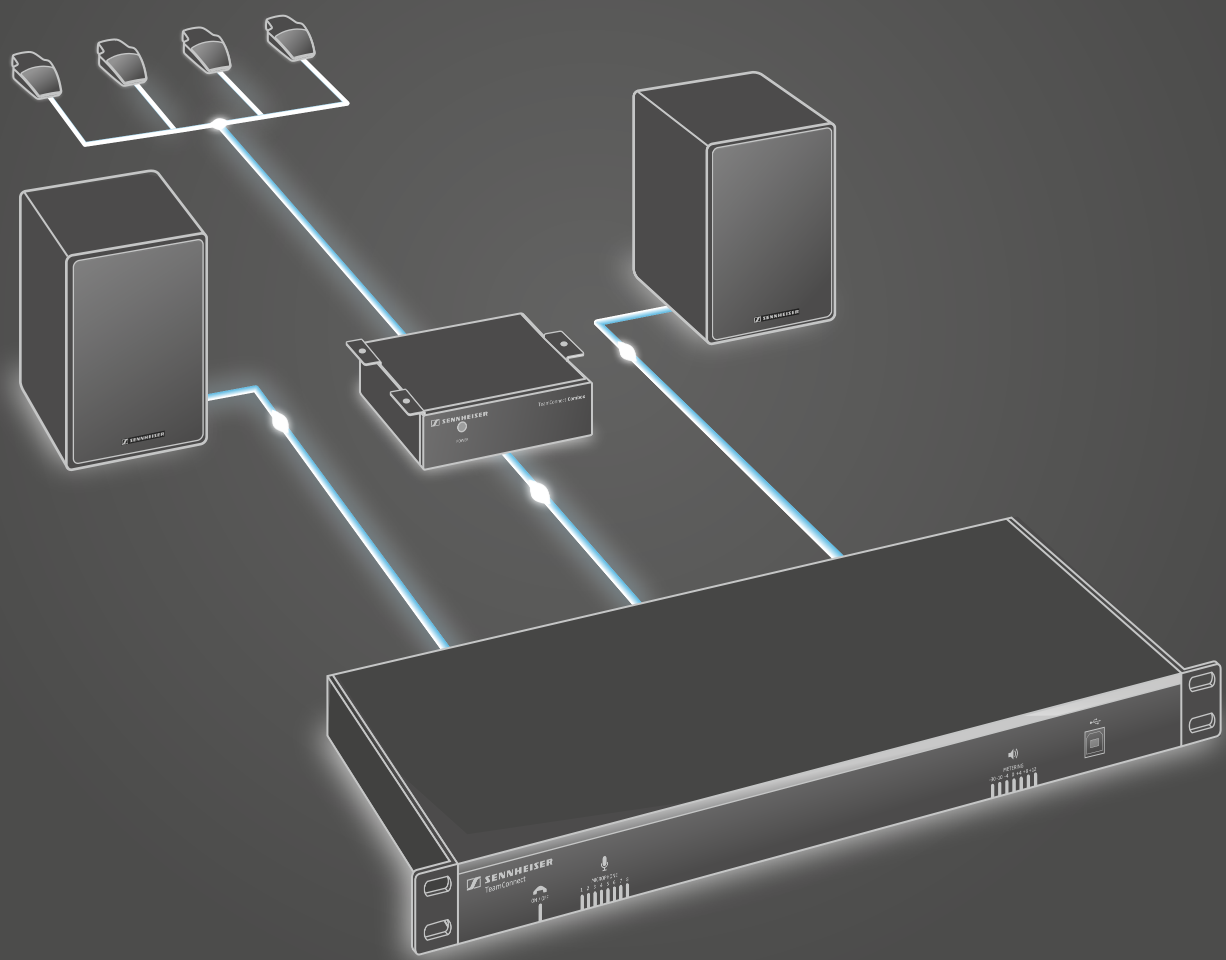
TeamConnect

Audio Configuration



TI IS-1006 v2.0

SENNHEISER

Contents

Introduction

This document is an addition to the **TeamConnect instruction manual**. The instruction manual is available for download on the TeamConnect product page at **www.sennheiser.com** and provides detailed information on how to set up, connect and configure the complete TeamConnect system.

This audio configuration document includes additional technical information which is required to set up and configure the audio settings of the TeamConnect system in order to achieve the best possible speech intelligibility.

Mixer Console

The **Mixer Console** is the central tool for routing the audio signals of the inputs to the desired outputs of the TeamConnect system. It is accessed via the Configuration Manager software on a computer connected to the SL TeamConnect CU1.

For detailed information on installing and using the Configuration Manager software and connecting a computer to the TeamConnect system refer to the TeamConnect instruction manual on the TeamConnect product page at www.sennheiser.com.

To open the Mixer Console:

- Connect a computer to the SL TeamConnect CU1.
- Start the Configuration Manager software.
- Click on the **Mixer Console** icon in the center of the configuration screen.

  The **Mixer Console** configuration window opens.

The inputs are listed on the left and the outputs are listed along the top of the matrix. The additional numbers and letters along the labels of the inputs and outputs show the cross point coordinates.

For better viewing we recommend labelling the used inputs and outputs and removing the labels from all unused inputs and outputs. That way only the channels which are actually used are labelled in the **Mixer Console**.

For information on labelling the inputs and outputs see page 6.

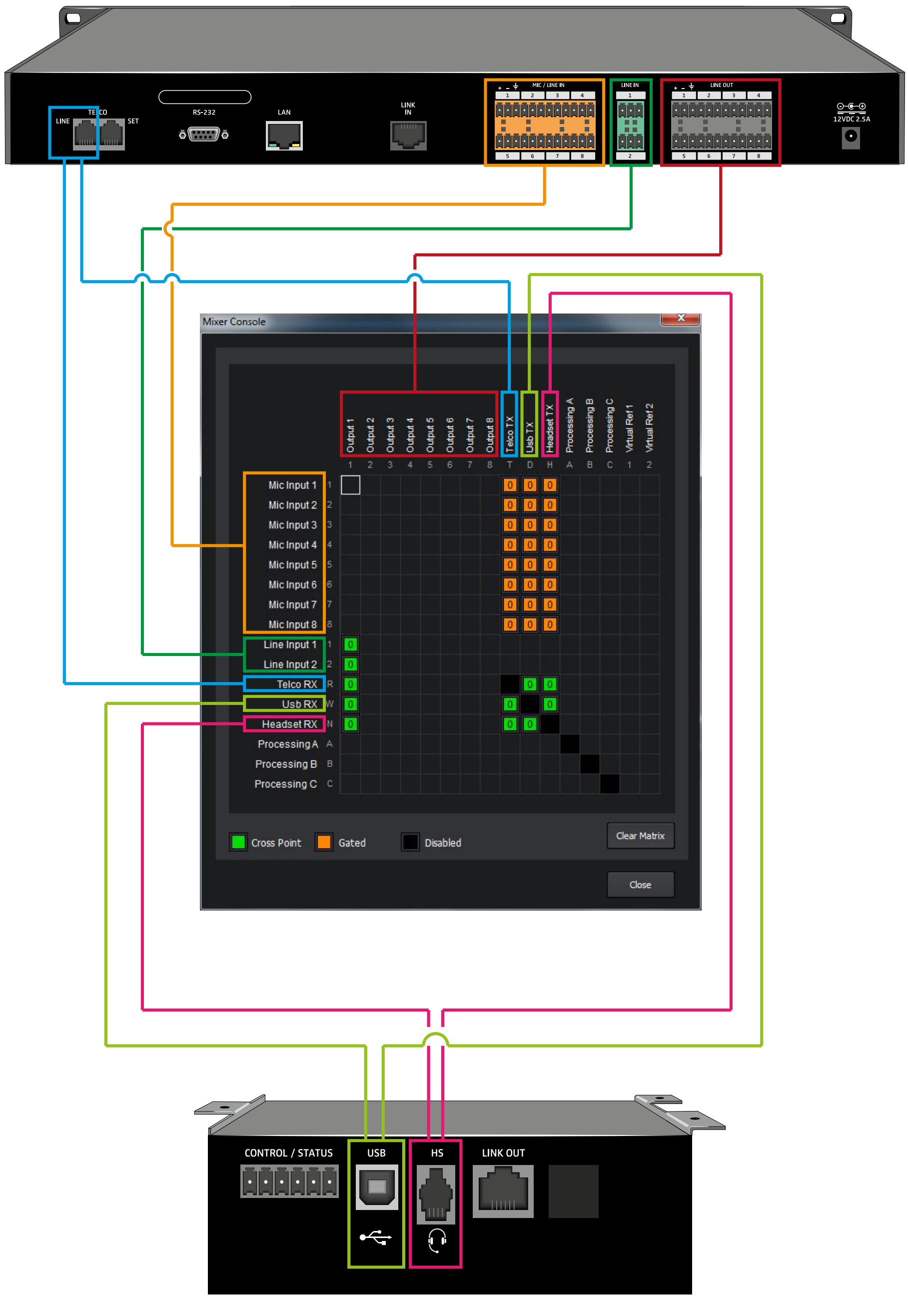
To route an input to an output:

- Click on the intersection between the desired input and the desired output in the routing matrix.

  The selected input is routed to the selected output. The cross points of the MIC inputs are displayed in orange. The cross points of the other inputs are displayed in green.

Overview of the inputs and outputs

The following illustration displays an overview of the inputs and outputs of the TeamConnect system indicating the corresponding channels in the **Mixer Console** of the Configuration Manager software.

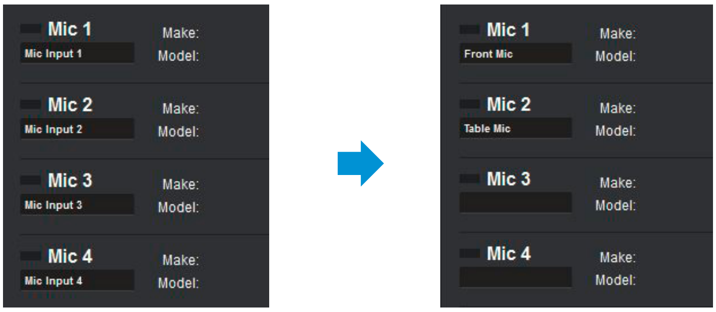


Labelling the inputs and outputs

After connecting all devices of your meeting room setup we recommend labelling all the inputs and outputs which are used. That makes the system configuration in the **Mixer Console** easier due to simplified viewing.

To label an input or an output:

- Open the desired input or output configuration window (e.g. **Mics** or **Outputs**).
- Click into the name field of the desired channel.
- Enter a name for each used channel and remove the labels from all unused channels.



After labelling, all the used inputs and outputs can be identified easier in both the **Configuration Screen** and the **Mixer Console**.

MIC setup

Sennheiser offers a variety of microphones which can be used with the TeamConnect system. Which microphone is the most suitable depends on the desired application and room setup.

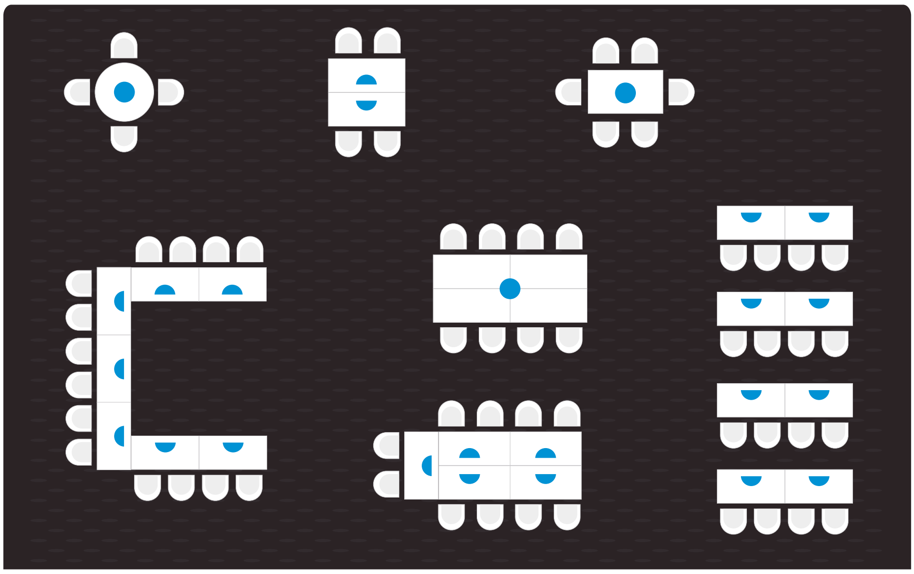
For detailed information on the Sennheiser SpeechLine IS microphones and accessories refer to the instruction manual of the SpeechLine IS microphone series at www.sennheiser.com.

Choosing microphones

Depending on the meeting room setup you can either use omni-directional or directional microphones. The following illustration shows various possible table and chair setups in meeting rooms.

- The blue circle represents an omni-directional microphone
- The blue semicircle represents a directional microphone

Possible meeting room table and chair setups



For the TeamConnect system we recommend using the following microphones from the Sennheiser SpeechLine IS microphone series:

Recommended Sennheiser SpeechLine microphones

MEB 102 | MEB 102-L | MEB 102-L TC:
omni-directional install boundary layer microphone

MEB 104 | MEB 104-L | MEB 104-L TC:
cardioid install boundary layer microphone

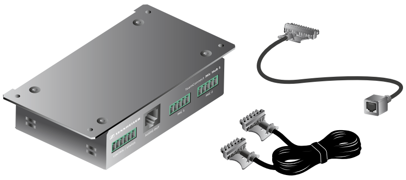
MEB 114-S | MEB 114-S TC:
cardioid on-table boundary layer microphone

Connecting and positioning microphones

The microphones positioned or installed in the room are connected to the SL TeamConnect CU1. You can connect up to eight microphones.

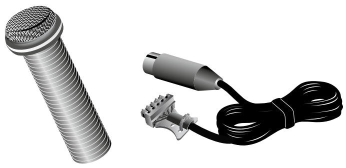
Connecting the TeamConnect microphones via the SL Mic Hub 1

The SL Mic Hub 1 allows for a very easy and quick connection of the TeamConnect microphones and control buttons, thus making the entire installation of the TeamConnect system very efficient.



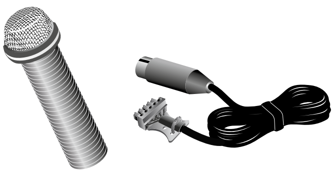
SL Mic Hub 1

- microphone hub for connecting the TeamConnect microphones and logic control buttons
- includes adapter cables for the connection to the SL TeamConnect CU1 and the SL TeamConnect CB1
- one SL Mic Hub 1 allows the connection of 4 microphones
- two SL Mic Hub 1 devices can be cascaded to enable the connection of up to 8 microphones



MEB 102-L TC

- omni-directional install boundary layer microphone
- includes XLR-5 to 5-pin terminal connector adapter cable (length: 3 m)

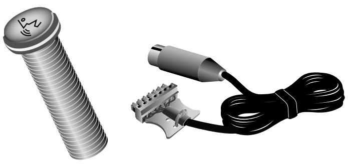


MEB 104-L TC

- cardioid install boundary layer microphone
- includes XLR-5 to 5-pin terminal connector adapter cable (length: 3 m)

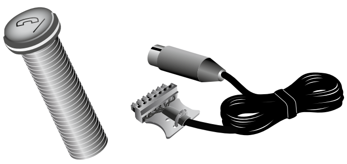
MEB 114-S TC

- cardioid on-table boundary layer microphone
- with fixed 5-pin terminal connector cable (length: 3 m)



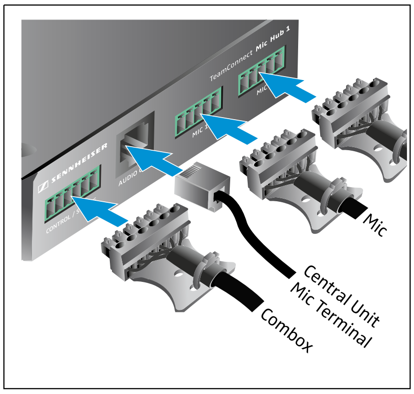
MAS 1 TC

- logic control button for mute function
- includes XLR-5 to 7-pin terminal connector adapter cable (length: 3 m)



MAS 2 TC

- logic control button for telephone hook function
- includes XLR-5 to 6-pin terminal connector adapter cable (length: 3 m)



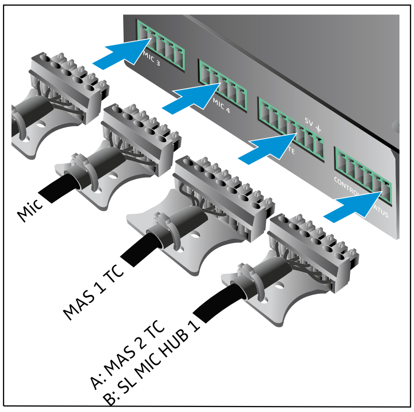
To connect the TeamConnect microphones and accessories:

- ▶ Connect the AUDIO OUT socket of the SL Mic Hub 1 to the MIC / LINE IN terminal of the SL TeamConnect CU1 using a shielded Cat 5 cable with RJ-45 connectors and the supplied RJ-45 to 12-pin terminal adapter.

- ▶ Connect the CONTROL / STATUS port of the SL Mic Hub 1 to the CONTROL / STATUS port of the SL TeamConnect CB 1 using the supplied 6-pin to 6-pin terminal connector cable.

- ▶ Connect the TeamConnect microphones to the MIC 1 to MIC 4 sockets of the SL Mic Hub 1.

  You can connect up to 4 microphones to the SL Mic Hub 1.



- ▶ Connect the MAS 1 TC mute button to the MUTE socket of the SL Mic Hub 1 using the supplied XLR-5 to 7-pin terminal connector cable.

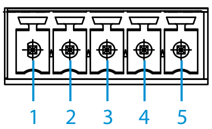
- ▶ Connect the MAS 2 TC hook button to the second CONTROL / STATUS socket of the SL Mic Hub 1 using the supplied XLR-5 to 6-pin terminal connector cable.

  OR, if you are using a second SL Mic Hub 1 for using four additional microphones:

- ▶ Connect the CONTROL / STATUS socket of the second SL Mic Hub 1 to the CONTROL / STATUS socket of the first SL Mic Hub 1 using the supplied 6-pin to 6-pin terminal connector cable.

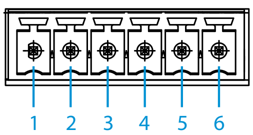
Pin allocation of the SL Mic Hub 1

MIC 1 - MIC 4 sockets



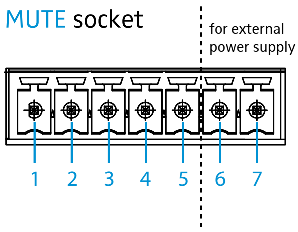
1 mute
2 mute status
3 ground
4 audio +
5 audio -

CONTROL / STATUS sockets



1 mute
2 mute status
3 hook
4 hook status
5 +5V
6 ground

MUTE socket



1 +5V out
2 mute
3 red
4 green
5 ground
6 +5V in
7 ground

Power supply

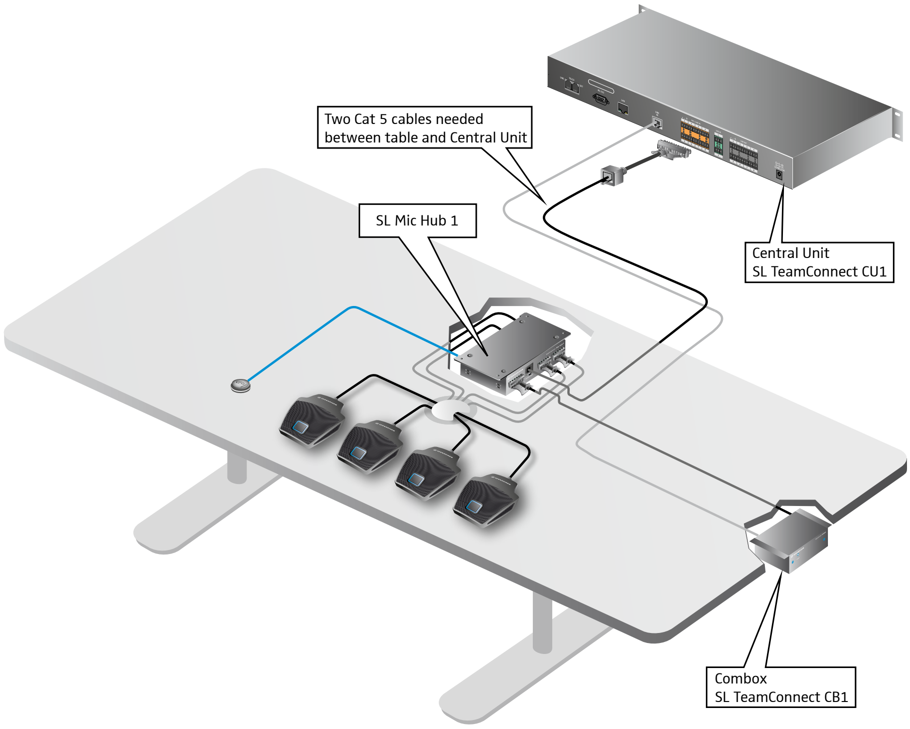
The SL Mic Hub 1 ist connected to the TeamConnect Combox CB1 via a 6-pin terminal connector cable. The Combox delivers the necessary power supply to operate the bi-color status LEDs of the mute button MAS 1 TC and the hook button MAS 2 TC.

The Combox supplies sufficient power for up to 5 MAS 1/2 buttons. If you want to use more than 5 buttons in your setup, you need to connect an external power supply (5 V, >100 mA, recommended Sennheiser Power supply Art.-No. 534480) to pin 6 and 7 of the MUTE socket.

TeamConnect system setups with the SL Mic Hub 1

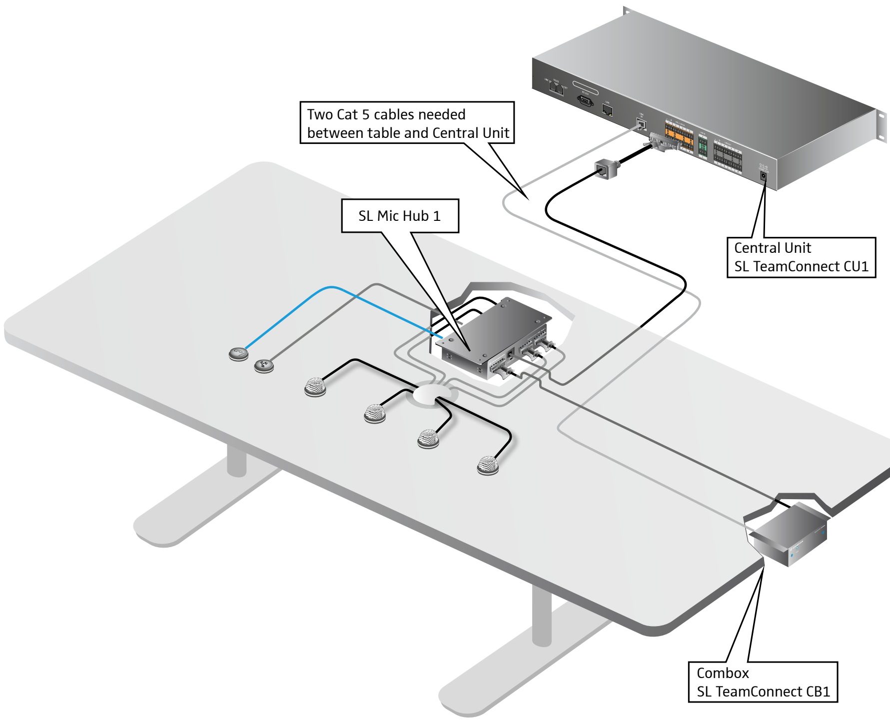
Application A. up to 4 on-table microphones

- 1 x SL Mic Hub 1
- 4 x MEB 114-S TC
- 1 x MAS 2 TC



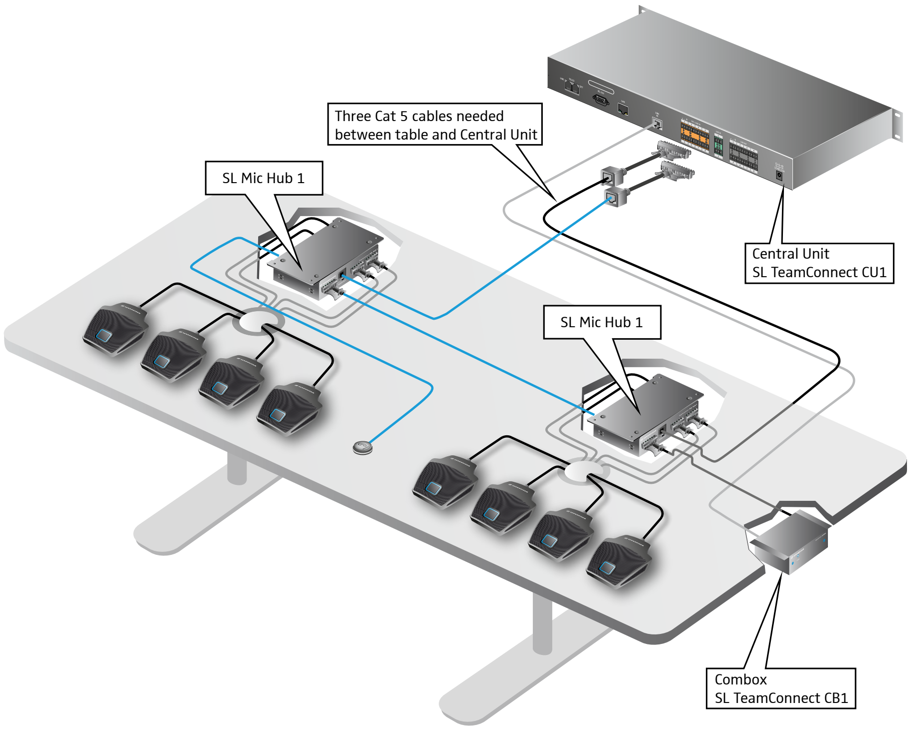
Application B: up to 4 in-table microphones

- 1 x SL Mic Hub 1
- 4 x MEB 102-L TC or MEB 104-L TC
- 1 x MAS 1 TC
- 1 x MAS 2 TC



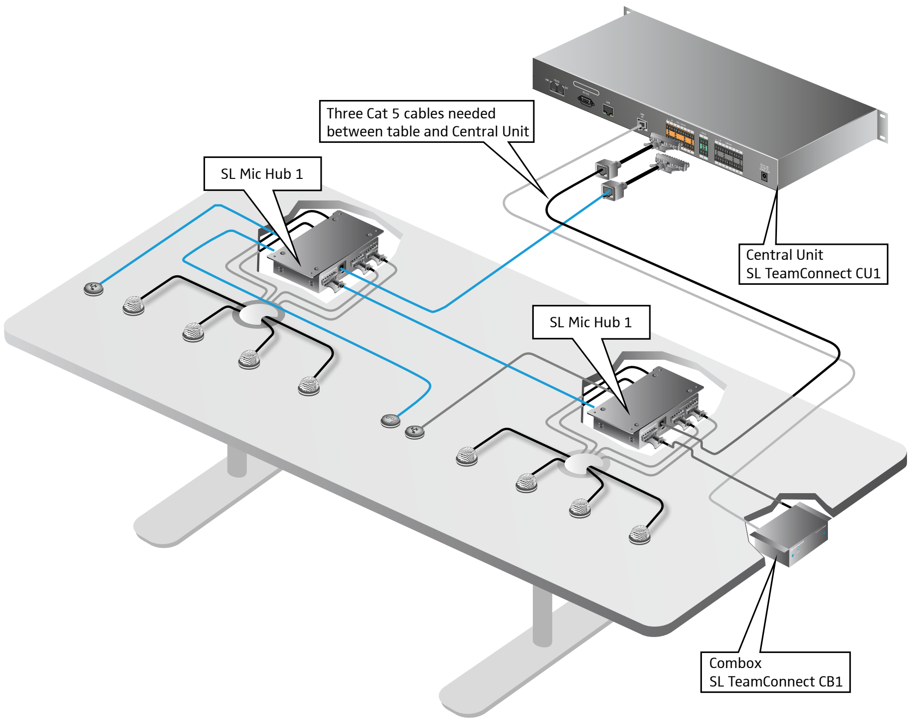
Application C: up to 8 on-table microphones

- 2 x SL Mic Hub 1
- 8 x MEB 114-S TC
- 1 x MAS 2 TC



Application D: up to 8 in-table microphones

- 2 x SL Mic Hub 1
- 8 x MEB 102-L TC or MEB 104-L TC
- 2 x MAS 1 TC
- 1 x MAS 2 TC

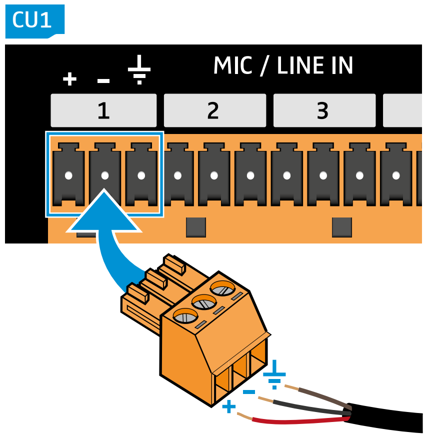


Connecting microphones with individual cabling

If you are not using the TeamConnect Mic Hub cabling system (see "Connecting the TeamConnect microphones via the SL Mic Hub 1" on page 8), you can also connect any regular microphone with individual cabling to the SL TeamConnect CU1.

To connect a microphone to the SL TeamConnect CU1:

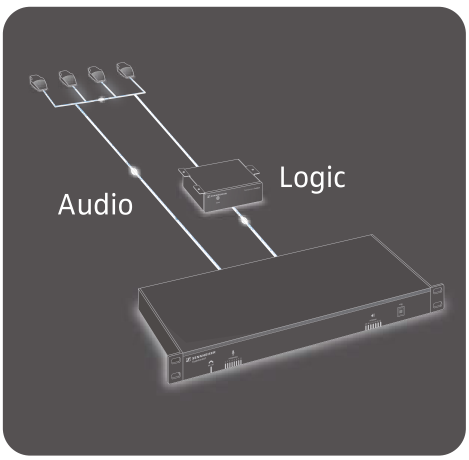
▶ Connect the microphone to the orange MIC/LINE IN terminal with the supplied terminal connector observing the correct pin allocation.

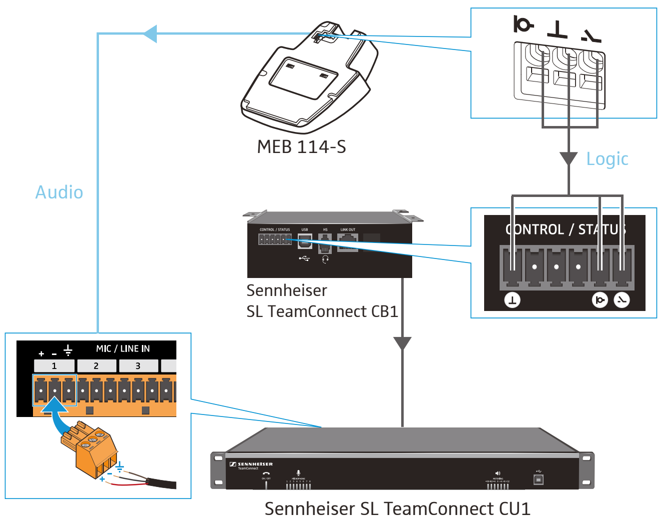


Connecting microphones with a logic circuit

Some of the Sennheiser SpeechLine microphones have an integrated logic circuit, e.g. the MEB 114-S.

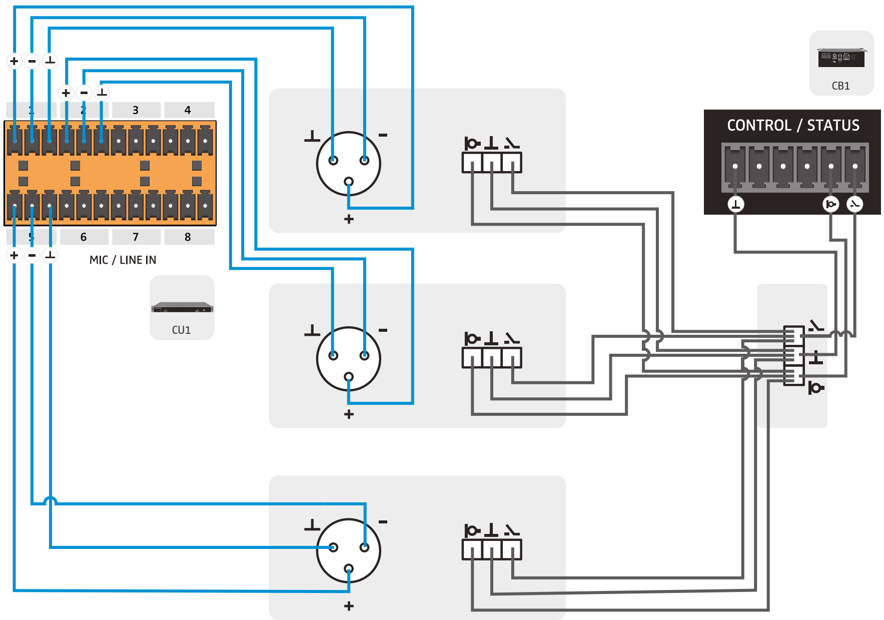
The XLR socket of the microphone must be connected to the SL TeamConnect CU1. The logic terminal of the microphone must be connected to the SL TeamConnect CB1 as shown in the figure below.





To connect a microphone with a logic circuit:

▶ Observe the following wiring.

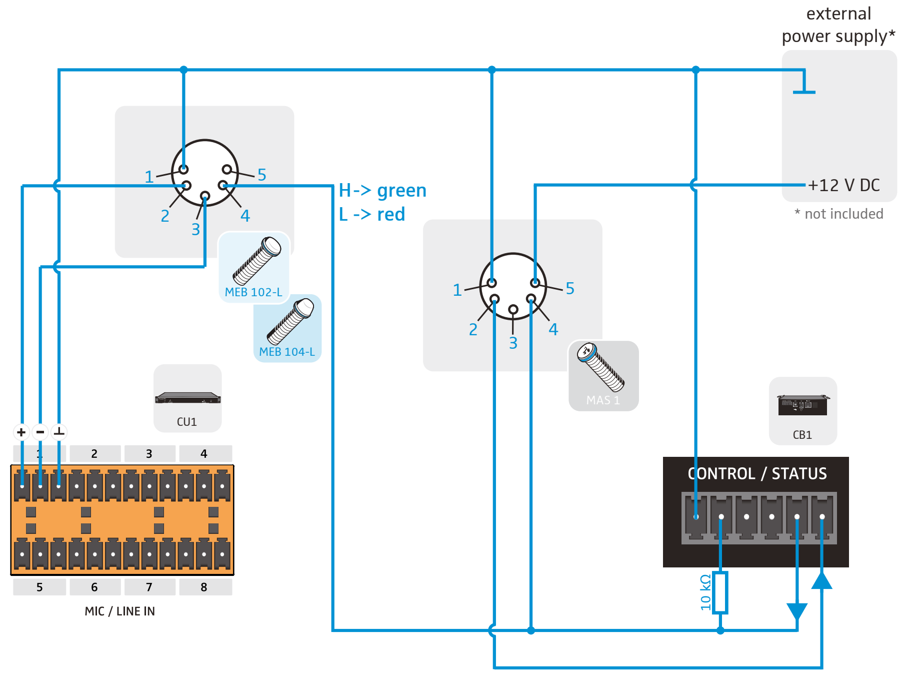


The figure shows an example with three microphones with an XLR 3 connector and a logic circuit.

Connecting microphones with an XLR 5 connector

To connect a microphone with an XLR 5 connector:

▶ Observe the following wiring.



The figure shows an example with one microphone with an XLR 5 connector and the microphone button MAS 1.

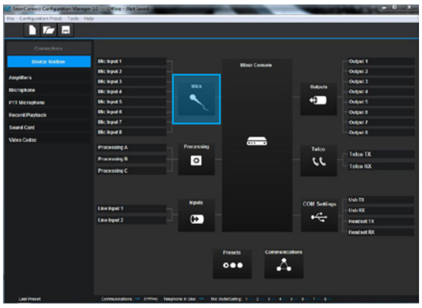
Configuring microphone settings

The audio settings of the connected microphones are configured in the **Mics** area of the Configuration Manager software.

For detailed information on installing and using the Configuration Manager software refer to the TeamConnect instruction manual on the TeamConnect product page at www.sennheiser.com.

Level and gain settings

In the **Level** tab of the **Mics** configuration window you can configure the level and gain settings of the microphone inputs to adjust the correct volume and sensitivity of the connected microphones.



To adjust the level and gain settings of the connected microphones:

- Open the **Mics** configuration window of the Configuration Manager software.
- Navigate to the **Level** tab.

Adjusting the gain settings

To calibrate the gain settings:

- Start talking into the microphone with an average distance of normal usage to the microphone.
- Adjust the **Coarse Gain** until the meter displays +6 dB.
- Then, adjust the **Fine Gain**.
- Enable the **P Pwr** function if the connected microphone requires phantom power.

Using the ALC function

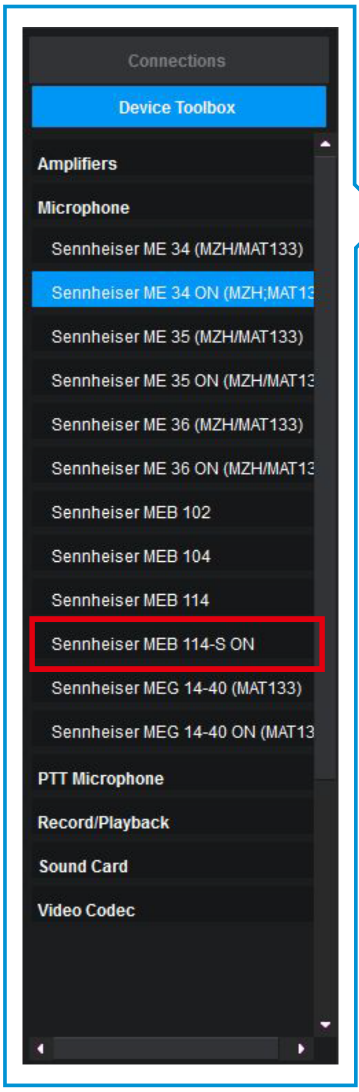
The Auto Level Control (**ALC**) function is a compander circuit which maintains a consistent level on the selected channel.

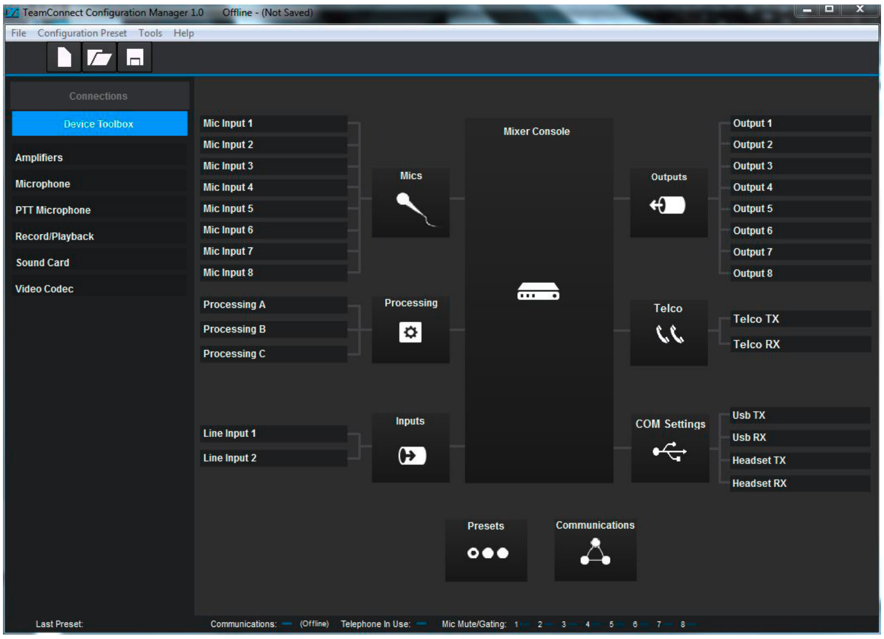
- Activate the **ALC** function only if the microphone is not locally reinforced in the room.
- Activate the **ALC** function only for teleconferences.

Using the Device Toolbox

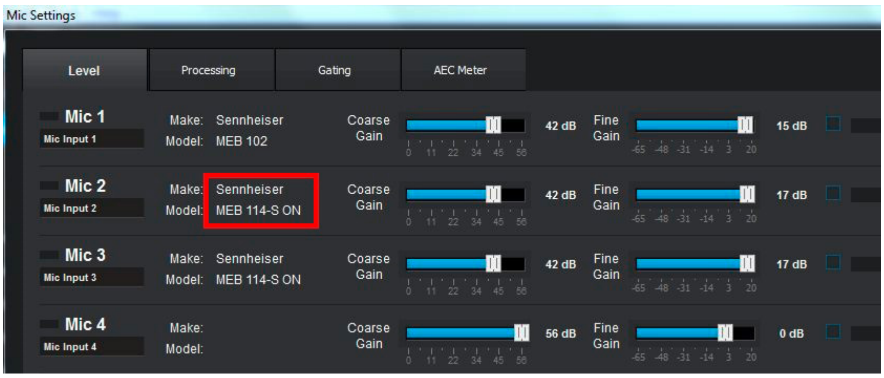
If you are using a microphone which has a preset in the **Device Toolbox**:

▶ Navigate to the **Configuration Screen**.
Open the **Device Toolbox** in the side bar.



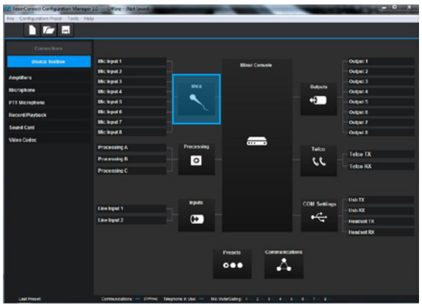


▶ Drag and drop the desired microphone preset to the desired microphone channel in the **Configuration Screen**.
In the Mics configuration window the preset is displayed under **Make** and **Model** next to the label of the microphone channel. The gain settings are adjusted according to the selected preset.



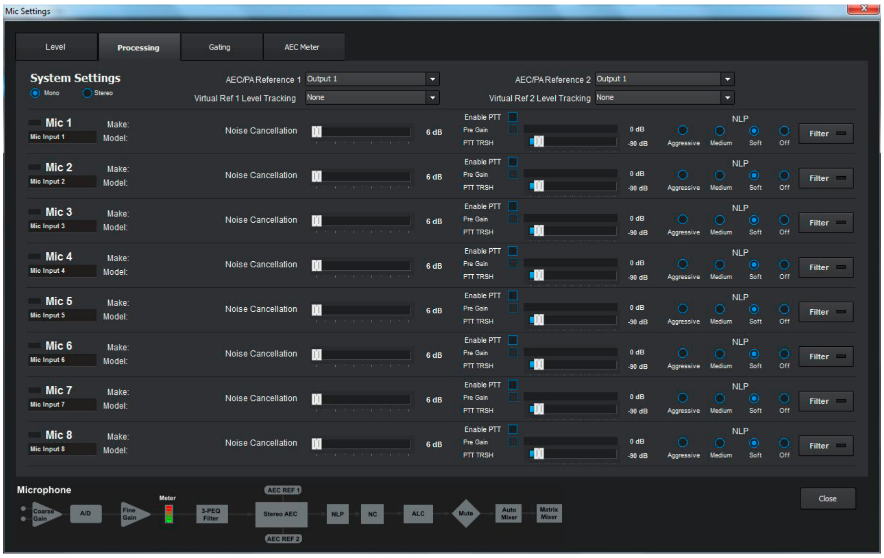
Processing settings and acoustic echo cancellation

In the **Processing** tab of the **Mics** configuration window you can configure audio settings for noise and echo cancellation and audio filters. This is done to reduce transmitted ambient noise and enhance the actual audio signal.



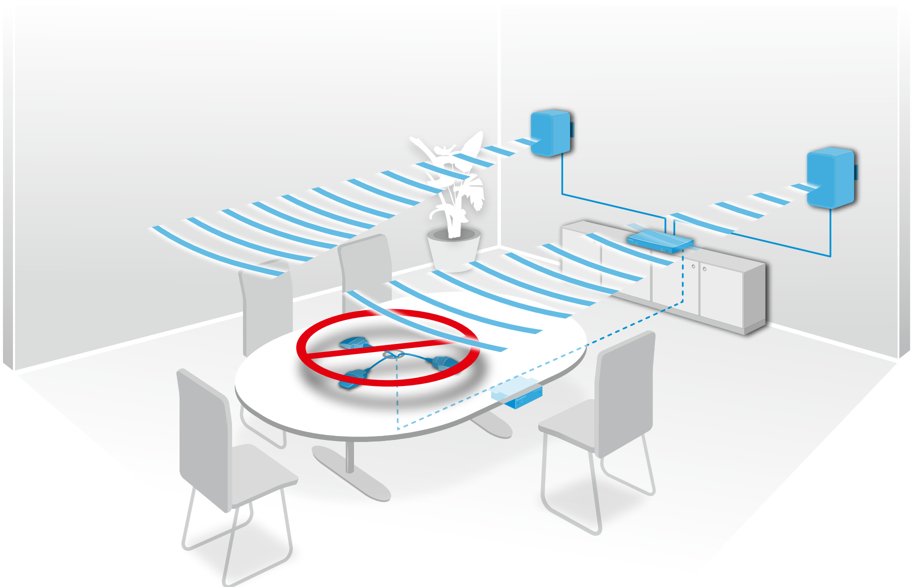
To adjust the processing settings of the connected microphones:

- Open the **Mics** configuration window of the Configuration Manager software.
- Navigate to the **Processing** tab.



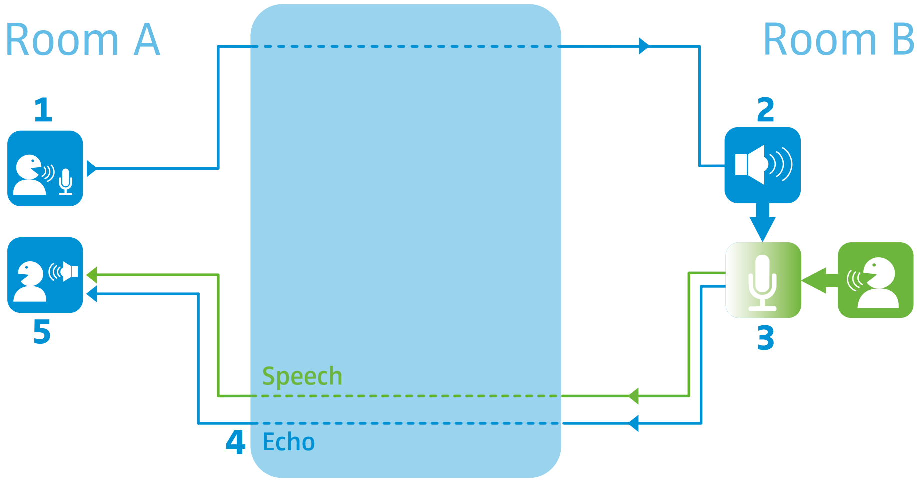
Using acoustic echo cancellation (AEC)

A common but unwanted phenomenon in teleconferences is acoustic echo. It is generated when the audio coming out of a loudspeaker is picked up by the microphone(s) in the meeting room and then re-sent to the far end of the teleconference. This effect can be prevented with the TeamConnect system.



How acoustic echo is generated

This is how the acoustic echo is generated:



1. The person in Room A is talking. The microphone picks up the audio and transmits it to Room B.
2. The people in Room B hear the audio signal from Room A, which is the desired effect.
3. However, the audio signal coming from Room A is also picked up by the microphone in Room B via a direct path from the loudspeaker and indirect paths after multiple reflections off of the surfaces of the room.
4. That audio (echo), along with the local speech from Room B, is sent back to Room A.
5. The person in Room A will hear an echo of themself whenever they speak. This is the acoustic echo.

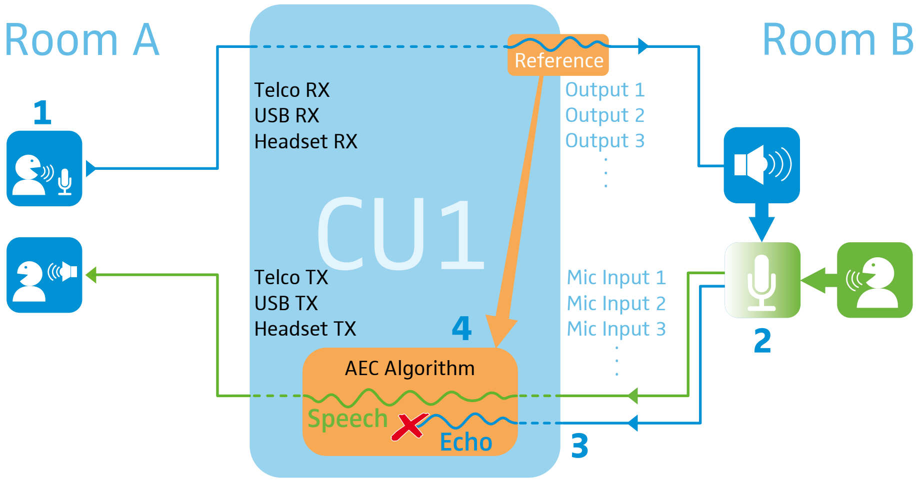
Acoustic echo is generated at the distant site and is apparent at all end points, i.e. wherever the microphone is routed.

Acoustic echo requires echo cancellation at the distant site.

One way to troubleshoot a system to determine if the echo you are hearing is an acoustic echo from the distant site is to have the distant site mute its microphones. If the echo disappears, it is probably acoustic echo and they need to invest in an acoustic echo canceller or recalibrate their existing system.

Acoustic echo cancellation with output reference

This is how the acoustic echo is cancelled out in the SL TeamConnect CU1:



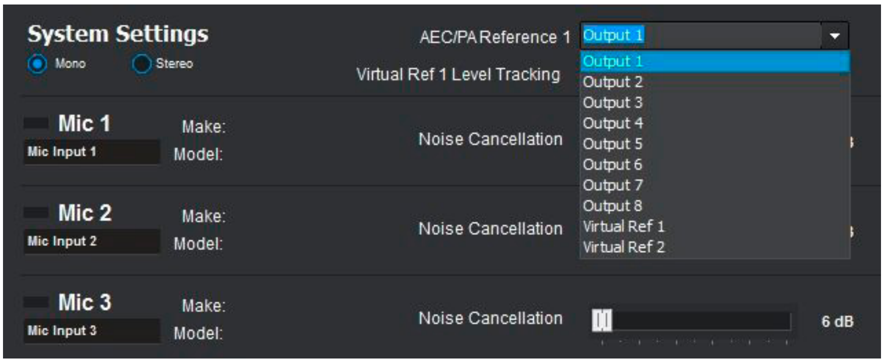
1. The audio signal of the person talking in Room A is received by the Telco RX, USB RX or Headset RX channels of the SL TeamConnect CU1 in Room B.
2. The microphone in Room B picks up the local talk of the person in Room B and the audio signal of Room A coming from the loudspeaker in Room B.
3. This mixed audio signal enters the SL TeamConnect CU1 via the Mic Input channels.
4. The AEC algorithm uses the signal of the Output channel as reference and cancels this signal out of the mixed signal coming in at the Mic Input channel.

That way, the TeamConnect system will prevent the loudspeaker audio, which is picked up by the microphone(s), to be re-transmitted to the far end of the teleconference.

In order to make that happen, you need to to identify the output channel which is used as the reference for the echo cancellation algorithm in the Configuration Manager software.

To set the output reference for echo cancellation:

▶ Open the **AEC/PA Reference 1** dropdown menu in the **Processing** tab of the **Mics** configuration window.



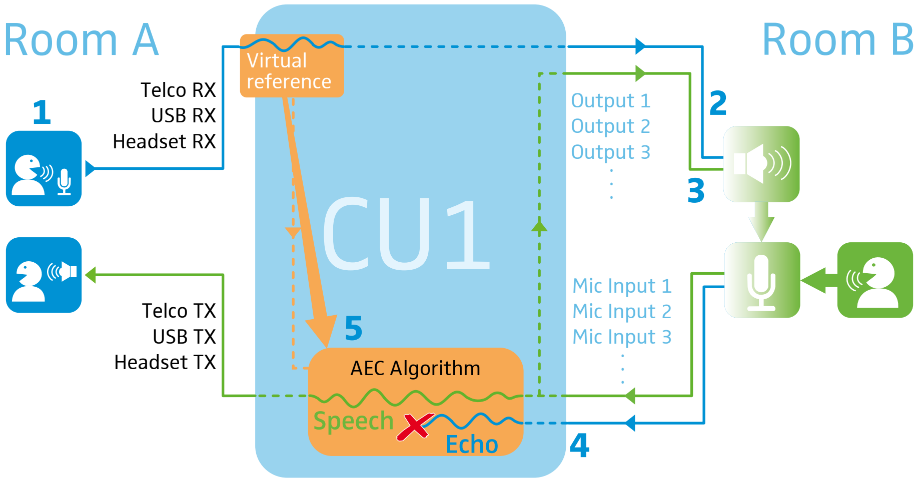
▶ Select the desired output channel from the channels Output 1 to Output 8.

The signal of the selected output channel is used as the reference signal for the echo cancellation.

For mono applications you only need to select the reference output channel for **AEC/PA Reference 1**. For stereo applications you also need to select a channel for **AEC/PA Reference 2** and activate the **Stereo** radio button.

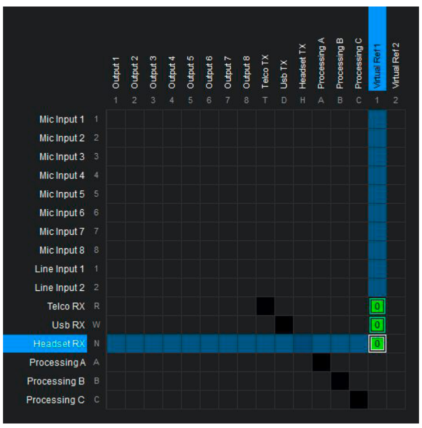
Acoustic echo cancellation with virtual reference

A second possible scenario for acoustic echo cancellation is a meeting room setup with the microphone signal being reinforced locally in the meeting room.



This scenario is similar to the previous scenario. The acoustic echo is generated the same way. However, the difference is that the microphone audio in Room B is reinforced locally in the meeting room via the output channels. That way the signal at the output channel of the SL TeamConnect CU1 is already a mixed audio signal and cannot serve as a reference for the acoustic echo cancellation algorithm like in the previous example. It does not deliver the correct audio signal which needs to be cancelled out.

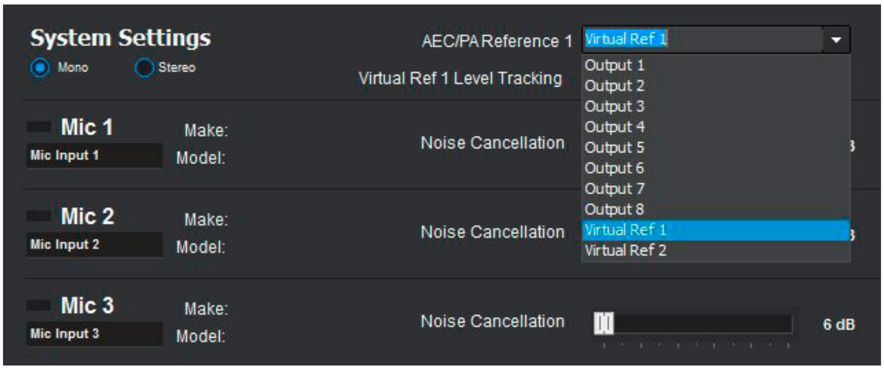
1. The audio signal of the person talking in Room A is received by the Telco RX, USB RX or Headset RX channels of the SL TeamConnect CU1 in Room B.
2. The microphone in Room B picks up the local talk of the person in Room B and the audio signal of Room A coming from the loudspeaker in Room B.
3. The microphone in Room B also picks up its own signal which is reinforced in the room and is also coming from the loudspeaker in Room B.
4. These multiple audio signals enter the SL TeamConnect CU1 via the Mic Input channels. The Output channel cannot be used as a reference here. It contains a mixed audio signal and does not deliver the single audio signal which needs to be cancelled.
5. In this case, the AEC algorithm uses the Virtual Reference channels of the Configuration Manager software as reference. These are no actual outputs of the SL TeamConnect CU1 but only virtual channels in the software, which pick up the audio signal before it arrives at the actual hardware outputs. All signals which need to be cancelled are routed to the Virtual Reference channels in the Mixer Console.



To use the Virtual Reference channels for echo cancellation:

- Navigate to the **Mixer Console** of the Configuration Manager software.
- Route all channels which need to be cancelled to the desired Virtual Reference channel.

- Navigate to the **Processing** tab of the **Mics** configuration window.
- Open the **AEC/PA Reference 1** dropdown menu.



- Select the desired Virtual Reference channel.

  The signal of the selected virtual reference channel is used as the reference signal for the echo cancellation.

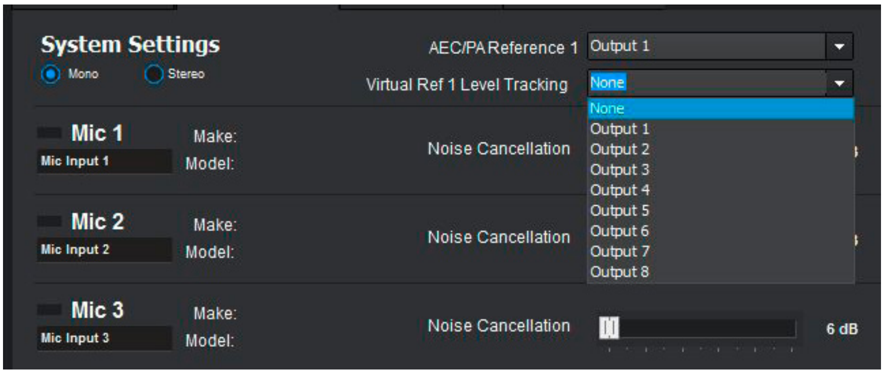
For mono applications you only need to select the reference output channel for **AEC/PA Reference 1**. For stereo applications you also need to select a channel for **AEC/PA Reference 2** and activate the **Stereo** radio button. Most meetings and conferences use mono audio signals.

Using virtual reference level tracking

If you are using the virtual reference channels for echo cancellation, the AEC process needs audio level information for its algorithm. As the virtual reference channels are not physical outputs they cannot provide level information. Therefore you need to select the output channel used for the speakers in the room as level tracking reference.

To set the virtual reference level tracking:

- Open the **Virtual Ref 1 Level Tracking** dropdown menu.



- Select the output channel which is used for the speakers in the room.

For mono applications you only need to select the reference output channel for **Virtual Ref 1 Level Tracking**. For stereo applications you also need to select a channel for **Virtual Ref 2 Level Tracking** and activate the Stereo radio button. Most meetings and conferences use mono audio signals.

Using noise cancellation

The noise cancellation algorithm looks for repeating ambient noise which is then cancelled out. This provides better speech intelligibility in conferences.

▶ Set the **Noise Cancellation** slider to 12 dB.

We recommend setting it to 12 dB. This setting will provide good noise cancellation for all standard meeting environments.

Using the PTT function

If you are using push-to-talk microphones (e.g. Sennheiser MEB 114-S), you should activate the push-to-talk function (PTT).

If the function is not activated, this might lead to incorrect behaviour of the echo cancellation every time the microphone is muted or unmuted.

Using NLP

The non-linear processing circuit is a smart suppressor that will aid the AEC circuit when it is not fully converged.

OUR RECOMMENDATION

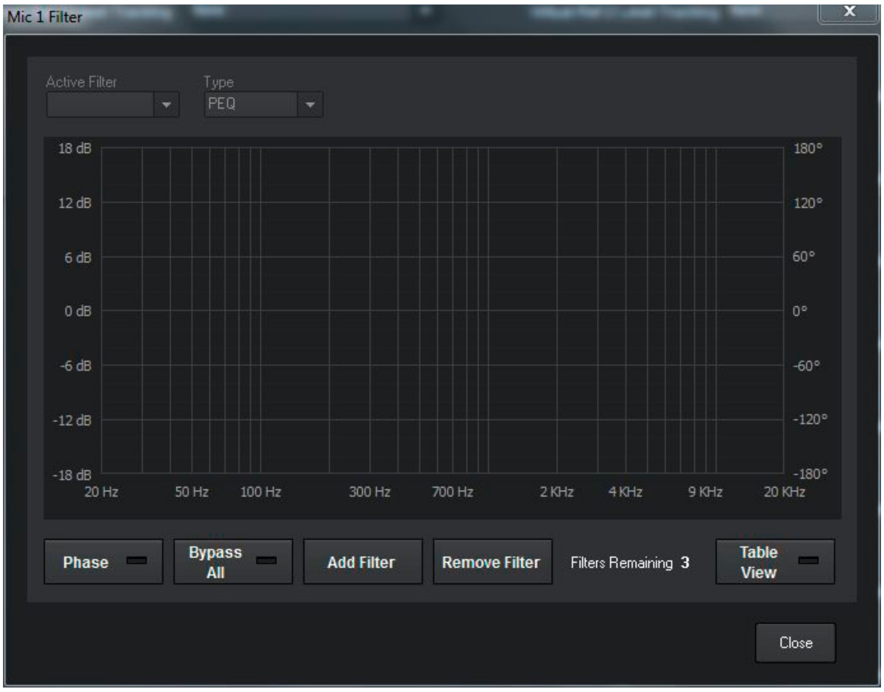
We strongly recommend leaving NLP at the default setting (soft). Only change this setting if you are advised to by an authorised Sennheiser service partner.

Using filters

Filters can be used to optimize the speech intelligibility by cutting off or emphasizing certain frequencies. For the best possible speech intelligibility and reduction of ambient noise we recommend setting a high pass and a low pass filter.

If you are using more than one microphone and you want to apply the same filters for all microphones, you can route all microphones to one of the **Processing Channels** in the **Mixer Console** and set these filters there. See "Processing channels" on page 29.

- Click on **Filter** to open the filter configuration window for the desired channel.



Setting the High Pass filter

- Click on **Add Filter**.
- Select **High Pass** from the **Type** dropdown menu.
- Enter **150** in the **Frequency** field.

Setting the Low Pass filter

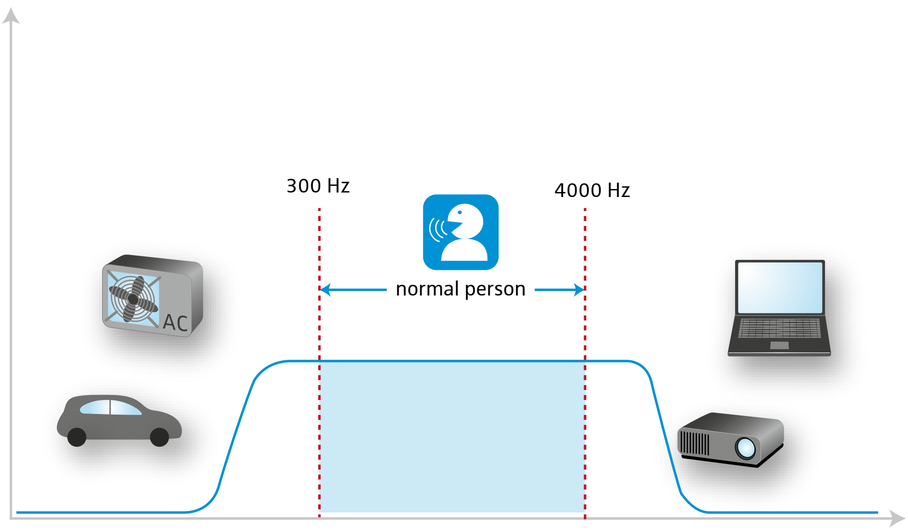
- Click on **Add Filter**.
- Select **Low Pass** from the **Type** dropdown menu.
- Enter **8000** in the **Frequency** field.

Setting these two filters results in frequencies below 150 Hz and above 8000 Hz being cut off. These could be the frequencies of possible interfering ambient noise, e.g.:

- traffic noise heard through open windows
- air conditioning
- beamer hum
- computer ventilation

These frequencies should be cut off as they might cause incorrect behaviour of the microphone gating function.

The frequency range of a communicating voice (300 - 4000 Hz) is passed through.



Gating

A gate can be used to automatically activate and deactivate the signal of a connected microphone. This will result in opening the microphone when a person is talking and closing the microphone when the person is not talking. When the microphone is closed, no ambient noise will be transmitted by the microphone.

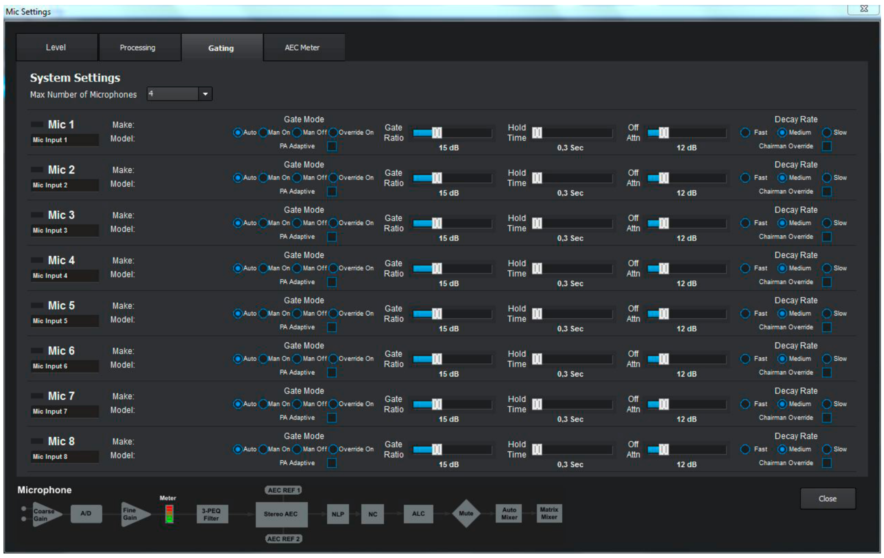
You can adjust the settings when the microphone gate should open or close in the Gating tab of the Mics configuration window.

OUR RECOMMENDATION

We recommend using the default gating settings, as they have been pre-adjusted to provide the best possible speech intelligibility.

To adjust the gating settings of the connected microphones:

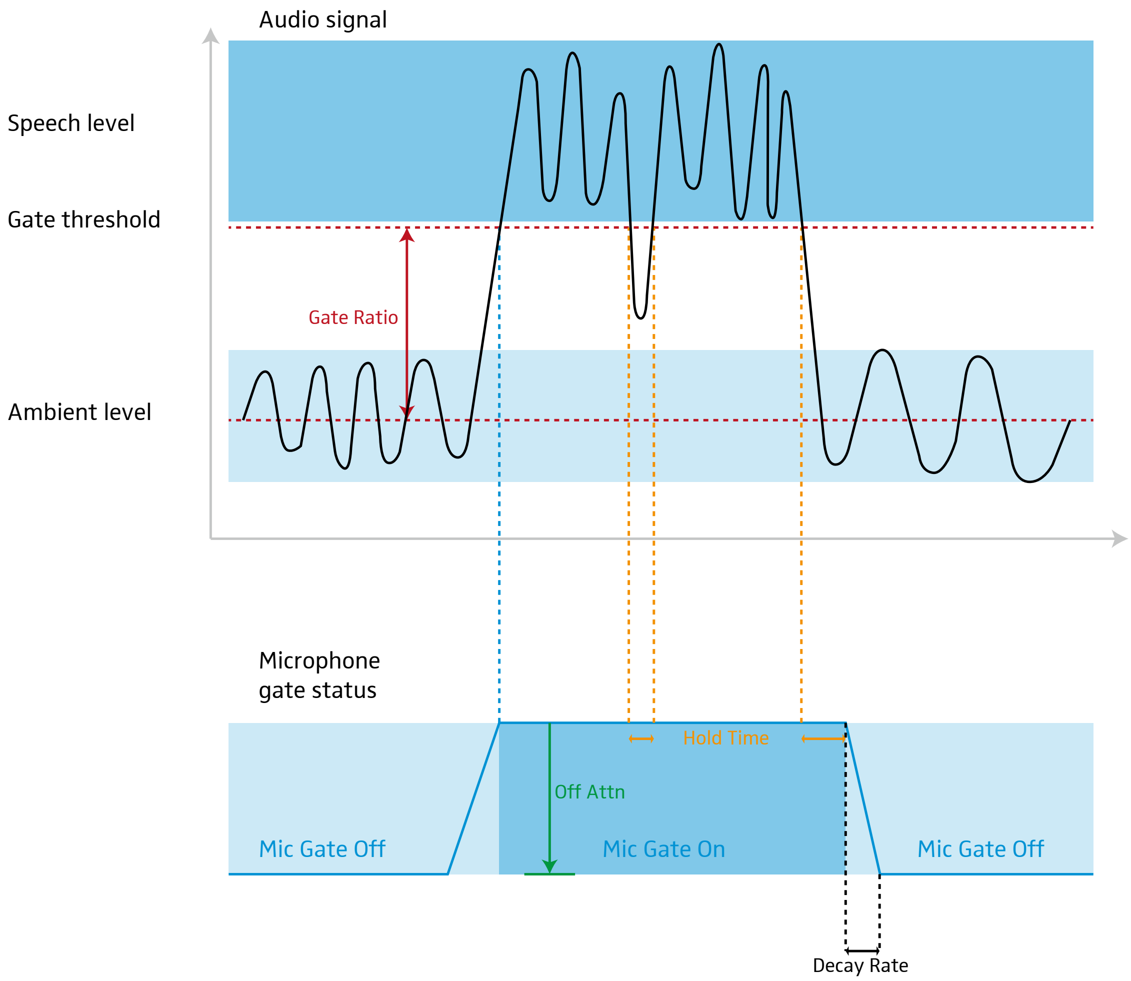
- ▶ Open the **Mics** configuration window of the Configuration Manager software.
- ▶ Navigate to the **Gating** tab.



Option	Settings
Max Number of Microphones	Defines the maximum number of microphones which are active simultaneously.
Gate Mode	• **Auto** (default): Gating depends on the maximum number of microphones. • **Man On**: Gates the channel on. The audio signal is on permanently. The channel is counted in the maximum number of microphones. • **Man Off**: Gates the channel off. The audio signal is attenuated by the specified **Off Attn** level. • **Override On**: Gates the channel on. The audio signal is on permanently. This channel is not counted in the maximum number of microphones. That means this microphone is always active. • **PA Adaptive**: The audio level of the output specified under **System Settings** in the **Processing** tab is used as the reference ambient audio level at which the mic is not gated on.

Option	Settings
Gate Ratio	Defines how much louder the audio level of the microphone must be compared to the ambient audio level before the channel automatically gates on if **Auto** is selected. The default value is 15 dB.
Hold Time	Defines how long the channel stays gated on after the audio level falls below the specified **Gate Ratio**. The default value is 0.3 seconds.
Off Attn	Defines the attenuation of the audio signal when the channel is gated off. The default value is 12 dB.
Decay Rate	Defines how fast the audio signal is turned off after the specified **Hold Time**. The default value is **Medium**.
Chairman Override	Activates gating priority of the selected mic over any other mic. When a mic with **Chairman Override** enabled gates on, all mics which do not have **Chairman Override** enabled will gate off.

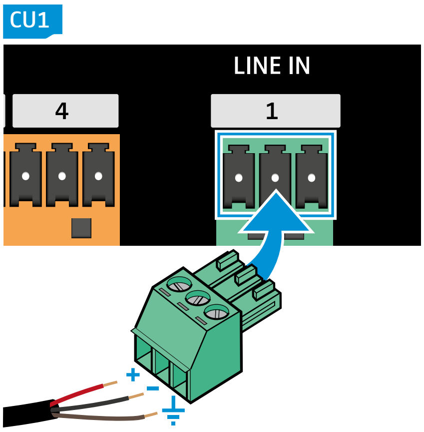
The following illustration explains the correlation between the audio signal and the gate status of the microphone.



LINE IN setup

If you are using line level devices with the TeamConnect system, e.g. DVD players, please observe the following aspects.

▶ Connect the line level device to the green LINE IN terminal with the supplied terminal connectors observing the correct pin allocation.



If you are using a true stereo device.

▶ Connect the left output channel of the device to the LINE IN 1 channel.

▶ Connect the right output channel of the device to the LINE IN 2 channel.

Connecting line level devices to the microphone inputs

You may also connect line level devices to the orange MIC/LINE IN terminal. In that case you will have to observe the following configuration:

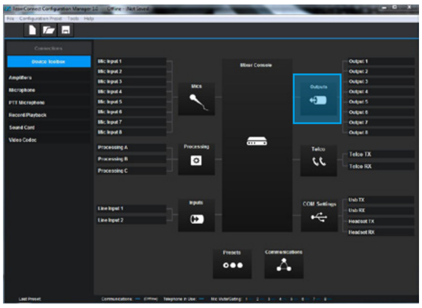
▶ Open the **Mics** configuration window.

▶ Navigate to the **Gating** tab.

▶ Set the microphone channel with the line level device to **Man On** and increase the **Max Number of Microphones** by the number of line level inputs connected to the MIC/LINE IN terminal.

LINE OUT setup

Standard meeting applications are usually mono as the audio signal is only speech. Therefore, no real stereo output is necessary. If you have two loudspeakers in the room, you can connect both to one output channel using one terminal connector for both loudspeaker cables.



To adjust the settings of the connected output devices:

- Open the **Outputs** configuration window of the Configuration Manager software.

You can adjust the following settings:

Option	Settings
Make / Model	Indicates if a device from the Device Toolbox is used (see page 15).
Output Gain	Adjusts the gain of the selected output channel.
Output Level	Displays the output level of the selected output channel.
NOM	Activates or deactivates the NOM function (Number of Open Mics) This function is activated by default. NOM maintains a constant output level by automatically adjusting gain levels based on the number of microphones gated on and routed to the selected output channel. NOM reduces the output level proportionally by 3 dB for every doubling in the number of open microphones.
Mute	Mutes the selected output channel.

Only if you have a true stereo signal (e.g. from a DVD player) it makes sense to set up a stereo output with two loudspeakers at two output channels. However, if you are using the TeamConnect iOS App, you can select only one channel for the volume control in the App. In that case we recommend routing all channels which need to be included in the volume control to a processing channel. Then select that processing channel as the volume control channel for the App.

For more information on the App refer to the TeamConnect instruction manual on the TeamConnect product page at www.sennheiser.com.

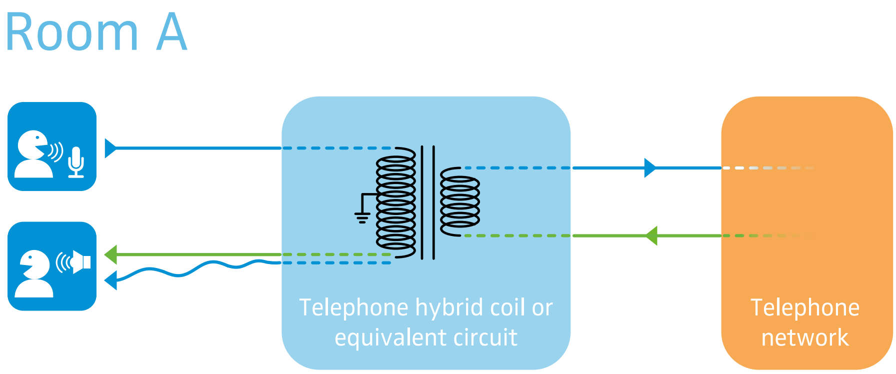
We recommend using only mono output configurations.

Telephone line setup

In addition to the acoustic echo which can occur in audio conferencing (see page), there is also telephone echo (or line echo) which can be generated in teleconferences.

In contrast to the acoustic echo, the line echo is generated at the local site and is apparent only there. It can be heard as feedback in a conference application.

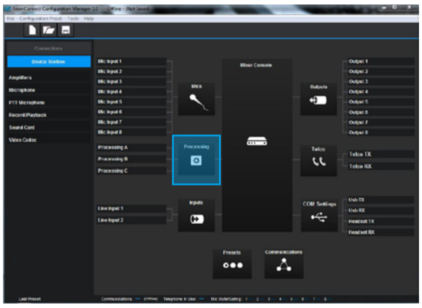
This line echo is generated in the telephone hybrid and is re-transmitted to the room. The line echo cancellation function (LEC) of the TeamConnect system will prevent this.



The line echo cancellation function is activated by default. It can be deactivated in the **COM Settings** configuration window of the Configuration Manager software by clicking on the **LEC** button. However, we recommend leaving it activated.

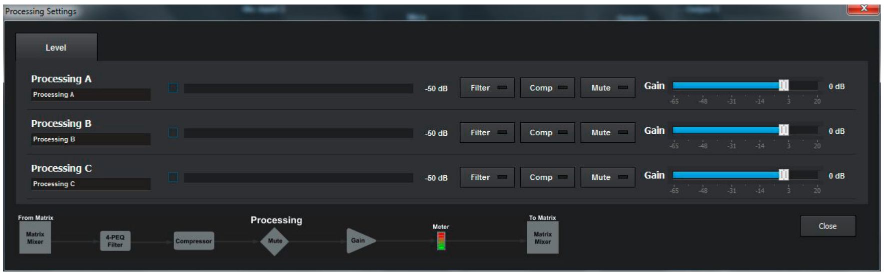
Processing channels

There are three processing channels available, which work similar to the subgroups of a common mixing desk. You can route any input channel to one or more processing channels and apply the same audio settings to a group of channels simultaneaously.



To adjust the settings of the processing channels:

▶ Open the **Processing** configuration window of the Configuration Manager software.



You can adjust the following settings:

Option	Settings
Filter	Activates and configures an audio filter for the selected processing channel (for filter settings refer to page 22).
Comp	Activates and configures a compressor for the selected processing channel.
Mute	Mutes the selected processing channel.
Gain	Adjusts the gain setting of the selected processing channel.

ℹ The channels to be included in the Processing Channels must be routed to the respective Processing Channel in the Mixer Console (see page 4).

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