Smart Audio & Control – Beyond Traditional Conferencing

The audio control and quality
of the future have arrived today.

CONFERENCE

Smart Audio & Control – Beyond Traditional Conferencing

Televic Conference has supported large scale managed meetings for decades. Institutions including the United Nations, US Department of State, European Union, Nato and many world Government institutions. Most of these systems require specific features including, request to speak, voting, agenda, timers and language interpretation and are considered "Traditional Conferencing Applications"

While most AV applications do not require these conferencing features, the discussion microphone also provided a solution for acoustic challenges including tall ceilings, glass or concreate walls, flex rooms or another solutions where local audio was needed at each seat.

During the pandemic, the requirement to meet at safe distances, work around plexi-glass and connect to remote platforms as Teams or Zoom was a requirement. Discussion systems were requested as a new solution to various applications and many will agree that the hybrid requirement is here to stay.

City Government

Hybrid Audio Flexibility with City Government purpose built features on demand – Voting, Agenda, camera control, etc

Corporations

Alternative to ceiling mics with local audio in front of each participant – Flush mount, desktop and video versions with built in web cam.

Courts

New microphone modes needed in courts, hybrid audio heard everywhere and flexible audio routing to the court recorder of choice: FTR, Liberty, etc.

Lecture Halls

New audio distribution capability streamlines the installation with redundant loop technology while maintaining the DSP design requirements and control.

New Smart Audio and Control Features

Televic's Smart Audio & Control presents a set of innovative audio developments. It introduces a new microphone mode and applied the latest processing and routing techniques to make your meetings crisper and more intelligible than ever before.

The audio control and quality of the future have arrived today.

+ Optimized intelligibility for both in-room and remote participants
+ Consistent audio coverage in every kind of meeting room set-up
+ Reduced acoustic feedback
+ Hands-free discussion mode
+ The ability to use your current system with our DSP audio-routing type

Dynamic Mix Minus

Create crystal clear in-room and hybrid meetings.

Don't worry about bringing in additional speaker systems. Thanks to our dynamic mix-minus, based on the open microphones, you can count on consistent audio coverage in the entire meeting room. Whether your participants are right next to you or overseas (through Teams or Zoom), they will be able to receive and convey messages in the same crystal-clear quality.

How is this achieved?

Traditionally, the built-in loudspeaker of a conference unit was muted when the microphone got active in order to prevent feedback.

TRADITIONAL

Televic has now added dynamic mix-minus in the conference architecture.

The term "mix-minus" refers to the audio mix that includes all audio sources except for the one that is being sent to the specific participant's loudspeaker.

DYNAMIC MIX-MINUS

This means that the loudspeaker stays active and no signal is present of the microphone of that conference unit and hence preventing feedback.

These mix-minus channels are dynamically created in the conference systems control unit as soon as a microphone unit gets activated.

Having mix-minus in the system creates a more consistent coverage throughout the room. Even rooms with a small amount of units or room layouts where units are spread out further from each other can count on the individual loudspeakers integrated in the conference devices to clearly hear and being heard.

It is like a musician receiving its own monitor mix.

Below picture show a typical example of a city council chamber.

A common issue in such application is that councilors can't hear the conversation another across the dais, especially nowadays where hybrid meetings are here to stay.

In the example you can see the speakers desk is at an individual location further away from the other conference devices. Not having the mix-minus functionality meant that when the microphone is active no audio of the remote meeting platform nor the audio of the councilors could be heard. As a consequence extra amplification or loudspeaker needed to be provided for that location with risk of creating feedback.

The addition of the Dynamix mix-minus prevents the need for additional equipment and provides crystal clear conversations straight from the conference units.

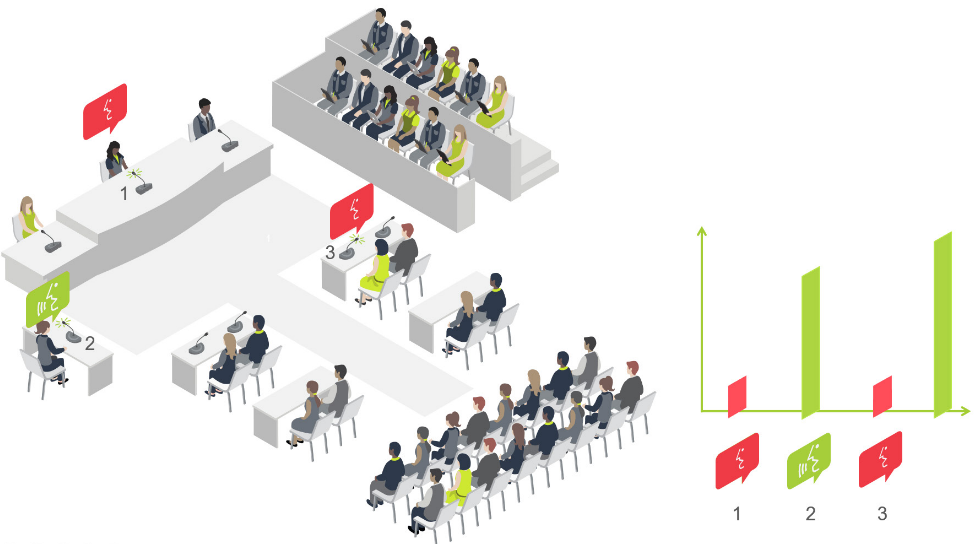
Gain Sharing

All open microphones get a dynamically calculated gain such that the total gain of all open microphones always remains constant, even if at some point more people begin to talk.

This means that if only one person is speaking, this person gets the full gain of the system, while the other microphone gains go down. This can be seen in the pictures below. Because of this, background noise picked up by the ‘silent’ microphone does not get amplified as much as with a traditional system where each microphone has a fixed gain.

If no one is speaking, each microphone gets equal gain.

If a participant starts speaking, this speaker gets more gain, while gain of the other microphones gets reduced.



If two persons are speaking at the same volume, they each get the same gain allocation.
The total gain always remains constant.

Applying this gain sharing principle offers several benefits:

- There are often multiple participants speaking at different volumes or distances from the microphone. The gainsharing algorithm can automatically adjust the gain of each microphone in real-time, ensuring that the audio levels are consistent and that the loudest and clearest sound sources are prioritized. This results in greater gain before feedback and a more natural and seamless listening experience for conference attendees.

- A conference system is often used in settings where clear communication is essential, such as business meetings, council debates, or educational events. Using gainsharing, background noise and other distractions are reduced, ensuring that attendees can hear and understand the speakers more easily.

- A conference system typically involves many microphones and can be complex to tune taking into account varying factors during the meeting like number of participants speaking. The gainsharing algorithm can help to simplify this process, reducing the need for adjustments and minimizing the chances of inconsistent behaviour.

Speech detection

Look who’s talking: speech detection

Guarantee vivid engagement without confusing your listeners. With the speech-detection feature, you will always know exactly who is speaking. Even in a discussion between multiple people, you can make sure that only the speaker is displayed. It significantly improves video capturing (no need to trigger cameras with a microphone button), and ensures more accurate transcriptions and data reports.

Hands-free mode

Not having to press a button to take part in the conversation has plenty of advantages in lots of meeting formats:

- Increased efficiency: Participants can speak freely and spontaneously without the need to interrupt the flow of the conversation by pressing a button to speak. This can lead to more productive and efficient meetings, as participants can share their thoughts and ideas more easily and seamlessly.

- Improved collaboration: Removing the need to push a button can foster a more collaborative environment, where participants feel more comfortable and empowered to speak up and contribute to the discussion. This can lead to a more inclusive and dynamic exchange of ideas.

- Better engagement: With a more natural conversation flow, participants are more likely to remain engaged and attentive throughout the meeting. This can lead to better comprehension, retention, and follow-through on the topics discussed.

- Ease of use: Conference systems that don't require button pushing are often simpler to use, reducing the need for training or technical support. This can save time and resources and make it easier for participants to join and participate in meetings.

- Accessibility: For people with disabilities, pushing a button to speak can be a barrier to participation. Removing this requirement can make meetings more accessible and inclusive for everyone.

This is exactly what Televic's new 'Hands-free' microphone mode provides.

Hands-free mode

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Mix-minus

+ Gainsharing

+ Speech detection

It allows to use our wired conference solutions without pushing any button at all. All microphones in the system are 'live' with each loudspeaker providing an individual 'mix-minus' channel.

This means every participant can join the discussion whenever he wants, hands-free.

When you need to cough or want the have a side conversation the microphone button can be configured as a mute button.

So is this hands-free mode the same as the typical called VOX mode?

No, the traditional voice operated microphone mode known in conference solutions works on a gating principle. This means that the microphone only gets activated when energy is detected.

There are a few potential disadvantages of using a voice-operated microphone (VOX) mode in conference systems, including:

- Background noise: VOX mode can be triggered by ambient noise in the room, such as shuffling papers or coughing, leading to interruptions and distraction for the speaker and other participants.

- Inconsistent audio quality: VOX mode can cause inconsistent audio quality, as the microphone may cut in and out depending on the speaker's distance from the microphone, the volume of their voice, and other factors.

- Delayed responses: VOX mode can cause a delay in the audio response, as the system may take a moment to detect and activate the microphone. This can result in a lag in communication and make it difficult to have a natural conversation. Moreover in case the microphone system feeds a recording or transcription application first syllables of sentences may get lost and makes it difficult to capture the record.

As the hands-free mode work on the gainsharing principle where all microphones are active all the time and the gain is dynamically adapted this results in:

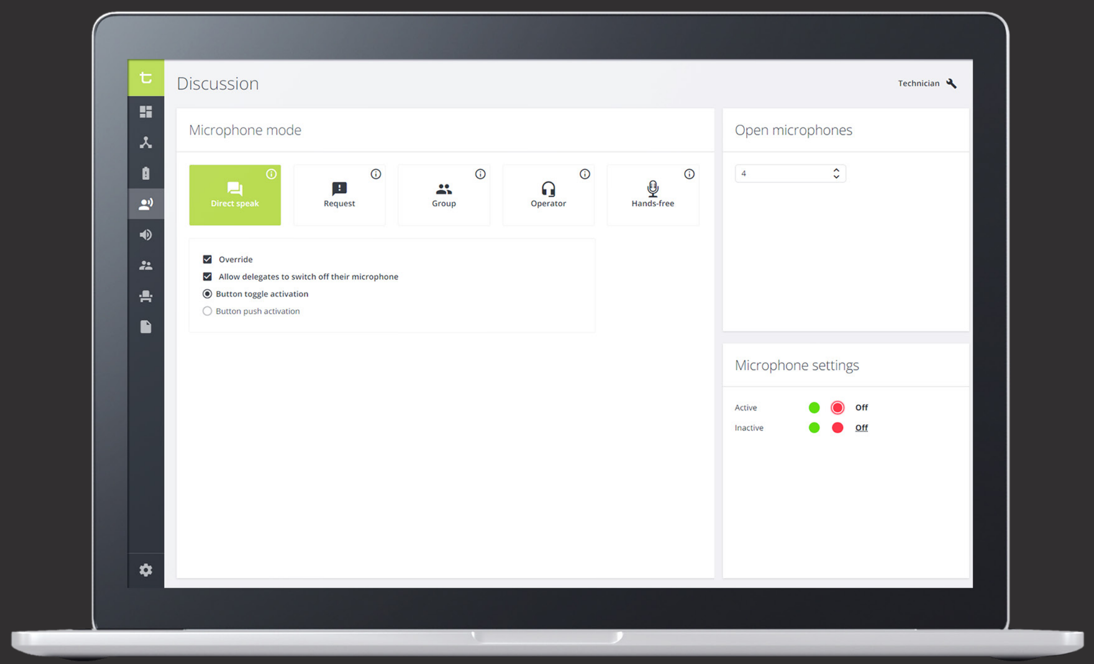
- Improved background noise reduction: Gainsharing algorithms can adjust the microphone gain based on the ambient noise level in the room, effectively reducing background noise without cutting off the microphone. This results in a more natural conversation and less disruption.

- Consistent audio quality: Gainsharing algorithms can maintain a consistent audio level for all participants, regardless of their distance from the microphone or the volume of their voice. This ensures that everyone is heard clearly and consistently throughout the conversation.

- Quick response time: Gainsharing algorithms can adjust the microphone gain instantly, providing a instantaneous response time. This results in a more natural conversation flow and avoids delays in communication.

Overall, gainsharing algorithms provide a more sophisticated and flexible solution for managing microphone gain and audio quality in conference systems, and can solve many of the problems associated with VOX mode.

Discussion microphone modes – Choices on demand!



Meeting rooms serve a variety of purposes, catering from those focused on decision-making to those centered around collaboration. These spaces are designed to provide a suitable environment for people to come together, discuss ideas, share information, and work towards a common goal.

The Televic Conference solution has a set of tools to facilitate effective communication and enhance productivity. Pick the mode that suits the type of meeting you're having. Would your next meeting benefit from using another mode, simply change it with a click of a button.

Number of open microphones

For all microphone modes with the execption of hands-free the maximum number of microphones that can be active at the same can be set. This helps to focus the conversation, increases intelligibility for both in-room and remote participants and provides a certain flow to meetings.

Direct Speak

This mode mimics the action of unmuting your microphone by pressing the microphone button, speaking as needed, then remuting your microphone that are employed in all major cloud-based meeting platforms, thus is very familiar functionality to most users, and increases audio quality both in the meeting space and remote by limiting the number of open microphones.

Activation mode option:
Sets the activation of a microphone to either push and hold while speaking or on/off toggle.

Override option:
When the number of simultaneous active microphones is reached, the next participant pushing the microphone button will be added to the conversation while the microphone of the first participant that joined the conversation will be deactivated. So, a first in – first out principle.

Request

This mode is typically used for meetings that require more structure or have a certain protocol to follow. Those types of meetings require moderation capabilities and request mode suits this need perfectly.

With request activated, participants don't get to speak when they push their microphone button, instead they are put into the request or queue list.

A chairperson or operator can add participants in the conversation or topic discussed. This can be done on the conference stations by pressing the "next-in-line" button resulting in adding the person who is on top of the queuing list.

With the use of Televic's comprehensive software a chairperson or whomever responsible for guiding the flow of the meeting can decide which participant from the complete queue list to add to the conversation. Also called, jump the queue. Reordering of the request list is also perfectly possible by a simple drag and drop principle.

A option available so the person on top of the queue gets notified to be ready to join the conversation. The LED of the microphone blinks also giving a sense of who will speak next to the chairperson.

Group mode

The Group mode is a semi-automatic mode for meetings that need focused conversations but less protocol or are ran without an operator.

Pressing the microphone button add participants to the conversation, unless the maximum amount of open microphones is reached. If that's the case, the participant will be added in the request list or queue. When a certain participant decides to leave the conversation, the participant on top of the queue / request list will become part of the conversation automatically.

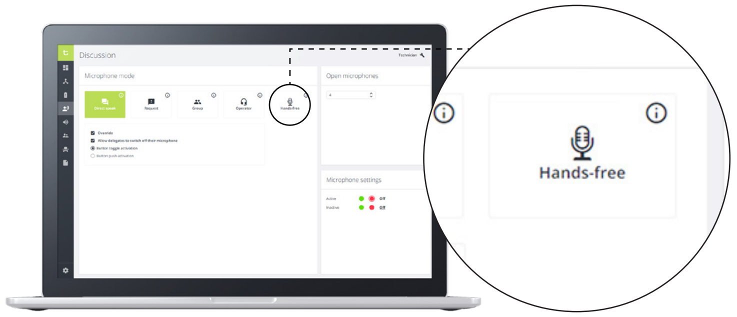
This mode can also be used with activation of microphones based on voice. Participants are added to the conversation when their voice goes above the configured threshold value. Swithing of the microphone happens after the defined number of seconds the voice drops below the threshold. This is a gated activation principle.

Operator mode

In operator mode, the delegates cannot activate their own microphone. The microphones can be activated by a chairperson or operator using the Confero 360 web-based software. In the software a representation of the room layout can be built so it is easy to navigate and control the microphones of certain participants. Participants can decide to leave the conversation by pressing the microphone button.

Handsfree

This is an automatic mode where no buttons needs to be pushed. All microphones are 'live' and gain is automatically controlled and optimized based on the ongoing conversation.



Moreover an algorithm defines who actively speaks so this information can be used for signage applications, reporting, fine grained timestamping and very important in todays hybrid world automatically trigger the camera system in the room to take the speaker into picture without the need for a camera operator.
When participants want to mute their microphone to cough or have a side conversation that should note be captured they can simple hold their microhone button.

LED configurations

The conference units have LED status indicators on the microphone button, backside of the unit and microphone ring. These indicators can be programmed to be off, red, or green to represent the active, muted, or request state of a microphone station. So full flexibility offered on how you want it configured.

LED's can operate on push to talk or with hands free mode.

New design tools for Integrators and AV/IT designers

Traditionally, Conference systems were self-contained ecosystems that provided a summed output of the microphones in the entire system. While this is a viable solution in a lot of circumstances, some AV designs need more fine grained audio processing and control capabilities.

Our partners who integrate Televic solutions worldwide can now use conference solutions in their DSP designs with complete control, while still utilizing the advantages of simplified installation of Televic's Plixus design: standard Cat6 infrastructure, daisy chain or loop cabling for additional redundancy, camera tracking solutions, identifying who speaks or add additional functionality like voting or interpretation when needed.

Combine best of both worlds and enjoy full flexibility in your AV designs.

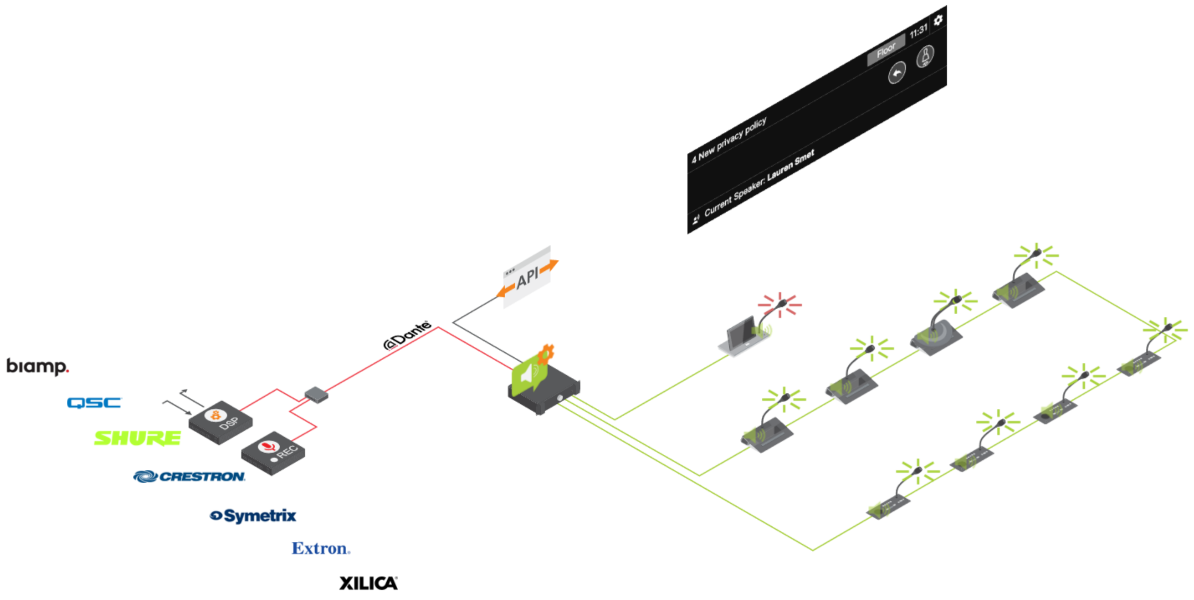
The Smart Audio & Control capabilities offers 2 main capabilities to integrate to your DSP design of choice:

- DSP mode: Activating this mode releases full control to your DSP design.

- Audio routing: Utilize the flexibility of built-in conference capability while optimizing and economizing on DSP resources.

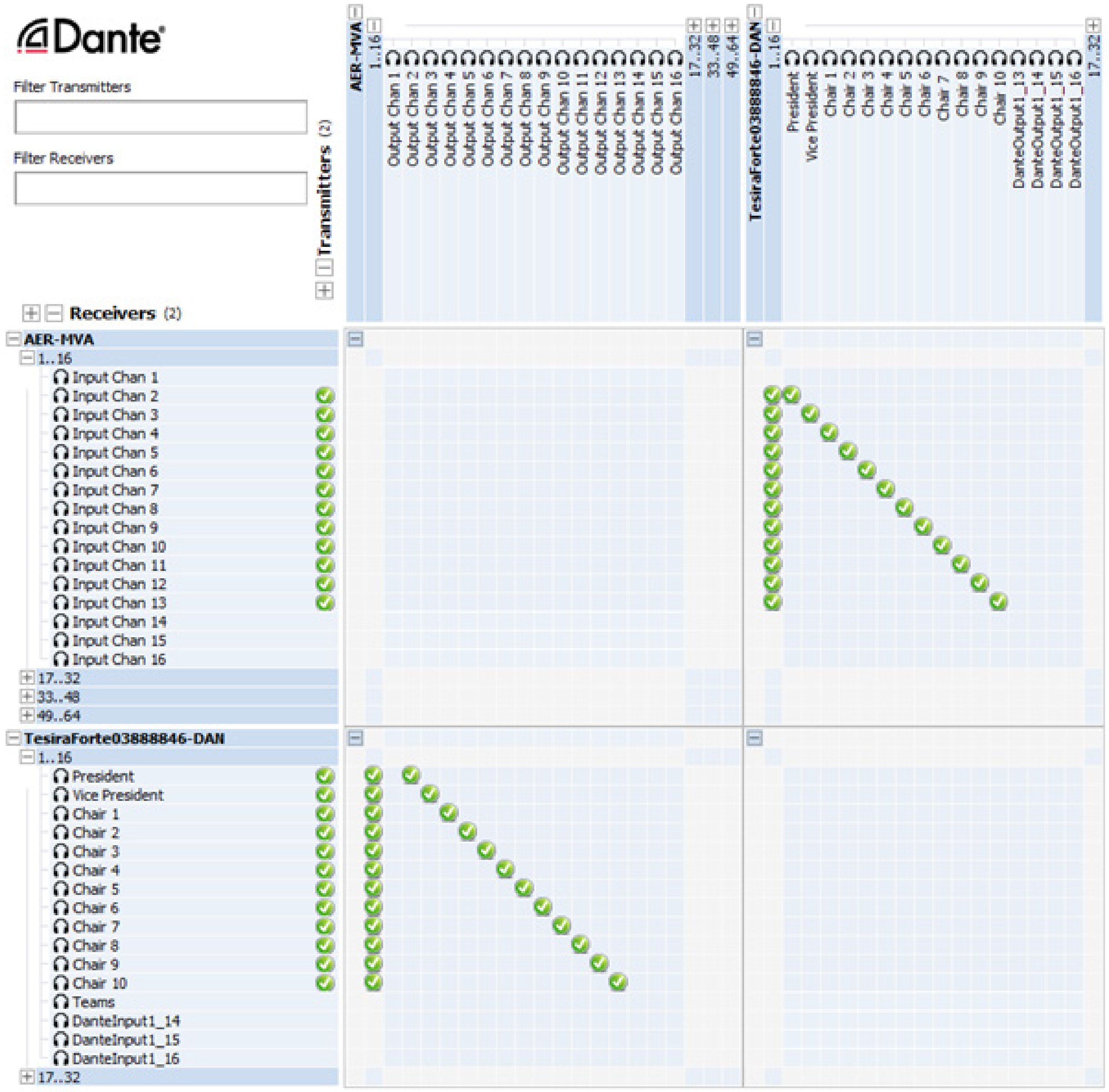
DSP mode:

This DSP mode automatically assigns individual microphones and loudspeakers from up to 32 conference units to individual Dante channels. All microphones are 'live' in the conference solution and releases unprocessed audio to dante channels so processing can be performed in DSP of choice. A separate dante channel is provided to address the built-in loudspeakers of the conference units.

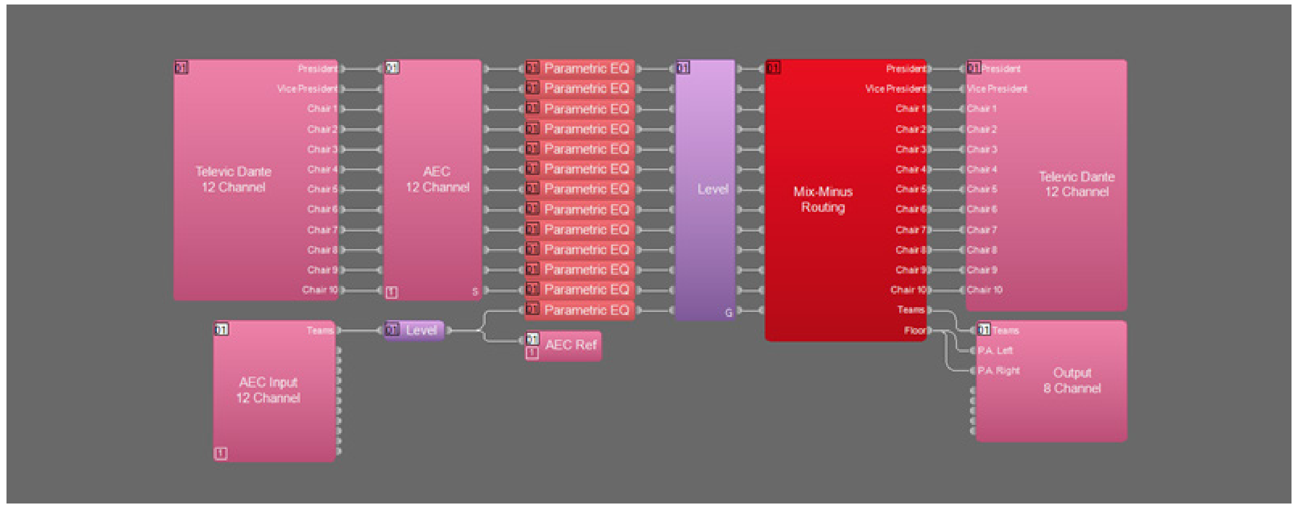


This enables designers to process each microphone and speaker as if it were a standard microphone and speaker in any room. With this, designers can apply AEC to each microphone, EQ specific microphones, group speakers as a zone, and more from their DSP.

Below an example of the routing of Dante controller between a Plixus system and a Biamp Tesira Forte and a Biamp design file providing AEC, EQ, mix-minus matrix mixing.



Dante controller routing



Biamp design file

This mode offers full audio processing capabilities to the DSP, so this is part of the Smart Audio capabilities.

The control part of Smart Audio & Control, references to the granular API capabilities our solution can also offer.

Want the LED of the microphone lid up to show who's speaking? No worries, an event is available to do so. Want to offer push to mute? No worries, capture the event of the microphone button and mute that channel in your DSP.

You can take it even a step further by utilizing default functionality our conference solution offers:

- The capability to configure names to devices means this information can be shared to participants in the room. The info can be made available on the screen of conference devices like Confidea FLEX or uniCOS, visualize the room layout with who's talking on a big screen in the room or use the naming information for reporting and timestamping in recording applications.

Or use this information to embed the name of who's speaking as an overlay onto the video provided from the PTZ camera solution, know that Televic has these solutions in their offering!

Confero Audio:

The Confero Audio software offers the capability to quickly and easily group and route audio input & outputs. Create a configurations and manage everything from the routing matrix.

Changing between different configuration is a matter of a click of a button.

This tool opens a wealth of flexibility.

- Creating groups of audio input and output components (microphones, auxiliary input/output, Dante input/output etc.)

- Visualizing the various routing groups as a matrix

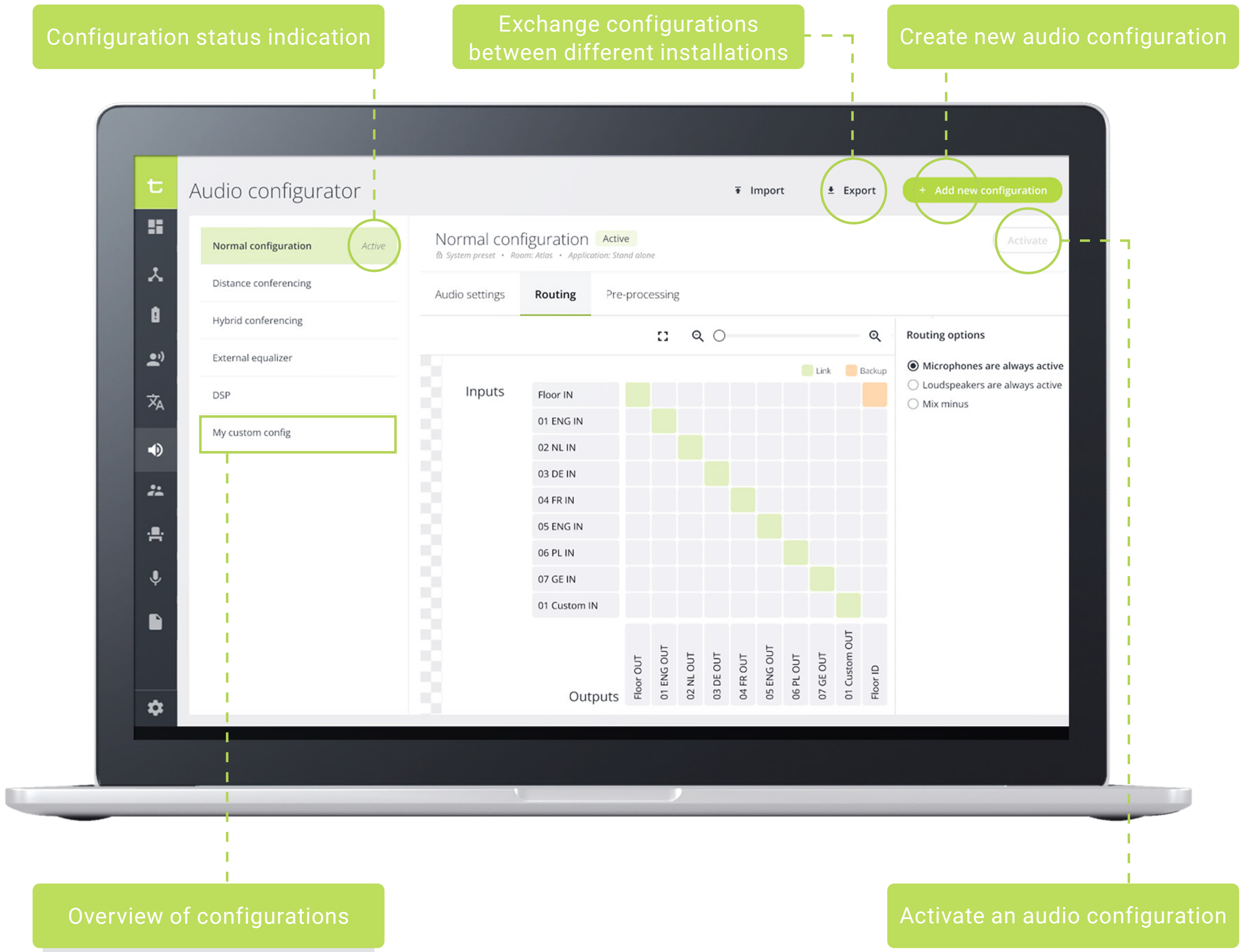
- Controlling the audio routing matrix

- Configuration of interpreter channels to external channels

- Integration towards Teams and Zoom applications for meeting application

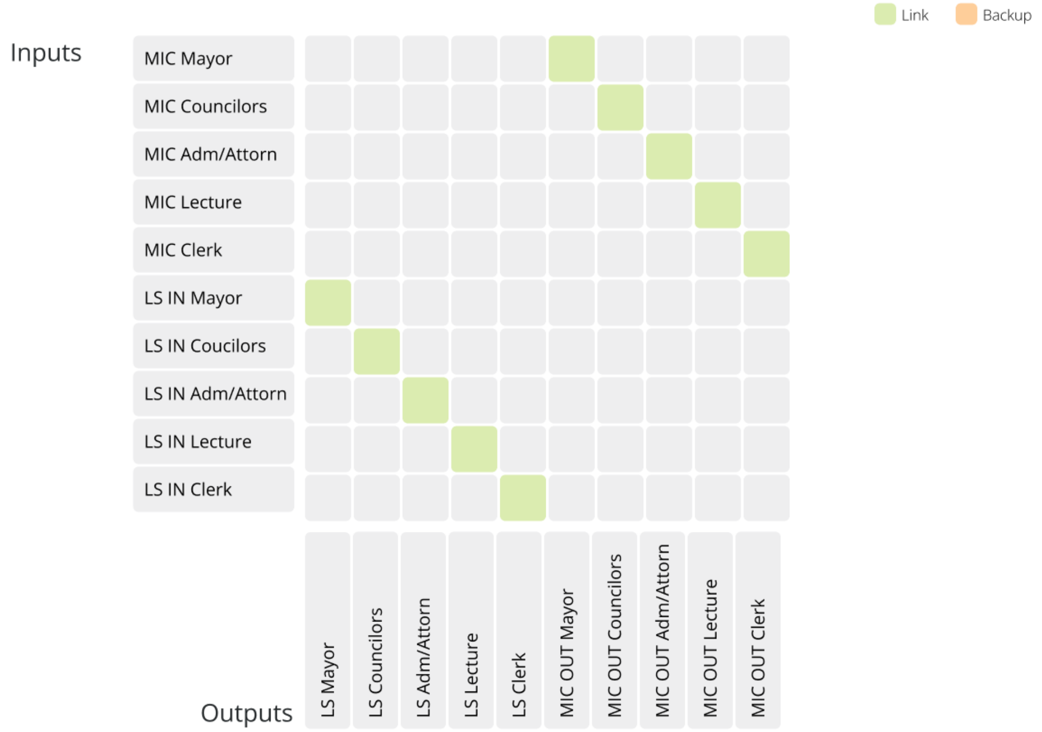
- Integration towards Remote Simultaneous Interpretation platforms

As this functionality is part of the Confero software, all functionality runs on the Plixus engine. There is no need for a dedicated application installed on a PC. The complete user interface is web-based so any computer with a browser can be used to create or adapt configuration.



Operators: activate
Technicians: edit

Below a couple of audio routing examples.

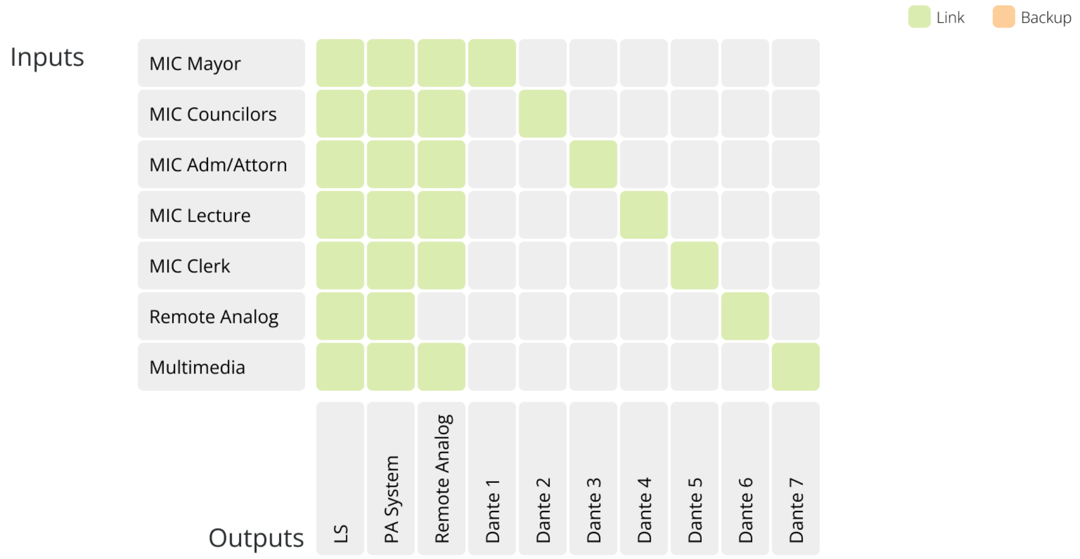


Example above shows a city council where the number of audio channels is reduced by grouping several stations into functional groups and have control over the audio of those groups of units.

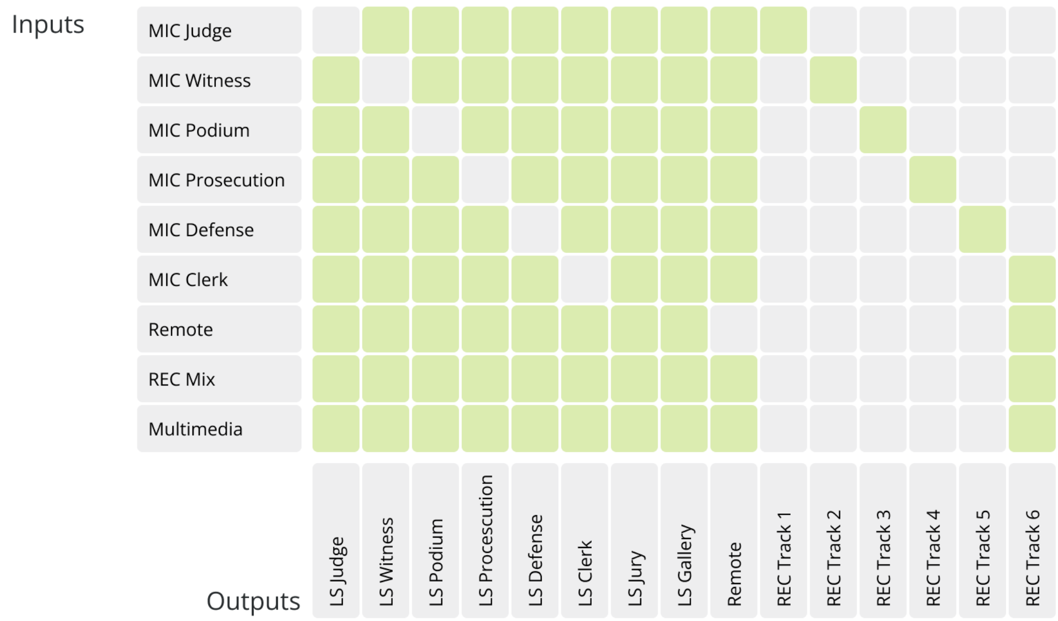
Eg. The MIC Councilors group is a mix of all elected officials, whereas the MIC Mayor is the chairperson station. The different groups are routed to Dante channels for external processing and make fine grained adjusted possible. Visa versa Dante channels can be routed to the loudspeakers of individual groups.

Different to the DSP mode Confero Audio offers the mix between integration capabilities towards DSP, Teams / Zooms, recording capabilities while economizing on DSP resources and needed channels.

When for this exact same project all audio can be handled in the conference solutions, it is still possible to provide the different groups on separate Dante channels for use in other applications.



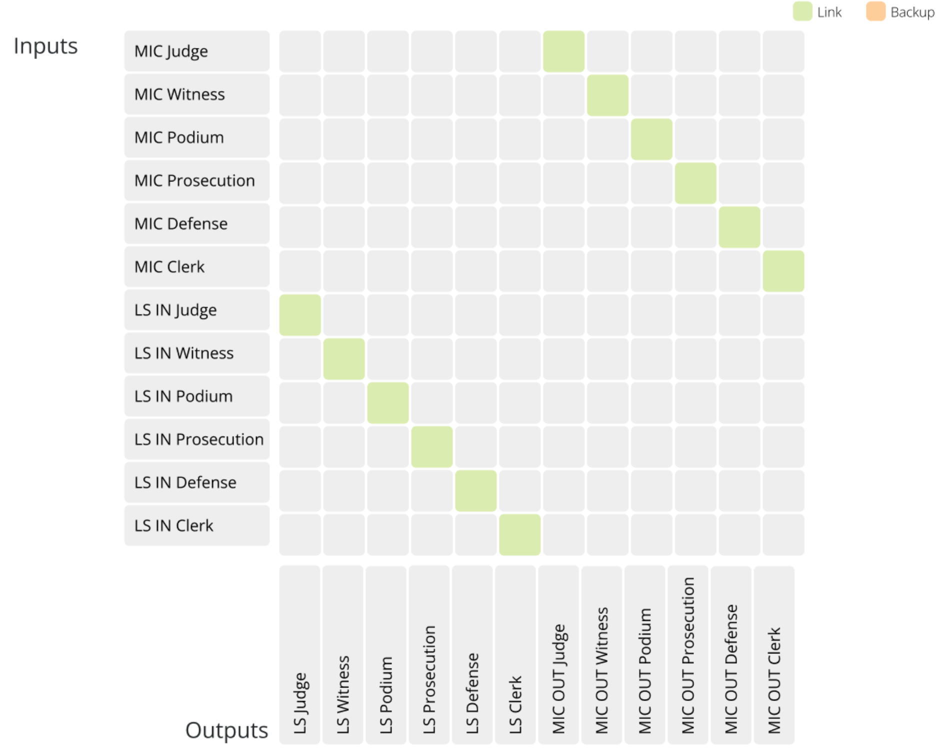
Below example is a Court application:



Here again groups of microphones and mix-minus channels are created internally in the conference solution, while still providing the routing of the different groups as individual Dante channels to send to recording applications.

Or do the same releasing the microphones into different groups to Dante channels, do the audio processing, matrix mixing in a DSP of choise and return Dante channels to feed into the loudspeakers of the microphone groups.

This is shown in pictures below.



DSP design file

Matrix mixer

API Expansion

Full Control of all Televic components and features are now available. The list is comprehensive and the possibilities are endless. Televic hardware design and audio performance has now been recognized as new solution for hybrid meetings in many applications. Third party control is required for many corporate, education and court applications.

API Partners and Modules

Televic continues to expand their control to new platforms world wide. The current list of DSP and Control partners have Televic modules:

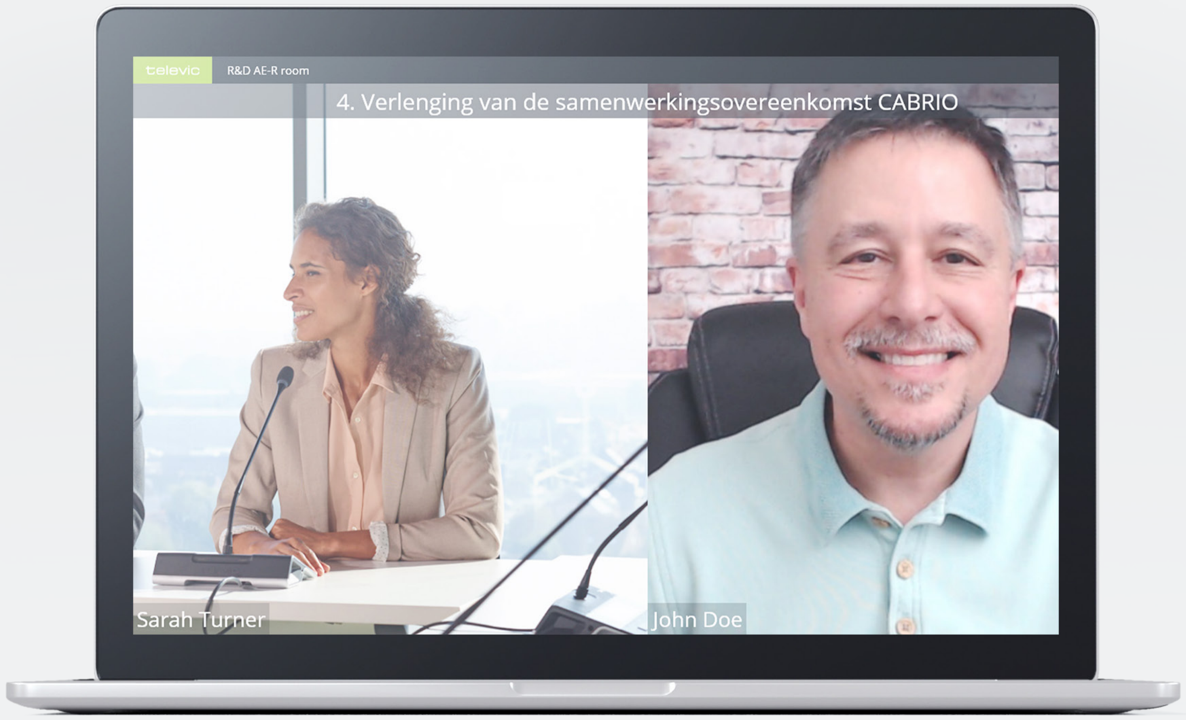
QSC biamp. Symetrix Extron. CRESTRON

Hybrid meetings are here to stay

Teams/Zoom Integration

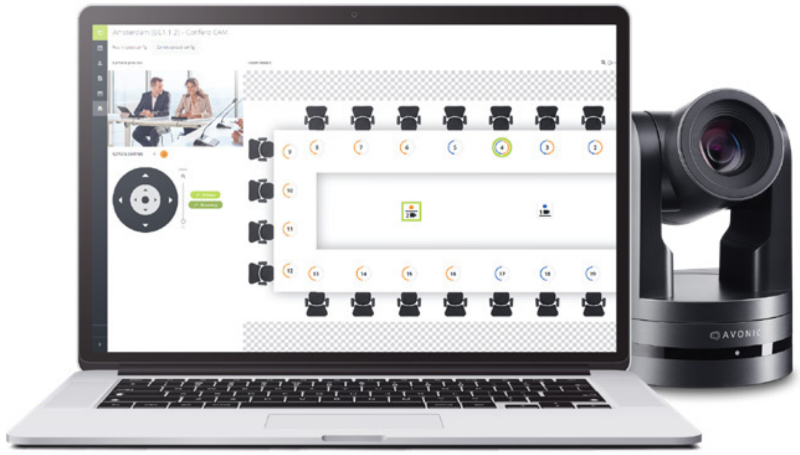
Many Televic systems have been interfaced with computers for Teams and Zoom via either the balanced analog in/out or Dante channels. Modes within the Televic system allow “distance conferencing” or “hybrid conferencing”

Depending on the number of mics open required, room acoustics and remote platform, many times a standard Televic system is the only thing needed allowing the AEC from the platform to manage the meeting. If the requirement are large numbers of open microphones, the DSP mode to use external DSP and AEC is recommended. Televic can now provide many options with wired and wireless solutions.



Confero Cam

In addition to third party camera controls with our speech detection API, Televic Confero Cam works natively with Televic systems. Expandable from one simple wide camera shot to 8 PTZ 4K HD cameras are all possible with Confero cam.



Unicos TT

Expanding beyond audio only, the Unicos TT allows all new features in this document in addition to having a bulti in Webcam on each unit and the capability to view up to 6 high def 1080P video feeds on the same standard shielded Cat 5 cable. Configurations to make a simple "Hybrid meeting station" to more advanced features to include voting, documents and more is the power of Unicos. Able to mix with audio only stations, room designs with line of site issues, no place to put a projector or video panel in a room and personal communication with someone on Teams/Zoom from an all in one station is what Unicos TT provides.

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