TeamConnect Ceiling 2

TruVoicelift.
Sennheiser's Interpretation of Voice Lift.

Sennheiser electronic GmbH & Co. KG

SENNHEISER

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Content

Introduction

TeamConnect Ceiling 2 is the leading microphone solution for video and web conferencing around the world. Unlike other products on the market, this powerful microphone offers easy installation and integration, along with the most flexible beamforming technology and advanced zone control. The new TruVoicelift feature provides an unmatched natural speech experience with outstanding speech intelligibility. Your voice can be heard throughout the room, all the way to the back row of a classroom, lecture hall, large boardroom and more.

This document describes what voice lift means in a typical audio-visual (AV) application and the challenges an installer might face during the installation. TeamConnect Ceiling 2 with TruVoicelift technology is surprisingly easy to set up and configure.

Features of the TeamConnect Ceiling 2

Automatic Beamforming Technology

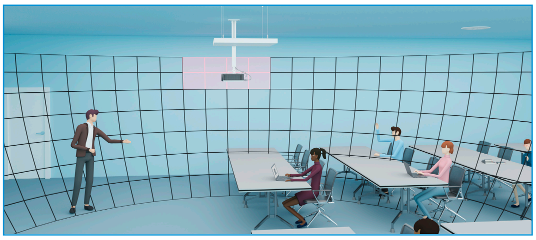
The revolutionary beamforming technology picks up all audio signals in the meeting room. The unit uses digital signal processing to locate the person speaking at all times, regardless of whether they are sitting, standing or moving around. Voices are picked up evenly throughout the entire room.

TruVoicelift

This feature allows the person speaking to move about freely without worrying about the technology, while providing perfect intelligibility for all participants and minimizing feedback issues. It significantly amplifies speech within the room, and achieves louder volumes than other solutions on the market.

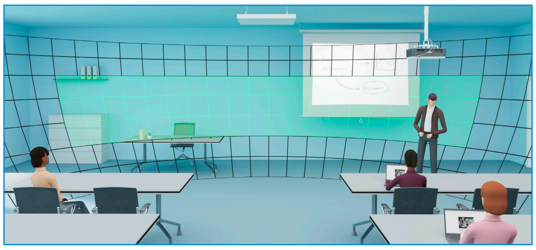
Advanced Exclusion Zones

With up to five advanced Exclusion Zones, the customer has full flexibility and can precisely define the position (both vertical and horizontal) of any noise sources to be ignored by the beam tracking. These zones are visualized in the powerful new 3D view in the Sennheiser Control Cockpit.



Priority Zone

Take full control of your speech by setting a Priority Zone for the microphone beam to focus on. This zone will be prioritized if audio signals are received from different locations in the room at the same time.



Voice Lift in General

The voice lift feature was introduced to evenly distribute the presenter's natural speech volume, even to listeners at great distances. Studies have shown that a voice lift is typically needed at distances of 8m (26ft) and more from the presenter. Voice lift can also be beneficial in smaller rooms in situations where the presenter is soft-spoken (or speaks predominantly in low pitches). The goal is to achieve speech intelligibility for all participants in a room.

Difference to Sound Reinforcement (SR)

Often, traditional AV conference rooms are equipped with sound reinforcement (SR) systems (SR) or public address (PA) systems. Although these systems focus on audio transmission, they may not be suitable for meeting or presentation applications.

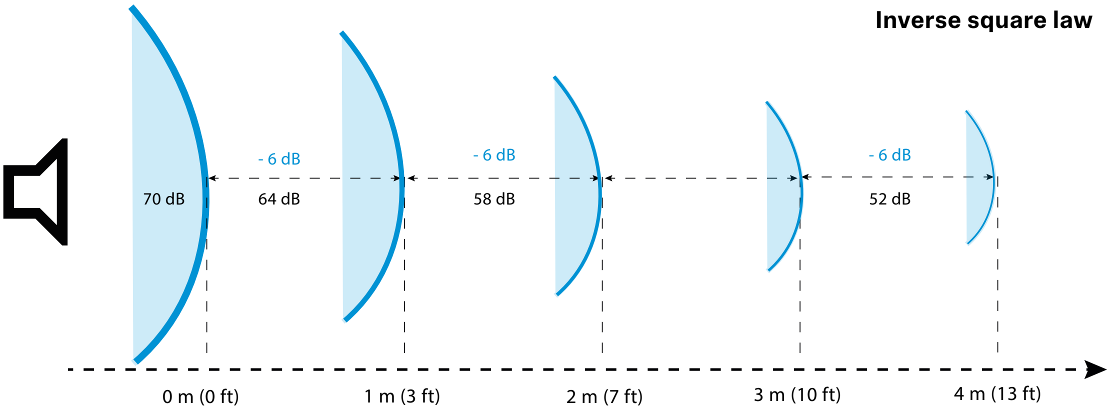
PA/SR System	Voice Lift System
With a conventional PA/SR system, all signals received in the room are boosted to the point where even the most distant presenter can be understood. To achieve a sufficient gain before feedback (GBF), the microphones must be worn very close to the body. And the front-of-stage loudspeakers have to be quite loud to reach the far end of the room, which leads to different volume levels in the room. During presentations in front of a large audience, communication is usually only possible in one direction, and direct interaction between participants is very challenging.	A voice lift system keeps speech at a constant level no matter the distances in the room—so that voices appear as clear, close and natural as possible without a hint of artificial tone or feedback. Voice lift systems amplify audio signals individually while increasing the speech level by 3 to 6dB above the noise floor. Microphones and loudspeakers are divided into different zones, and their levels can be controlled by an external DSP (digital signal processing) system.

Technical Challenges

Despite key advantages over a PA/SR system, there are technical factors to consider when designing a typical voice lift application (see „Checklist for Voice Lift Design"). These special requirements are described in detail below. They illustrate the complexity that an intelligent voice lift system must take into account in order to respond well to everyday challenges.

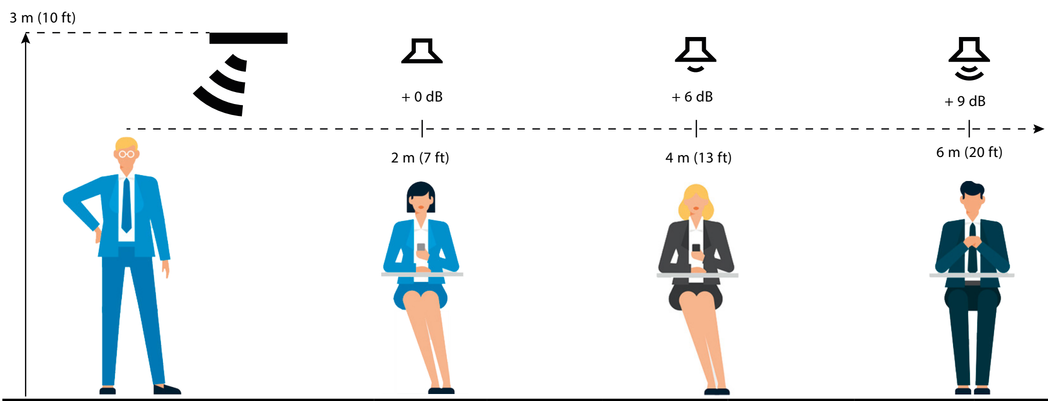
Distribution of Sound

The inverse square law (ISL) states that every doubling of distance from the sound source reduces the sound pressure level by 6dB. This means that the intensity of sound energy decreases with the square of the distance from the sound source when distributed over a large area. Based on ISL, most voice lift applications are designed for rooms of 10 x 10m (33 x 33ft) or larger. The distance from the presenter to the farthest listener should be at least 8m (26ft). A voice lift may also be required in smaller rooms, depending on the room acoustics, ambient noise and the presenter's speaking volume.



Calculation example

Given a natural speech level of 70dB, different listeners will receive different levels depending on their location. Every doubling of the distance means a loss of 6dB. While no additional gain is required at a distance of 1m because the speech intelligibility is high, at a distance of 8m (26ft) the direct sound will have a level of approximately 46dB with a delay of 23ms. This loss of sound energy must be compensated by a supplemental voice lift.



Distance to person talking	0 m (0 ft)	1 m (3 ft)	2 m (7 ft)	4 m (13 ft)	6 m (20 ft)	8 m (26 ft)	10 m (33ft)
Level attenuation	70 dB	64 dB	58 dB	52 dB	49 dB	46 dB	43 dB
Gain to be added	0	0	0	+ 6 dB	+ 9 dB	+ 12 dB	+ 15 dB
Approx. delay to be added	0	2.9 ms	5.8 ms	11.7 ms	17.5 ms	23 ms	30 ms

Distribution of Sound with TeamConnect Ceiling 2

Zone Control

You can ensure even distribution of sound with no feedback by splitting the loudspeakers and microphones into different zones within the room. The quantity and locations of the zones will be determined by the size and acoustics of the room, the number of listeners, the distance between the microphones and the loudspeakers, and the polar pattern of the chosen loudspeaker. To determine how loud your system needs to be for even coverage and speech intelligibility in the room, you first need to calculate the potential acoustic gain (PAG) and the needed acoustic gain (NAG) (see „PAG/NAG Calculation").

Presenter sound control

Example:

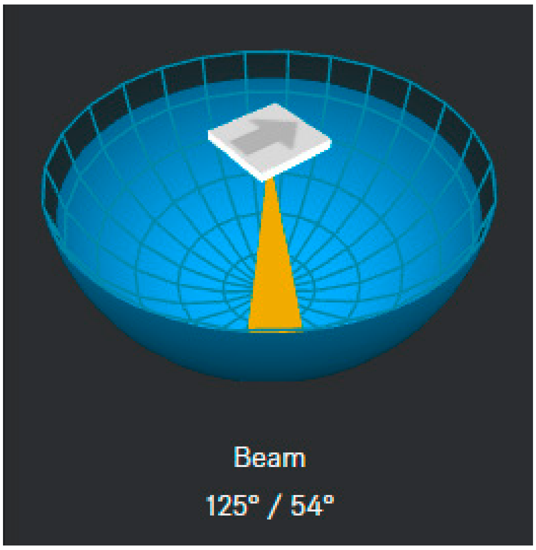
If the presenter is at the front of the room, the sound can be distributed evenly by increasing the loudspeaker level as you move from this zone to the furthest listener. An ideal mix-minus is calculated in the DSP system based on the distances between the person speaking and the listeners, and a matrix mixer ensures that a well-balanced mix is distributed to each zone.

Audience sound control

Mix-minus refers to the muting or lowering of loudspeaker volume in zones that have active microphones, eliminating the potential for feedback. Mix-minus is also used when incorporating more than one TeamConnect Ceiling 2 microphone zone in order to enable audience participation.

Sound Control

When multiple open microphones are operated at the same time, the systems must be balanced against each other. As interactions between presenters and the audience can often be lively, it is important to manage the interaction between microphones, loudspeakers and the persons speaking. To manage active microphones and loudspeakers, we recommend using an external DSP system to control the zones with automatic mixers. This both improves intelligibility and scales up GBF significantly. To give each loudspeaker zone the desired sub-mix, automixers are complimented by powerful matrix mixers. Equalizers (EQ) and notch filters can be used to fine-tune mixed signals to the given room size.



Pick-up of audience voices

Room Characteristics

While reflections are negligible outdoors due to the distances involved, they must be considered when designing a voice lift system in an enclosed space.

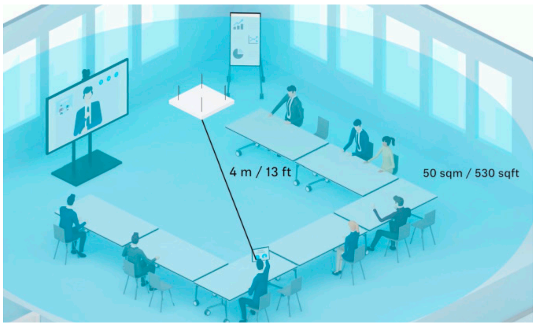
Nearly all surfaces have properties that cause them to resonate differently at different frequencies. Depending on the surface, different phenomena can occur, such as reverberation, echo, standing waves, etc. Controlling these phenomena and making them acoustically compatible with the overall system can be a challenge.

Intelligent beamforming technology utilizes sound wave reflections

To achieve the highest possible speech intelligibility, the background noise level in the room should not exceed 45dBA. Based on the ISL, the distance from the presenter to the farthest listener should be at least 8m (26ft). A voice lift may also be required in smaller rooms, depending on the room acoustics, ambient noise and the presenter's speaking volume.

Adaptation to the Room Characteristics with TeamConnect Ceiling 2

The intelligent technology in the TeamConnect Ceiling 2 makes use of the reflective properties of a room. The narrow beam of 30° ensures that excellent speech intelligibility is achieved even in rooms with many reflective surfaces. This focused beam is able to pick up each individual's speech cleanly, even when the presenter is standing with their back to the microphone (e. g. speaking and using a white board at the same time).

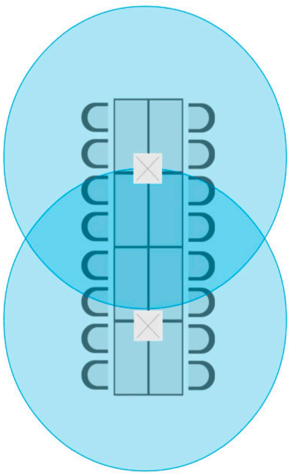


Positioning of the TeamConnect Ceiling 2

The „PAG/NAG Calculation" determines the optimal loudspeaker positions for achieving the highest possible GBF, depending on the room configuration, ambient noise, and the orientation of the TeamConnect Ceiling 2. This positioning ensures that every listener in the room hears the speaker's natural speech without interference.

Rule of thumb

We recommend installing the TCC 2 at a height of 3m (10ft) to achieve a coverage radius of 5m (16ft) and a coverage area of approx. 50 to 60m² (approx. 540 to 650 ft²). These values can also be exceeded, depending on the situation. The adjacent illustration shows the relationship between the required speech area and the range of the microphones.



TeamConnect Ceiling 2 coverage area

Risk of Feedback

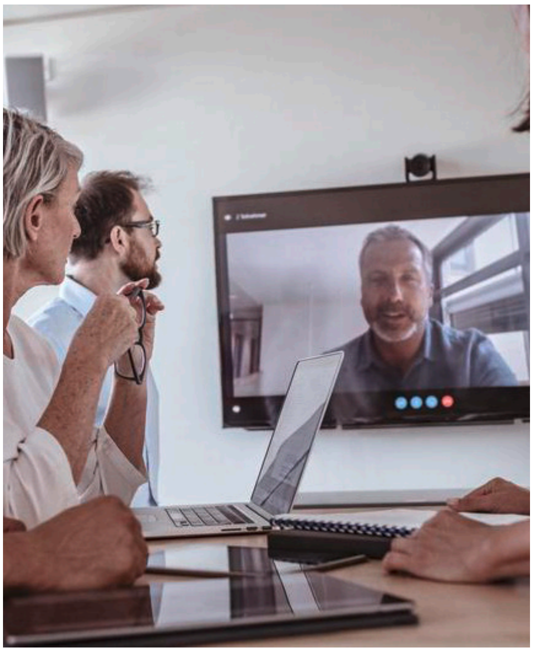
Based on the ISL, it is necessary to boost the incoming sound to achieve good speech intelligibility. However, sound reinforcement is directly related to feedback from the loudspeakers.

How does feedback occur?

When a critical level is exceeded, the sound reproduced by the loudspeaker re-enters the microphone system, forming a continuous loop. Each doubling of the number of open microphones means that the system gain must be reduced by 3dB to avoid feedback.

Factors that might lead to feedback

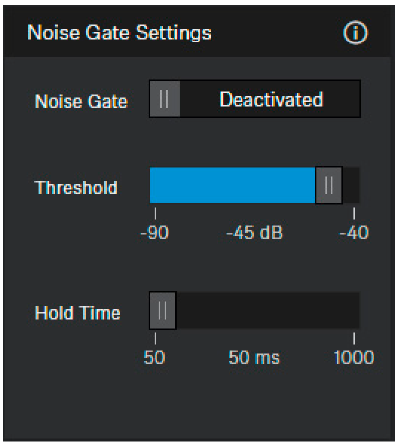
- Relative positions of microphones and sound reinforcement loudspeakers
- Number of open microphones (NOM)
- Polar pattern of microphones and sound reinforcement loudspeakers
- Acoustic conditions of the environment (reverberance)



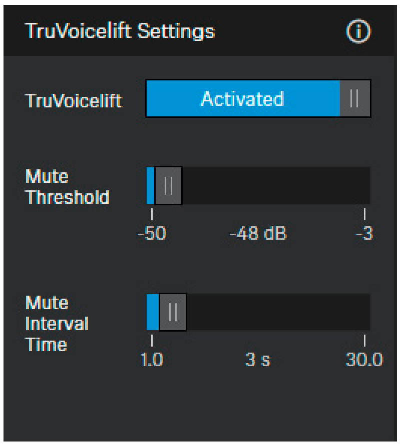
Automatic Feedback Protection with TeamConnect Ceiling 2

During pauses in speech, the „Noise Gate Settings“ suppress the reinforcement of background noise. This is especially important when multiple microphones are used simultaneously (see „Noise Gate Settings“).

With TruVoicelift, a „Unique Frequency-Shifting Technique“ is used to reduce the risk of feedback while simultaneously increasing the possible GBF. If feedback occurs, the „Mute Threshold“ feature temporarily shuts off the microphone output (see „Unique Frequency-Shifting Technique“).



Noise gate settings



TruVoicelift settings

PAG/NAG Calculation

Below is a sample PAG/NAG calculation for typical use cases with TeamConnect Ceiling 2.

A PAG/NAG calculation is needed to determine if the system can provide sufficient voice lift to achieve good intelligibility. The goal is to achieve a value of >=0 dB for the difference of PAG – NAG. This indicates a potentially stable voice lift system. If the calculated value is negative, the system is potentially unstable and could suffer from feedback and insufficient intelligibility.

The following formulas are needed to determine the PAG/NAG value:

$PAG = 20*\log((D_0 * D_1)/(D_2 * D_s)) - 10*\log(NOM) - FSM$

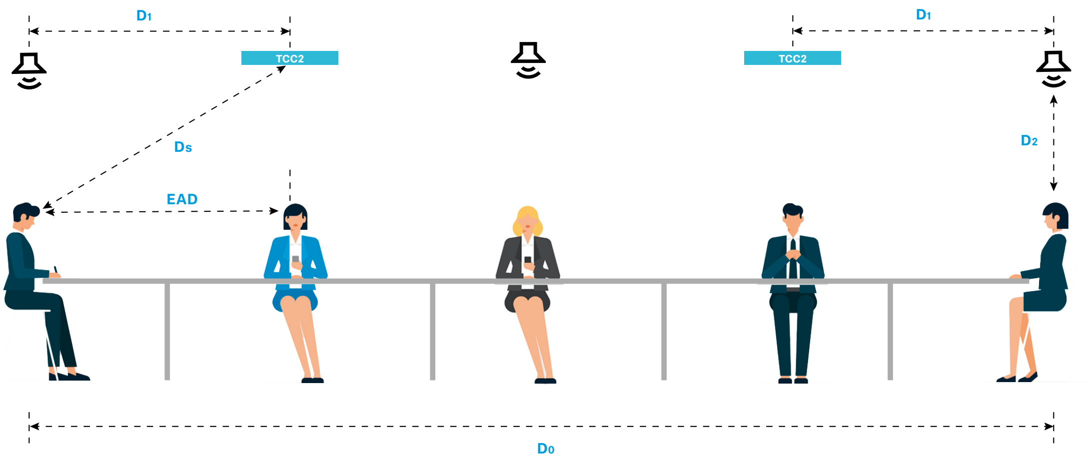
$NAG = 20*\log(D_0 / EAD)$

$PAG - NAG = x$ dB

If $x >= 0$, this value indicates a potentially stable system.

If $x < 0$, this value indicates a potentially unstable system, which tends to suffer from feedback.

PAG/NAG Calculation Example for a Large Boardroom



Distance (d)		d in m (ft)
D_0	Talker <-> farthest listener	10 m (33 ft)
D_1	TeamConnect Ceiling 2 <-> loudspeaker	3 m (10 ft)
D_2	Farthest listener <-> loudspeaker	1.8 m (6 ft)
EAD	Talker <-> nearest listener	3.7 m (12 ft)
D_s	Talker <-> TeamConnect Ceiling 2	2 m (7 ft)
NOM	Number of open microphones	2
FSM_{TCC2}	TeamConnect Ceiling 2 feedback stability margin	6 dB
Calculation example		
PAG	20*log((33 * 10)/(6 * 7))	17.9
– NOM	10*log(2)	– 3
– FSM	6	– 6
– NAG	20*log(33 ft / 12 ft)	– 8.78
		0.12

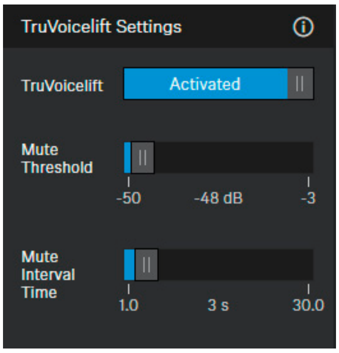
This example considers two open microphones (NOM). Applying the formula, we obtain a value of 17.9 for PAG and 8.78 for NAG. If we subtract the NOM and FSM from PAG, we get a value of 8.9. This value is greater than the calculated value of NAG (8.78), which means that there is enough voice lift to achieve good intelligibility without feedback.

Whether this positive value is sufficient for the required performance or not always depends on the room acoustics. You can find more information about PAG/NAG here.

TruVoicelift Mode in TeamConnect Ceiling 2

TruVoicelift is the optimal solution for both conferencing and in-room audio for classrooms, lecture halls, boardrooms and more. It enables the customer to significantly amplify speech within the meeting room. With the help of flexible controls, you can define Exclusion Zones and a Priority Zone, and activate powerful algorithms to control feedback. These and other features set the TeamConnect Ceiling 2 apart from the competition and deliver a sound experience that no other ceiling microphone can achieve.

Automatic Feedback Protection



TruVoicelift settings

When TruVoicelift is switched on, various features are activated to automatically suppress feedback. One is the special frequency-shifting procedure, which occurs automatically, and the other is the “mute threshold” and “mute interval time” settings.

Unique Frequency-Shifting Technique

TruVoicelift uses a unique frequency-shifting technique that applies algorithms to the microphone output signal to mitigate the risk of feedback from the loudspeakers in the room.

The various frequencies are shifted in real time while maintaining the voice quality. Shifting the frequencies considerably reduces the risk of feedback, as sound is transmitted not as a single wave bundle but in multiple wave packets. It also allows you to achieve higher gain.

This frequency-shifting technique is used to achieve the best possible speech results. This method is therefore ideally suited for voice lift applications in meeting rooms of all kinds. It is not suitable for use in professional music applications.

Mute Threshold

TruVoicelift has a built-in automatic mute function that temporarily shuts off the output if the microphone level exceeds the configured Mute Threshold. This situation can be caused by an unexpected increase in volume. You can use the slider to adjust the Mute Threshold relative to the microphone level from -50dB to +3dB in steps of 3dB.

Mute Interval Time

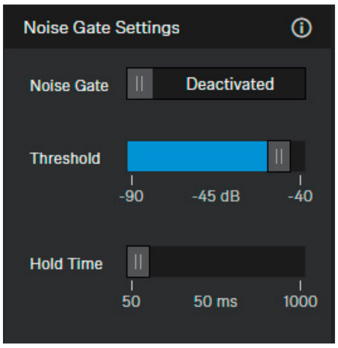
The Mute Interval Time defines how long the microphone is muted after the Mute Threshold has been exceeded. You can adjust this setting to the time needed for the acoustic situation to return to normal. The slider lets you adjust the interval time from 1s to 30s in steps of 1s.

Recommendation:

We recommend carrying out a test on site, as the room acoustics will determine the outcome of the test. Leave the value at -50dB. To adjust the unit, clap your hand several times while gradually increasing the intensity of your clap. Observe the value at which the system mutes the sound. Use this hand clap test to find a good noise level to use as the starting value and estimate a suitable interval time.

Noise Gate

The Noise Gate ensures that background noise is not amplified during pauses in speech. This is especially important when multiple microphones are used simultaneously. During pauses in speech, the system usually boosts the gain, as it assumes there is not enough sound pressure. This causes the background noise to be amplified unnecessarily.



Noise Gate Settings

Noise Gate – Threshold

To avoid this, you can set a threshold level at which the system will mute the microphone. You can also set a mandatory Hold Time. The noise gate will open the microphone audio output only after the microphone in question exceeds the defined threshold value.

You can use the slider to adjust the minimum threshold level from -90dB to -40dB in steps of 1dB.

Recommendation:

To achieve the highest possible speech intelligibility, the background noise level in the room should not exceed 45dBA. This is a good initial value for testing and, if necessary, adjusting the noise level while the meeting is taking place.

Noise Gate – Hold Time

This setting determines how quickly the microphone is opened again. A delay of up to 1000ms can be set. You can adjust the Hold Time between 50 and 1000ms.

Recommendation:

You can delay opening of the microphone channel depending on the type of speech and the ambient noise in the room. We recommend leaving the setting at 50ms to start to achieve the lowest possible latency. You can gradually increase this value depending on the situation and the type of speech.

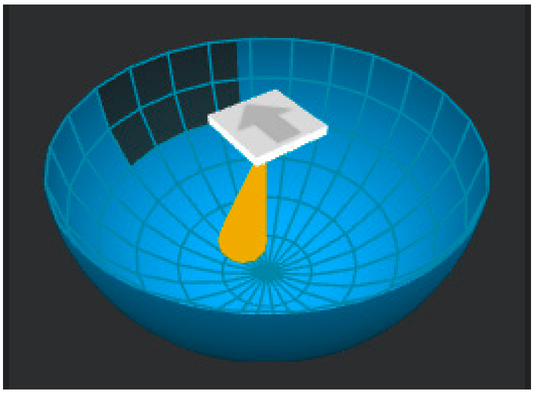
Advanced Exclusion Zones

Meetings often suffer from unwanted background noise, for example from air conditioning systems, side doors, loud coffee machines and adjacent rooms. Loudspeakers with audio from far-end participants can also be a source of interference for the microphone. To suppress this unwanted background noise, you can define Exclusion Zones in which the beam tracking will ignore audio signals. By specifying the vertical and horizontal positions of Exclusion Zones, you can easily suppress sources of noise.

Maximum Flexibility with 5 Exclusion Zones

With the TeamConnect Ceiling 2, you can configure up to 5 Exclusion Zones based on their position relative to the ceiling microphone. You select the vertical and horizontal positions of these Exclusion Zones, which can be activated simultaneously.

Once the unit is initialized, the TeamConnect Ceiling 2 uses a real-time algorithm to detect the sources of interference, which are then visualized as a 3D model directly in the Control Cockpit. This enables you to quickly and easily define a precise Exclusion Zone.



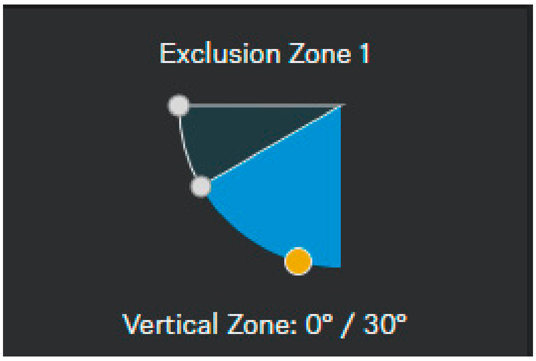
Automatic beam technology

Flexible Coverage Area

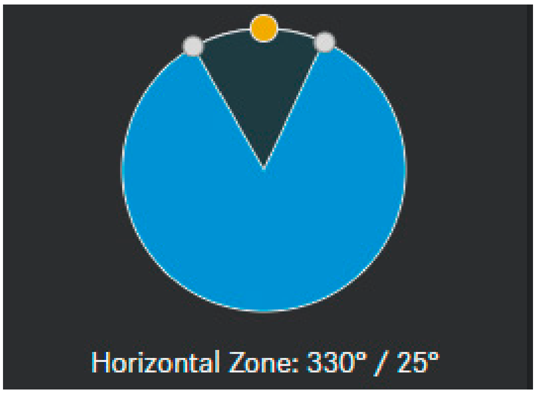
The vertical angle can be adjusted from 0° to 90°. The horizontal angle can be adjusted from 0° to 360°. These coverage ranges encompass a complete hemisphere originating from the ceiling microphone. You can set the individual zones in the Control Cockpit software with just a few clicks.

You can use the Exclusion Zones for different voice lift scenarios, depending on the situation:

- If there is no noise in the room, the zones can remain switched off.
- If permanently installed devices (e.g. air conditioners, loudspeakers) are to be excluded, you should define a vertical zone as well as a horizontal zone with the appropriate height and width.
- If there are multiple sources of noise, you can configure and use multiple Exclusion Zones simultaneously.

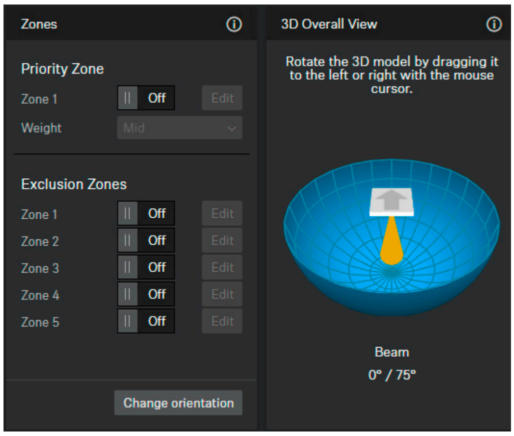


Adjustment of the vertical zone



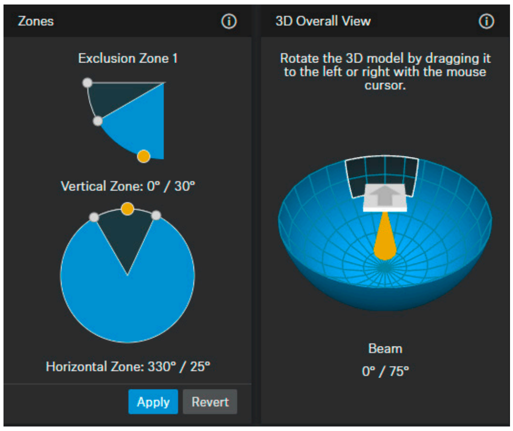
Adjustment of the horizontal zone

Example 1 – Small Classroom
Small classrooms have less ambient noise or standing waves that could affect the device. In this case, you can define an Exclusion Zone for the ceiling to ignore the ceiling-mounted voice lift loudspeakers. Independent of the Exclusion Zones, you can also set up a „Priority Zone“ to prioritize the teacher’s voice.



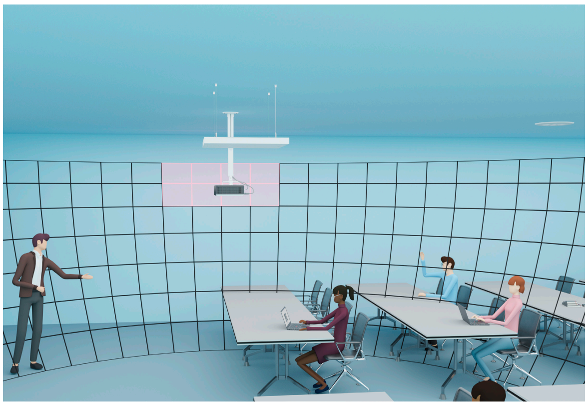
Example 2 – Meeting Room

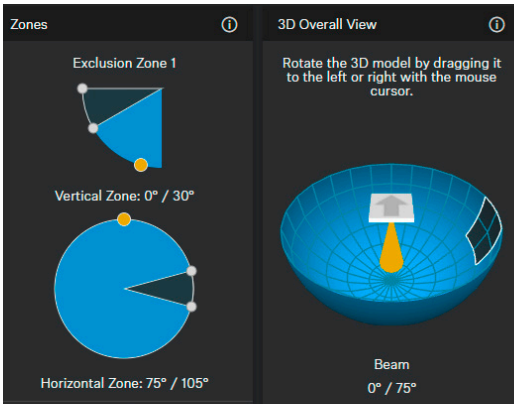
Far-end audio from the loudspeakers produces constant background noise that poses a risk of feedback. If you define a vertical and horizontal Exclusion Zone for this area, the beam will no longer track audio signals from this area.



Example 3 – Lecture Hall

During a lecture, constant noise from the projector fan will cause the microphone to be permanently activated. Setting up an Exclusion Zone for the projector will attenuate the noise from this fan.



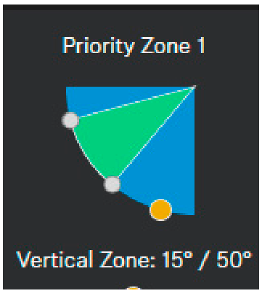


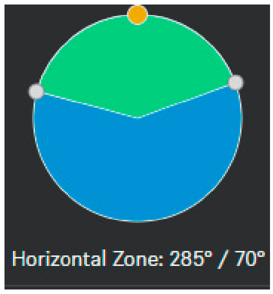
Priority Zone

During lively discussions in meeting, the moderator needs to be able to maintain control of the conversation. You can set up a Priority Zone so that voices are not given precedence based on volume alone. The moderator will always be prioritized in the incoming signal, even if their voice is quieter. This ensures that the person in charge also has vocal control of the situation.

Highlighting a Preferred Area

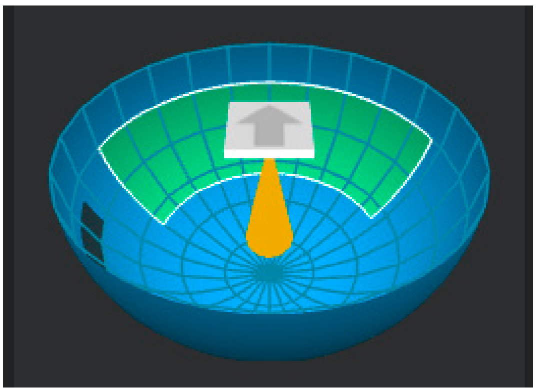
The Priority Zone is used to keep the focus on the moderator's voice. You can easily configure a Preferred Zone in the Control Cockpit, taking into account the moderator's range of movement.



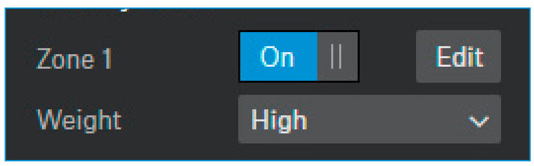


You can adjust the dimensions of the zone by 90° vertically and 360° horizontally.

You can also set the weighting of the Priority Zone. The weighting determines how intensively the beam focuses on this area. You have the following options:

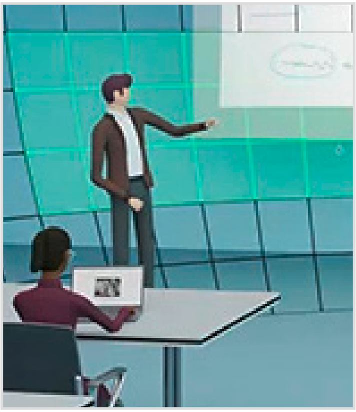


- **Mid:** Increases the weighting of Priority Zone audio to about 1.5 times the normal audio output (e. g. in rooms with normal ambient noise).
- **High:** Increases the weighting of Priority Zone audio to about 2 times the normal audio output (e. g. in rooms with high ambient noise).
- **Max:** Increases the weighting of Priority Zone audio to about 3 times the normal audio output (e. g. in rooms with high ambient noise and a quiet presenter).



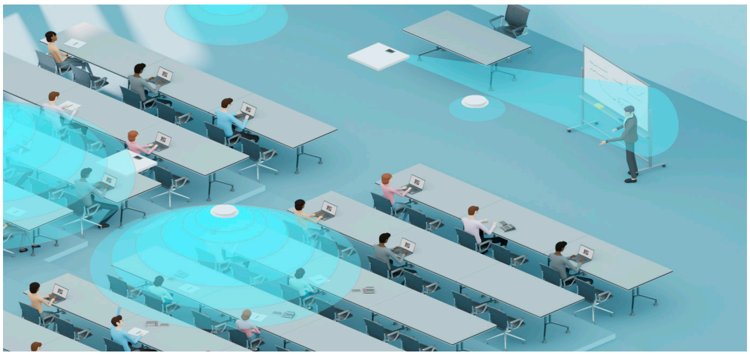
Example 1 – Classroom

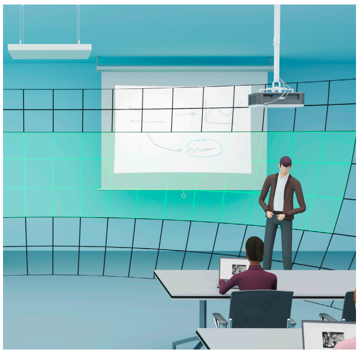
When the system is integrated in a classroom, multiple TeamConnect Ceiling 2 units are used simultaneously and divided into coverage zones. The first coverage zone includes the teacher's area. A generous Priority Zone is set up for this area, based on how the teacher typically moves about the room. Because of the teacher's authority in the classroom, the weighting can be set to "Mid."



Example 2 – Lecture Hall

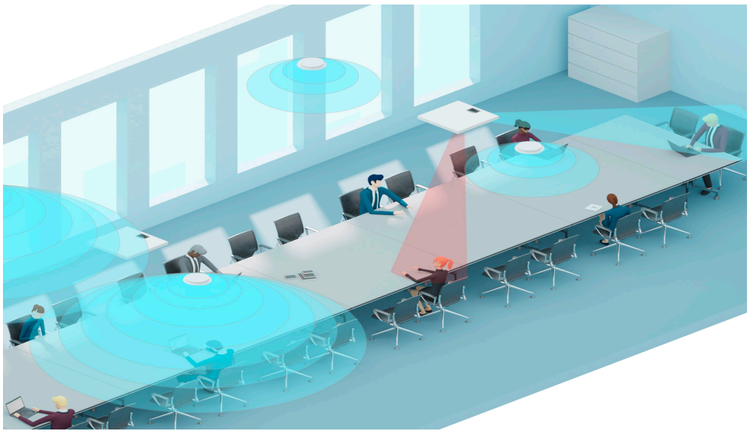
A lecture hall usually contains more occupants than a classroom. As a result, the potential noise level is most likely higher. The weighting can be set to "Mid" or "High."

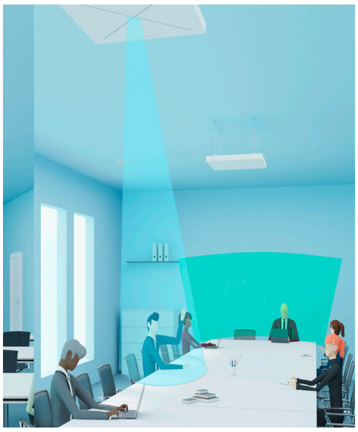




Example 2 – Boardroom

When discussions get lively in a meeting, it can be difficult to control the conversation. In this case it is a good idea to configure a Priority Zone for the moderator and set the weighting to "Max." The TeamConnect Ceiling 2 captures all participants in its range but gives priority to the configured Priority Zone. This enables the moderator to keep acoustic control of the situation.





Additional Benefits of the TeamConnect Ceiling 2

Implementing a voice lift system with traditional microphones is a real challenge for installers and sound engineers. Depending on the system design and the ambient conditions, there may be additional challenges with feedback, ambient noise, maintenance, operation, hygiene and more. The TeamConnect Ceiling 2 offers the following benefits:

Easy Handling

Users do not have to be constantly reminded of correct handling, operation and positioning, or what to do in special situations (e. g. feedback).

Freedom of Movement

A ceiling microphone offers a high degree of flexibility for all participants. The unit uses digital signal processing to locate the person speaking at all times, regardless of whether they are sitting, standing or moving around.

Automatic Adaptation to the Room

The microphones do not need to be physically adjusted if the room is rearranged. You can configure the system in the Control Cockpit with just a few clicks.

Easy Setup and Maintenance

The microphone's central position on the ceiling eliminates the need for individual microphones on the table.

For system engineers, integrating the system into an existing network does not require any complex analyses or installation work.

TeamConnect Ceiling 2 eliminates the need for maintenance and additional personnel.

No Hygiene Concerns

Touchless audio systems are designed for communication scenarios where multiple people are involved. Hygiene standards are met despite the constant turnover of users.

Checklist for Voice Lift Design

When considering a voice lift system, many criteria need to be taken into account. This checklist provides an overview of the most important points to consider when integrating the system.

✓	**Consider all the presenters and listeners in the room.** **How many people will be speaking in the room and where is their audience?** This question helps to determine the minimum space required and the coverage area needed to capture everyone who will be speaking.
✓	**Determine the number of microphones needed.** **Is a microphone needed for the audience to interact with the presenter?** This question helps to determine the number of microphones needed to ensure proper interaction between the presenter and the audience.
✓	**Plan the microphone coverage.** **Have you considered the distances between the microphones as well as the distances from the microphones to the sound reinforcement speakers, the presenters and the audience?** Proper positioning of the devices helps to ensure that distances are suitable for picking up incoming signals and preventing potential feedback.
✓	**Determine the loudspeaker coverage and loudspeaker zones based on the listening area.** **From which directions do speech signals come?** This question will help you achieve the most natural sound pressure distribution. Based on the value measured, you can calculate the loss of pressure according to the inverse square law. This value is needed for the important PAG/NAG calculation.
✓	**Consider the ambient conditions in the room.** **Have you determined the reverberation time (RT60) and ambient noise level in the room?** Reverberation is the accumulation of many sound reflections in a room. The "reverberation time" (RT60) indicates the period of time it takes for reflected sound to fade out in an enclosed space after the sound source has stopped. This time is important in determining how a room will respond to acoustic sound. **Recommended RT60 values:** (see table below)
✓	**Identify all interacting systems.** **Will the voice lift system be operated with an existing conference system or a DSP system with AEC? Will remote participants be actively speaking?** This question helps to identify all systems that must be considered as potential audio signal inputs (including far-end signals). These systems must be harmonized to produce a balanced distribution of sound.
✓	**Calculate the Potential Acoustic Gain (PAG) and Needed Acoustic Gain (NAG).** **What PAG/NAG level can be achieved based on the proposed placement of microphones, loudspeakers and people?** A PAG/NAG calculation is needed to determine if the system can provide sufficient voice lift to achieve good intelligibility. A value of >=0dB for the PAG – NAG difference indicates a potentially stable voice lift system (see „PAG/NAG Calculation").
✓	**Manage all client expectations.** **Were all the requirements met?** Based on all the data collected (number of presenters/listeners and their relative distances in the room, room characteristics, possible sources of interference, possible GBF, DSP mixing requirements, etc.), the overall infrastructure should be closely matched to the customer's expectations.

Recommended RT60 values:

Location	Volume	Critical distance D_c	Recommended RT60
Classroom	< 200 m³	2 m	0.4 – 0.6 s
Office	< 1000 m³	3.5 m	0.5 – 1.1 s
Lecture hall	< 5000 m³	6 m	1.0 – 1.5 s

Glossary

AEC **Acoustic Echo Cancellation**
AEC is used to remove echoes, reverberation and unwanted additional noise from a signal passing through an acoustic space. This function is mostly needed when a person is connected via a remote connection, known as "far end."

EAD **Equivalent Acoustic Distance**
Distance between the talker and an un-aided listener.

AV **Audio-Visual**
Involving the use of recorded pictures and sound, or the equipment that produces them.

dB **Decibels**
A unit for measuring the loudness of sound.

DSP **Digital Signal Processing**
Used for digital encoding of "live" signals such as audio, video etc. and allows these signals to be stored, manipulated, edited, replayed, and transferred much more efficiently and accurately than with strictly analog methods.

EQ **Equalizer**
Electronic equipment that adjusts (= makes slight changes to) the frequency of recorded sound to make it sound better.

FSM **Feedback Stability Margin**
Stability value which is determined and used for robustness against disturbances.

GBF **Gain-Before-Feedback**
A practical measure of how much a microphone can be amplified in a sound reinforcement system / voice lift system before causing audio feedback.

NAG **Needed Acoustic Gain**
Value, in decibels, that the system requires to perform effectively.

PAG **Potential Acoustic Gain**
Potential maximum amplification, in decibels, that the system can produce before feedback occurs.

PA **Public Address System**
Equipment for making sound louder in a public place.

SR **Sound Reinforcement System**
System of microphones, signal processors, amplifiers and loudspeakers that ensures controlled mixing and distribution of live sound to a larger or more distant audience.

RT **Reverberation Time**
The time required for reflecting sound to fade away in an enclosed space after the source of the sound has stopped.

RT60 **Reverberation Time 60**
The period of time required for the sound pressure level to decrease by 60dB after a sound source is abruptly turned off. RT60 is also a common abbreviation for Reverberation Time.