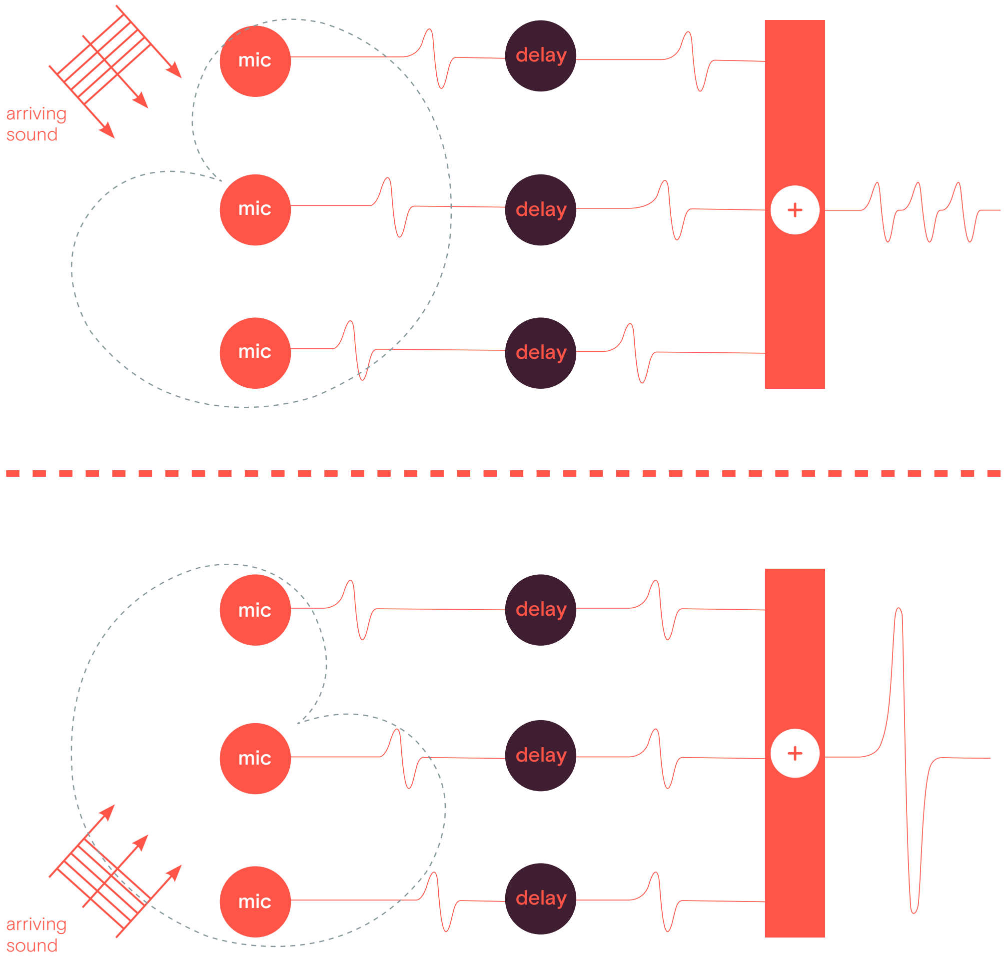


Figure 2.1: 'Delay and Sum' beamformer system.

Advanced beamforming in EXPAND 80

Microphone array beamforming
In the EXPAND 80 speakerphone, six low-noise digital MEMS microphones are placed in a specific configuration optimized to enable speech to be picked up from any angle. Without signal processing, the EXPAND 80 would work as an omnidirectional microphone. When all six microphones are engaged and the advanced signal processing algorithms are applied, however, the directional pattern is focused into a tight beam.

Adaptive beam steering
A focused, steerable beam like this is useful for optimizing the target sound while simultaneously rejecting sound coming from other directions, allowing the pickup of speech from any angle desired. The system is capable of analyzing the content in all directions and automatically selecting the direction of interest. In figure 3.1, it can be seen that, even if the target speech signal changes position, (when two different people are talking in a meeting room, for example), EXPAND 80 will automatically steer a focused beam to the desired direction for the target signal.

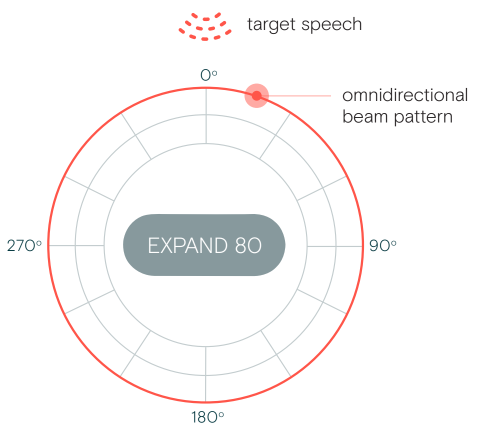


Figure 3.0 (a): Omnidirectional pickup pattern.

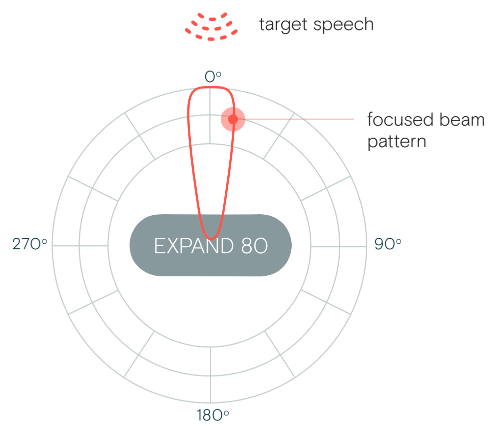


Figure 3.0 (b): Focused beam pickup pattern.

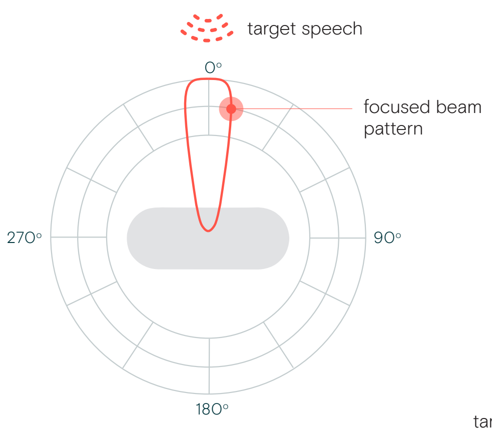


Figure 3.1 (a): Focused beam steered to 0°.

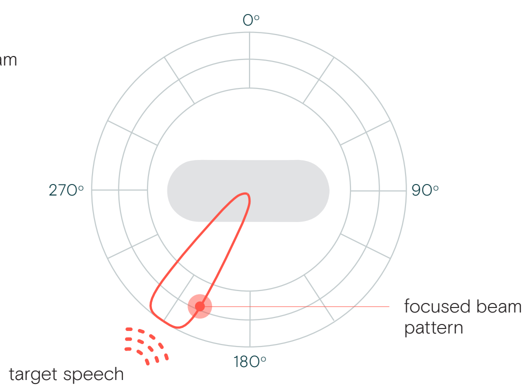


Figure 3.1 (b): Focused beam steered to 210°.

Advanced beamforming in EXPAND 80

Attenuation of ambient noise

An omnidirectional microphone will pick up target speech as well as unwanted surrounding noise sources equally – see figure 3.2 (a). However, this is not the case when advanced beamforming is used. In the EXPAND 80, speech arriving from the direction that the beam is pointing in will be picked up without any change, as compared with an omnidirectional microphone. Sounds arriving from other angles, such as noise and reverberation, will be greatly attenuated – that is, reduced.

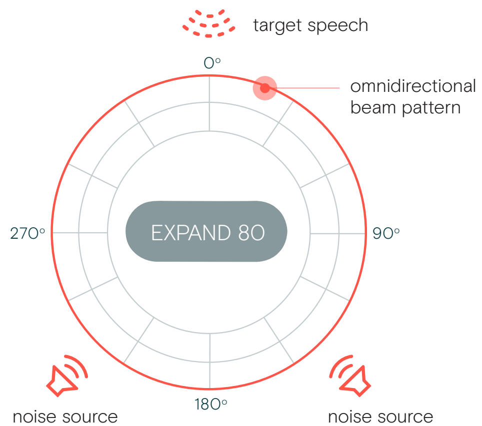


Figure 3.2 (a): Omnidirectional pickup pattern.

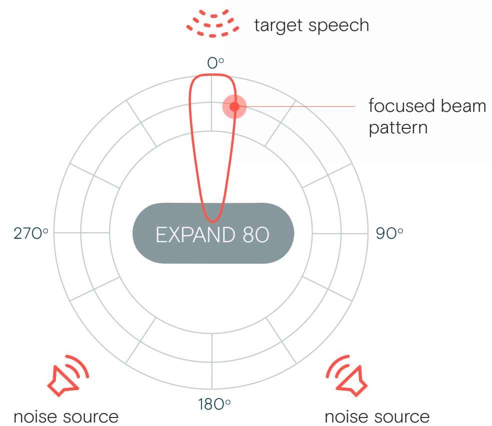


Figure 3.2 (b): Focused beam pickup pattern.

Advanced beamforming concept

Active de-reverberation
As we have seen, reverberation causes sounds to arrive at a speakerphone with additional delay and from additional angles. This results in a blurring of the signal in time, reducing the intelligibility of speech.
An enhanced ratio of speech to reverberated sound is maintained through the use of a beam focused in the direction of the target signal. Sounds arriving at an angle, reflected from the room's surfaces, will be conveyed with attenuation, in comparison to the target signal:

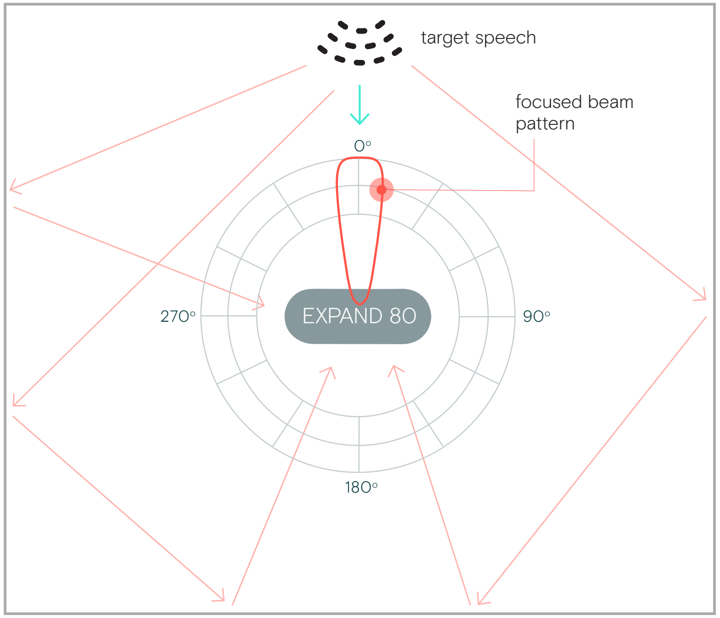


Figure 3.3 (a): Focused beam singles out the target speech and rejects the speech reflections from room surfaces.

Functional multiple-array system

EXPAND 80 is designed to provide quality speech pickup in medium to large-sized meeting rooms. For even larger meeting rooms, the EXPAND 80 can be augmented with up to two additional EXPAND 80 Mic units, all operating with identical microphone array technology. When connected, larger meeting spaces can be covered by up to three microphone arrays working as a single network. Speech will be most effectively picked up by the most appropriate of the total beams available across all the connected arrays. This configuration is demonstrated opposite for an EXPAND 80 and two EXPAND 80 Mic units in figure 4.0.

EPOS enables closer collaboration

Thanks to microphone array beamforming in the EXPAND 80 Series, a focused, steerable beam can optimize individual voices in the meeting room for remote listeners. Through this advanced EPOS technology, decentralized teams are empowered to collaborate with the same confidence and clarity as if being there in person.

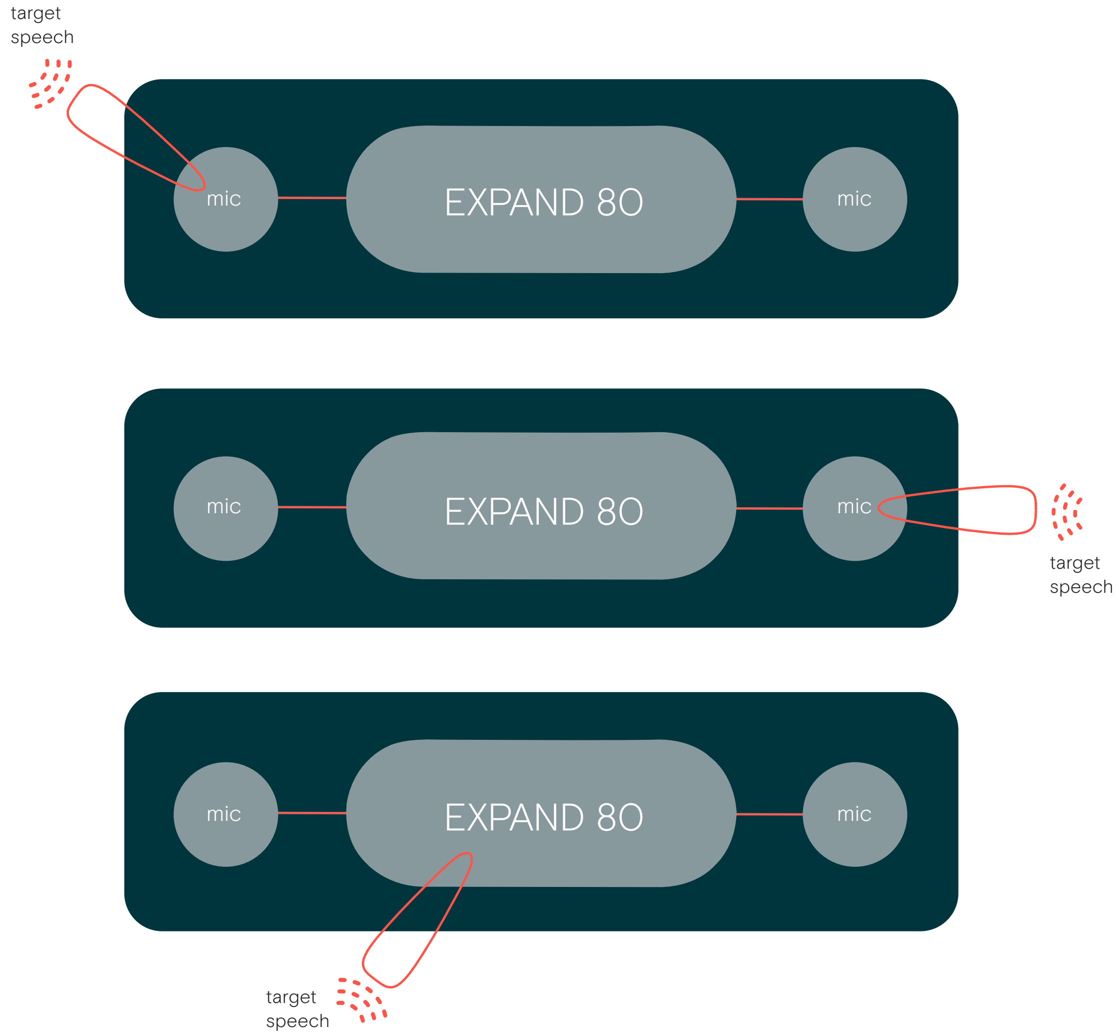


Figure 4.0: Multi-array operation in a large meeting space.

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