AXIS Audio Manager Pro 3

User Manual

AXIS Audio Manager Pro 3

Solution overview

AXIS Audio Manager Pro 3 provides a user interface to control large and advanced audio installations. With AXIS Audio Manager Pro 3 you can manage all your Axis audio devices from a single point. You can set up zones, audio sources and scheduling.

If you are looking for the user manual for the latest version of AXIS Audio Manager Pro, see *AXIS Audio Manager Pro 4 – User Manual*.

Session

The audio is managed through sessions. A session is an audio source with a configuration and a destination (a zone). In short you play a specific audio source, with a defined configuration at a specific place.

Example

Audio source: Announcement

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Solution overview

Configurations: Audio volume, priority, etc.

Destination: Zone 1

Configure a session

To configure a session you must first:

- Connect devices to the AXIS Audio Manager Pro

- Configure zones

- Configure audio sources

To watch this video, go to the web version of this document.

www.axis.com/products/online-manual/#t10137911

Basic configuration of AXIS Audio Manager Pro.

Connect devices

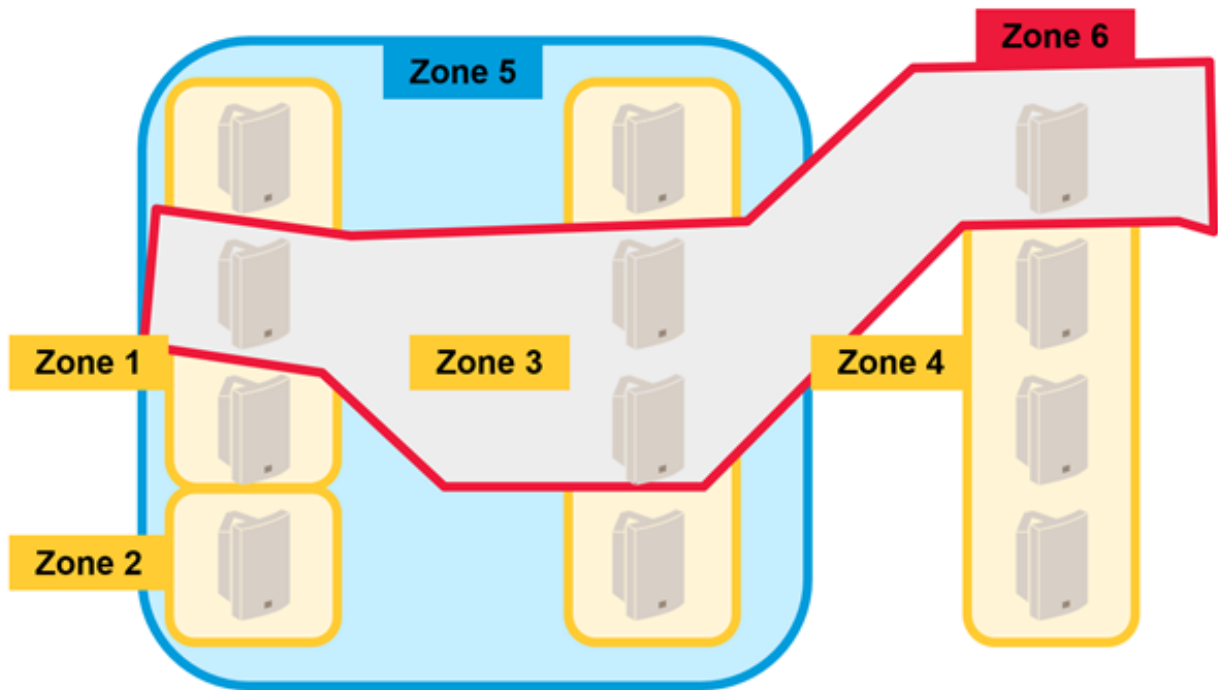First scan the network to find devices to connect, see *Connect audio devices on page 9* .

Create zones

A zone is a collection of several audio devices that is connected to AXIS Audio Manager Pro. One audio device can belong to several zones at the same time, see *Create a zone on page 10.*

Example of zone management in an audio installation.

Configure audio sources

An audio source is the type of audio input that is used with AXIS Audio Manager Pro. AXIS Audio Manager Pro has the following type of audio sources:

- *Internet radio on page 10*

- *Playlist on page 11*

- *Line-in on page 11*

- *2N SIP Mic (local network) on page 11*

- *2N SIP Mic (remote network) on page 11*

- *VoIP input (SIP) for peer-to-peer setup on page 12*

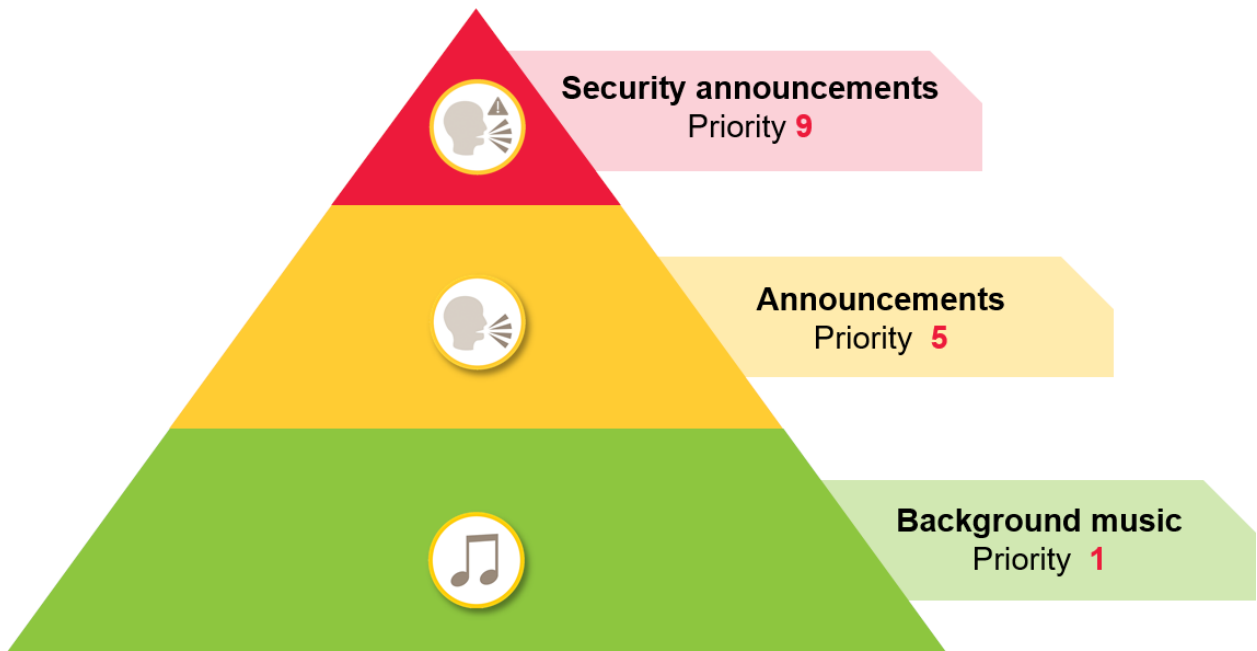- *VoIP input (SIP) for PBX setup on page 13*

- *RTP source on page 13*

Create a session

Combine an audio source and zones to a session. You can set different priorities to the sessions. If several sessions play to the same zone at the same time, it is the one with the highest priority that will be played, see *Create a session on page 13*.

Example of how to use priorities.

Scheduler

Use the **Scheduler** to create events and actions for controlling playlists or sessions. A set of actions for a playlist is called an event, and you can control multiple playlists in multiple sessions within one event, see *Create a schedule on page 14*.

Dante^TM system

You can connect AXIS Audio Manager Pro to a *Dante*^TM system, using virtual soundcards. Dante is a protocol for Audio over IP commonly used in the AV industry. With Dante you can connect and bridge Axis network audio systems to other AV systems, see *Connect to Dante on page 15*.

Considerations

Network topology

Take the following into consideration before using AXIS Audio Manager Pro:

- Make sure to have a low latency and congestion free network with enough bandwidth available.

- It is recommended to use Multicast to lower overall bandwidth for audio and to enable a scalable audio system. The network needs to be multicast enabled to transmit the audio source in multicast.

- It's important to ensure that AXIS Audio Manager Pro and all audio devices are a part of the same broadcast domain and discoverable using Multicast DNS / BonjourTM.

General

Take the following into consideration before using AXIS Audio Manager Pro:

- If your Windows® system goes to sleep, then audio transmission will stop. Go to **Start** > **Settings** > **System** > **Power & sleep** and set sleep mode to **Never**.

- Only Axis audio devices are compatible with AXIS Audio Manager Pro.

- The audio devices must be reset to factory default settings before connecting it to AXIS Audio Manager Pro. To find out how, see the user manual of the device.

- SIP-settings needs to be activated in the local interface on each audio device before configuring SIP in AXIS Audio Manager Pro.

- Currently none of the speaker models have line-in.

- HTTPS radio streams are not supported.

- We do not recommend more than one server in a local subnet. Using more than one server in a local subnet may cause interference between the servers and audio devices may switch location.

- We do not recommend to set highest priority to an audio source with line-in. Line-in is always active, even if there is no sound from the audio source. If a session with that type of audio source is given the highest priority, no other audio sources can be used.

- A 2N SIP Mic can't access its native web interface when it is connected to AXIS Audio Manager Pro.

Port list

Port	TCP	UDP	Incoming	Outgoing	Service name
23	x		x		Telnet
69	x		x		TFTP
123		x			NTP
6543	x		x		XML API
6990	x		x		Proprietary protocol speaker server
6991	x		x		Proprietary protocol speaker server debug mode

Considerations

6992	x		x		Proprietary protocol
6996		x	x	x	Broadcast listener
6998	x		x		Passive mode
6999	x		x		Active mode / search mode
7000	x		x		Expander connection
8088	x		x		HTML API

First time use

Login for the first time

1. Install the hardware according to the installation guide.

2. Turn on the PC.

3. Follow the instructions in the setup wizard to configure the PC.

4. Click on the AXIS Audio Manager Pro Client icon.

5. Leave the password field empty and click **Connect**.

Change admin password

1. Go to **Administrator** > **AXIS Audio Manager Pro Authorization** > **Administrator login authentication password**.

2. Click **Change password**.

3. Enter a password and click **Save**.

4. Click **Save configuration**.

Configuration

Connect audio devices

This section describes how to connect devices from the same broadcast domain. If you want to connect devices that are not in the same broadcast domain, then go to *Add audio device with AXIS Device Manager on page 9* .

1. Reset the Axis audio devices to factory default settings. To find out how, see the user manual of the device.

2. Go to **Administrator** > **Audio devices scanner**.

3. Select the audio devices to pair with AXIS Audio Manager Pro and click **Start using Axis devices**.

4. Follow the AXIS Audio Manager Pro wizard.

The status icon turns green when the audio device and AXIS Audio Manager Pro are successfully paired.

Add audio device with AXIS Device Manager

This topic describes how to add an audio device that is located outside the broadcast domain.

Requirements:

- All ports must be open. See *Port list on page 6* .

- The audio devices you want to add must be connected to your network.

1. If you don't have AXIS Device Manager, then you need install it:

    1.1 Download AXIS Device Manager from the *AXIS Device Manager product page*.

    1.2 To start the installation, unpack and run the downloaded file.

2. Make sure AXIS Device Manager is accessible and can reach the devices you want to configure:

    2.1 Run AXIS Device Manager.

    2.2 Set password.

    2.3 Select **Use this password for all devices without a root password**.

    2.4 Click **OK**.

3. Deploy the AXIS Audio Manger ACAP on the devices:

    3.1 Select all the speakers.

    3.2 Right-click the selection and choose **Install Camera Application...**.

    3.3 Click **Browse**.

    3.4 Select the file **AXIS_Audio_Manager_3_3–6_mipsisa32r2el.eap** and click **Open**.

    3.5 Click **Next**.

    3.6 Select **No** and click **Next**.

    3.7 Click **Finish**.

4. Configure the AXIS Audio Manger ACAP via AXIS Device Manager:

    4.1 Select all the speakers.

    4.2   Righ-click the selection and choose **Configure Devices** > **Configure…**.

    4.3   In the filter box, type "netspeaker".

    4.4   Type the IP address of the AXIS Audio Manager Pro server.

    4.5   Click **Next**.

    4.6   Click **Finish**.

5. In AXIS Audio Manager Pro, check that all devices show up as connected:

    5.1   Go to **Destinations & Zones** > **All Destinations**.

    5.2   Check that the status is **Connected** for your devices.

Create a zone

1. Go to **Destination & Zones** > **All zones**.

2. Click **Create zone**.

3. Right-click on the newly created zone.

4. Click **Rename** and enter a name for the zone.

5. Click **All Destinations** and select the audio devices you want to include in the zone.

6. Drag and drop on the zone.

A device can belong to more than one zone.

Configure audio sources (background music)

These audio sources could be relevant for background music:

- *Internet radio on page 10*

- *Playlist on page 11*

- *Line-in on page 11*

Internet radio

1. Go to **Administrator** > **Network settings** > **Proxy**.

2. Set **Access** to **Manual settings**.

3. Enter a **Host-URL** and set a **Port**.

4. Click **Save configuration**.

5. Go to **Audio Sources** > **Create Internet radio**.

6. Enter a name for the Internet radio.

7. Enter an URL from where to broadcast.

8. Set a value for **Delay** (buffering time in milliseconds).

9. Click **Save configuration**.

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Configuration

Playlist

1. Go to **Audio Sources** > **Create Playlist**.

2. Right-click on the newly created playlist.

3. Click **Rename** and enter a name for the playlist.

4. Click **Add song**-icon.

5. Browse and select what audio tracks to include.

Line-in

1. Connect an audio source to the AXIS Network Audio Bridge or the AXIS Network Audio Amplifier with a line-in cable.

2. Connect the Axis Network Audio Bridge or an AXIS Network Audio Amplifier to AXIS Audio Manager Pro by following the same procedure as for other audio devices, see *Connect audio devices on page 9* .

3. Go to **Destination & Zones** and select the audio device.

4. Scroll the property list of the audio device and select **Audio source** from **Advanced settings**. The audio device can now be used as an audio source with line-in as input.

Configure audio sources (public address)

These audio sources could be relevant for public address:

- *2N SIP Mic (local network) on page 11*

- *2N SIP Mic (remote network) on page 11*

- *Axis speakers (remote network) on page 12*

- *VoIP input (SIP) for peer-to-peer setup on page 12*

- *VoIP input (SIP) for PBX setup on page 13*

2N SIP Mic (local network)

To connect a 2N SIP Mic on the local sub network to AXIS Audio Manager Pro, follow the same procedure as for other audio devices, see *Connect audio devices on page 9* .

2N SIP Mic (remote network)

This example explains how to configure a SIP Mic on a remote network and connect it to an AXIS Audio Manager Pro server.

In the 2N SIP Mic web page:

1. Access the 2N SIP Mic.

2. Go to **System** > **AXIS Audio Manager Pro**.

3. Enter a device description.

4. In **Device mode** select **Server active**.

5. In **TCP channel** select **Server**.

6. Enter the IP address for the AXIS Audio Manager Pro server.

7. Keep **Local port** to the default choice.

8. Enter a password.

9. Click **APPLY**.

In AXIS Audio Manager Pro:

1. Go to **Administrator** > **Audio device settings** > **SIP Mic properties**.

2. Right-click in the **Clients settings** field and select **Create new connection**.

3. Double-click on the newly created connection and rename it.

4. Select the **IP address** field and enter the IP address of the 2N SIP Mic.

5. Keep the **Port** to the default choice.

6. Click **Change password** and enter the same password as in 2N SIP Mic web pages.

7. Click **Save configuration**.

Axis speakers (remote network)

To connect an Axis speaker to an AXIS Audio Manager Pro server:

1. Start AXIS Device Manager.

2. Connect AXIS Device Manager to the same network as your speakers.

3. Click **Add devices**.

4. Select the devices you want to add and click **Next**.

5. Set passwords if AXIS Device Manager ask you to do so.

6. Click **Finish**.

7. Select your speakers, right-click the selection and choose **Install Camera Application...**.

8. Browse to the application file.

   The file is called `netspeaker` and is downloaded from *axis.com*.

9. Click **Next**.

10. Click **No** for the license and then **Next**.

11. Click **Finish**.

12. On the **Ready to Install** page, select your speakers, right-click the selection and go to **Configure Devices** > **Configure...**.

13. Search for `netspeaker`.

14. Mark the checkbox for **Server ip addr** and enter the ip address of the AXIS Audio Manager Pro server.

15. Click **Next** and then **Finish**.

16. Go to the AXIS Audio Manager Pro client.

17. Enter passwork and click **Connect**.

18. Go to **Destinations & Zones** > **All Destinations** to find the speakers.

VoIP input (SIP) for peer-to-peer setup

1. Go to **Audio Sources** > **Create VoIP input (SIP)**.

2. Right-click on the newly created VoIP Input.

3. Click **Rename** and enter a name.

4. Select **Local VoIP input (SIP)** to set up a VoIP configuration using a peer-to-peer setup.

5. Enter a port number in **Listening port**.

6. Enter an IP address or a domain in **Realm (Domain)**.

VoIP input (SIP) for PBX setup

Note

> Before setting up a VoIP input you need to set up a PBX-server including a username and a password. For instruction about setting up a PBX-server refer to the product's manual.

1. Go to **Audio Sources** > **Create VoIP input (SIP)**.

2. Right-click on the newly created VoIP Input.

3. Click **Rename** and enter a name.

4. Select **VoIP (SIP) Trunk and Proxy registration** to set up a VoIP configuration using a PBX-server.

5. Enter an IP address in **Connect to**.

6. Keep **UDP/TCP** as a protocol.

7. Select **Register line**, enter a value for register line and keep the value for **Expiry**.

8. Enter the **Username** and **Password** for the number to use on the PBX-server.

9. Click **Save configuration**. When **Stack status** changes from **Gateway registering** to **Ready** the configuration has been completed.

Configure audio sources (other)

This audio source could be relevant for other type of installations than background music or public address:

- *RTP source on page 13*

RTP source

1. Go to **Audio Sources** > **Create RTP source**.

2. Right-click on the newly created RTP source.

3. Click **Rename** and enter a name for the RTP source.

4. Configure the RTP source settings.

   - If the source is a multicast stream: Select **Multicast IP address**, enter an IP address and set a **Port**.

   - If the audio source is specified stream: Select **Source IP address** and enter an IP address.

5. Click **Save configuration**.

Create a session

1. Go to **Sessions** > **Create session**.

2. Drag and drop an audio source to the session.

3. Drag and drop zones to the **DESTINATIONS** section of the session.

4. Lower the volume in the **MASTER CONTROL** section of the session.

5. To start playing music, turn on the session in the **MASTER CONTROL** section of the session.

6. Repeat the process to create sessions for other audio sources. If several sessions have been created you can prioritize the sessions. Set a value between 1 and 9 in **Priority**.

Note

• Priority 1 is the lowest priority.

• We recommend you to only add one audio source to each session. If multiple audio sources are added to the same session the audio will be mixed.

Create a schedule

Schedule recurring background music

This example explains how to schedule background music. The playlist starts daily at 09:00 and ends at 17:00.

Prerequisites

• A session including a playlist with background music.

• The playlist is set to loop (configured in the session menu).

• The session is set to ON in (Master Control in the sessions menu).

How to set up the schedule

1. Go to **Scheduler** and select **Tab view**.

2. Select **Create event** in the left main menu.

3. Set **How to schedule** to **Daily**.

4. Set **Start time** to 09:00.

5. In **List of sessions**, select the session including background music.

    The session will appear in **Tab view**.

6. Select **Play** as action for the session and select **Create new action**.

Note

When you create the first action, **Activate session** and **Deactivate session** are automatically created.

7. Select what track to start playing from.

8. Select **Stop** as an action, enter 08:00:00 in the left field and select **Create new action**.

Note

The time in the left field indicates the time between the start time of the scheduled event and the start of the action. Since the event started at 09:00 and you added eight hours, the playlist will stop at 17:00.

9. Select **Save configuration**.

Configuration

Schedule an interrupting announcement

This example explains how to schedule an announcement to interrupt the background music. The background music plays daily between 08:00 to 17:00. The announcement will play daily at 16:45. After the announcement, the background music starts to play again.

Prerequisites

- Background music, see *Schedule recurring background music on page 14*.

- A session that includes a playlist with the announcement.

- The playlist for the announcement is <u>not</u> set to loop.

- The session for the announcement has the same destination zones as the background music.

- The session for the announcement is set to **ON** under **MASTER CONTROL**.

- The background music session has lower priority than the announcements session (1=the lowest priority).

How to set up the schedule

1. Go to **Scheduler** and select **Tab view**.

2. Select the event for the background music (you find it below the calendar).

3. Under **List of sessions**, select the checkbox of the session with the announcement playlist.

   The session will appear in **Tab view**.

4. Select **Play** as action for the announcement session, enter 07:45:00 in the left field and select **Create new action**.

   Note

   > The time in the left field indicates the time between the start time of the event and the start of the action. Since the event started at 09:00 and you added seven hours and forty-five minutes, the announcement starts at 16:45.

5. Select **Save configuration**.

Configure 2N SIP MIC buttons

This example explains how to set up one of the buttons of the 2N SIP Mic as a peer-to-peer SIP input to AXIS Audio Manager Pro.

Prerequisites:

- The 2N SIP Mic needs to be connected to AXIS Audio Manager Pro, see *2N SIP Mic (local network) on page 11*, *2N SIP Mic (remote network) on page 11*.

- One or several zones need to be created, see *Create a zone on page 10*.

- One or several playlists need to be created, see *Playlist on page 11*.

1. Go to **Audio Sources** and select the SIP MIC from the list.

2. Press one of the numbered buttons on the representation of the 2N SIP Mic.

3. Enter a name for the button.

4. Select **pre-recorded message to zones**.

5. Drag-and-drop zones to **Selected zones**.

6. Drag and drop the playlist.

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Connect to Dante

Prerequisites

- Dante Virtual Soundcard software from Audinate^{TM} installed.

- Dante Controller software from Audinate^{TM} installed.

1. Start Dante Virtual Soundcard and click **Start**.

2. Restart AXIS Audio Manager Pro.

3. Log in to AXIS Audio Manager Pro.

4. In **Audio Sources**, check that Dante Virtual Soundcard receivers exist.

5. Start Dante Controller and connect receivers and transmitters.

6. In AXIS Audio Manager Pro, go to **Sessions**.

7. Create a new session, using the Dante Virtual Soundcard receivers as audio source. For information on how to create a session, see *Create a session on page 13*.

User Interface

This section gives you basic information about the user interface for AXIS Audio Manager Pro.

Sessions

Sessions are used for combining audio sources and zones to be played in audio devices that are connected to AXIS Audio Manager Pro.

Session Creation and Preparation for Playing

Create a session by:

- Clicking on **Create session** empty session.

- Pointing the cursor within the Audio Sources area, right-click and select **Create session**.
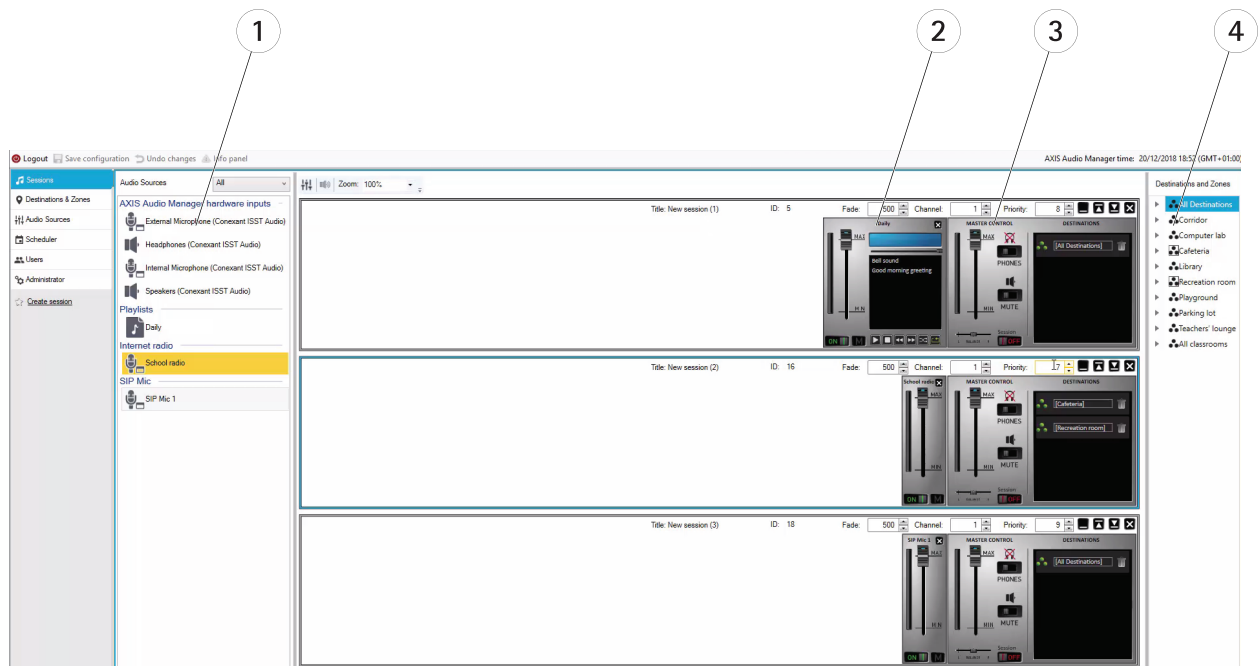
1 Available audio sources
2 Audio source in session
3 Master control of session
4 Destination and zones

Drag-and-drop audio sources to a created session. Zones can be dragged-and-dropped to the destination list.

Session Control and Launch

Click ON to launch a session.

- **Balance** – set balance for the right/left channel.

- **Listening in** – listen to remote destinations locally.

- **Volume** – set the main session volume.

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- **Sound activation/deactivation** – enable/disable sound for the active session.

- **Destinations** – list of used destinations and zones.

- **Priority** – set the session priority. The priority range is 1-10. A higher value equals a higher priority.

- **Channel** – set the channel for each session. The audio devices supports four channels from four different sources.

- **Fade** – set the session fade. The parameter can be used for a gradual sound volume change. The fade range is 500–2000 ms.
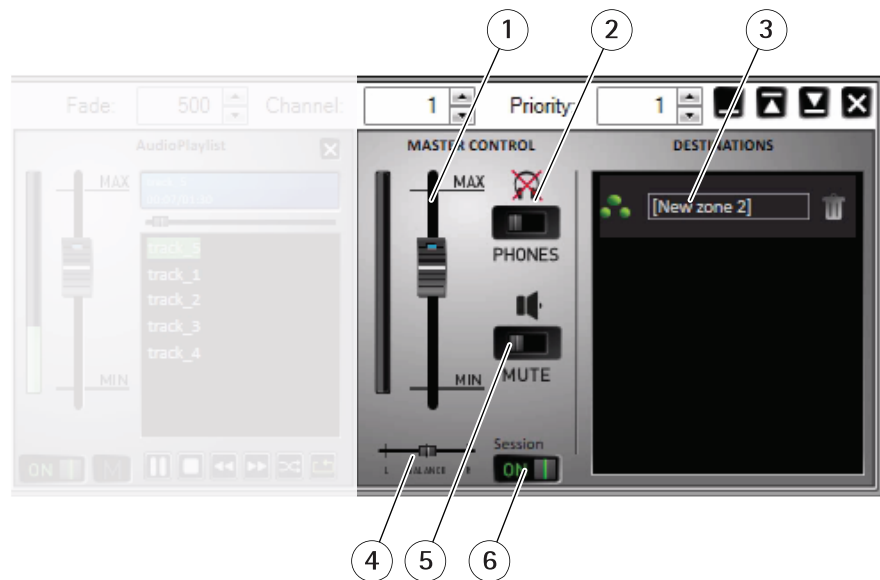
1   Master volume
2   Listening in
3   Zones in session
4   Balance
5   Mute
6   ON/OFF session

Audio Sources

Audio sources within a session are controlled with the master control and an audio source control panel. The audio source control panel looks the same for all type of audio sources except for playlist.

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1 ON/OFF
2 Mute
3 Playlist control
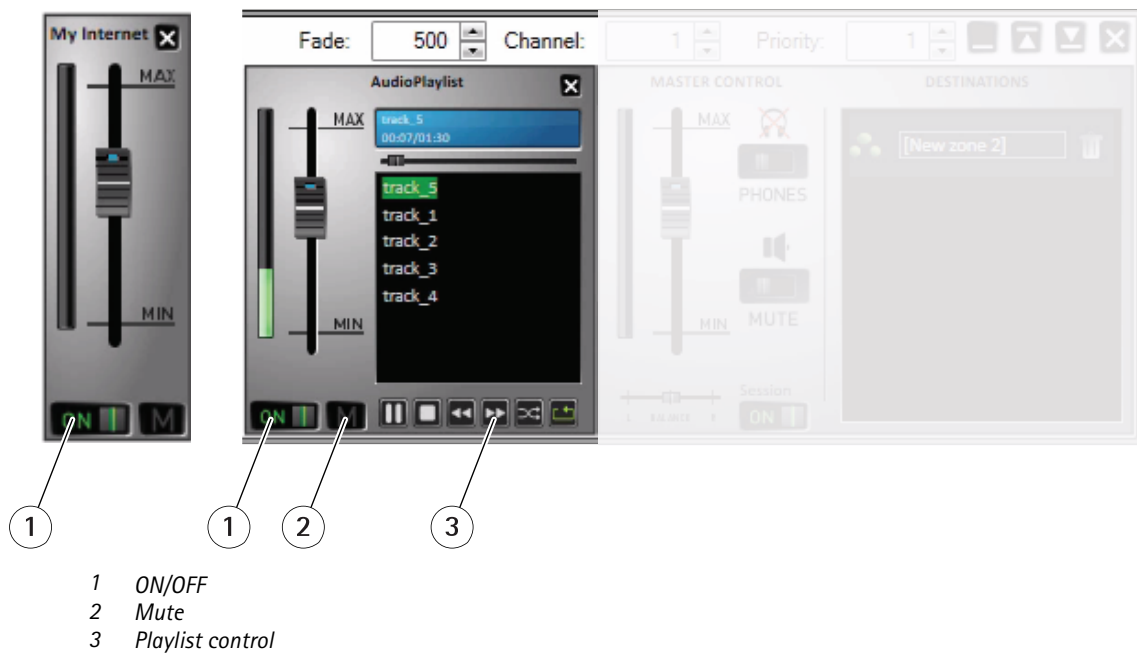
Audio source control panel – helps you control volume, balance and sound activation/deactivation.

Playlist – controls mp3 files with play/stop, pause, next, previous, shuffle and repeat.

Destinations & Zones

A zone is a collection of several audio devices which can be managed together to play music from playlists, play audio streams, announcements etc.

Create a zone:

- Click **Create zone**.

- Point the cursor on **All zones**, right-click and select **Create zone**.

Drag-and-drop audio devices from the list of audio devices to a created zone.
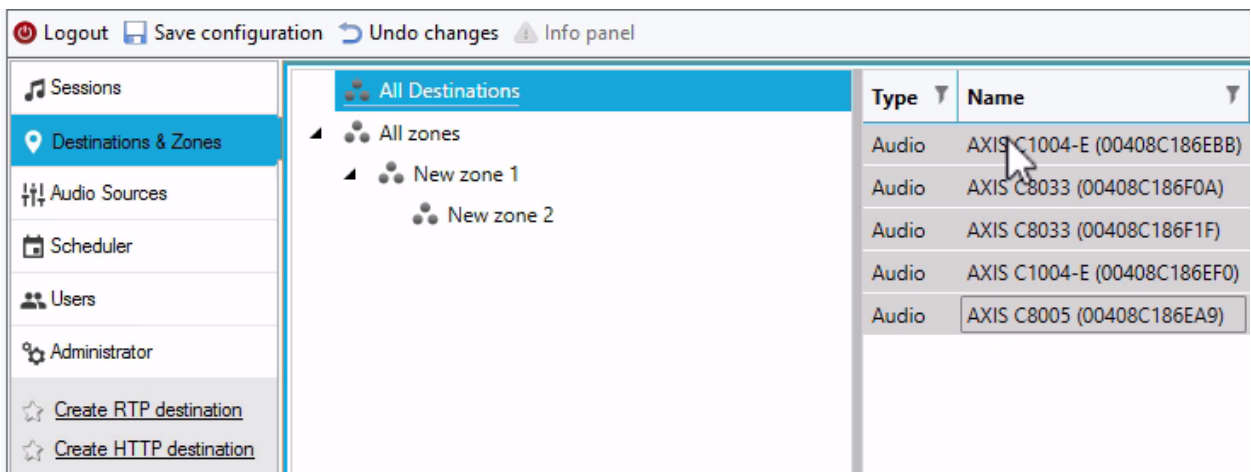
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Destination List and Filter

In the list of audio devices in the network you can select which columns shall be visible. Right-click anywhere in the list and select **Select columns**.

Column Name	Show:
Type	☑
Name	☑
Status	☑
Address	☑
Multicast	☑
Multicast address	☑
Serial Number	☐
MAC	☐
Codec	☐
Port	☐
Sound path delay...	☐
Bit rate [kbit]	☐
Currently playing f...	☐
Impedance left re...	☐
Impedance right r...	☐

OK

Cancel
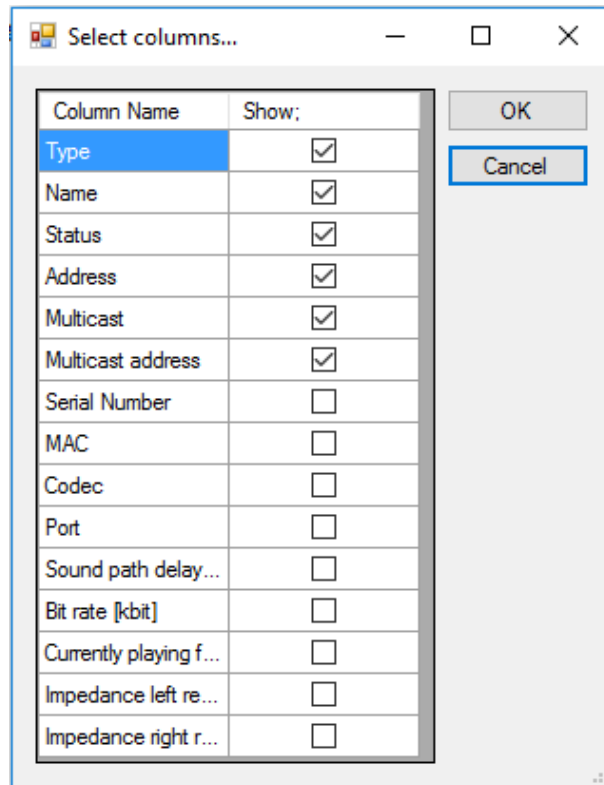
Click on one of the headings to go to the filter settings.

Properties

You can go to **All Destinations** and click an audio device to see its properties:

- **Restart** – Restart the audio device
- **Blink** – Enable LED blinking and sound notification.
- **Speaker name**
- **Assigned connection** – name and assigned connection in case the audio device is in the passive mode.
- **Serial number**
- **MAC address**

Playback Priority Settings

- General 3rd party RTP stream

Audio Parameters

- **Volume and mute** – sets the audio device volume.
- **Sound path delay [ms]** – set the expected value.
- **Channels** – set number of used channels
    - stereo – two independent audio channels are used for two independent speakers.

- mono – only one channel is used for audio transmission and both speakers produce the same sound.

- **Bit rate [kbit]** – define the count of bits to be transmitted in 1 s of the audio signal. The IP packet overhead is not included in this information.

Advanced Settings

- **Audio source** - Enable or disable audio from line-in.

- **Disable multicast** – multicast can reduce the network bandwidth and server resources. Multicast can be adjusted for each speaker individually.

- **Enable Telnet** – Telnet connection is disabled by default. Select this option to enable Telnet during this connection. Only used for advanced troubleshooting.

- **Channel** – set the audio device.

Generic RTP Output

Generic RTP output enables sending of the audio from AXIS Audio Manager Pro to all the devices that support audio via the RTP protocol with codec G711, L16. Codec is defined in this output. The audio stream will be sent to the defined IP address and port. A licence is necessary for a proper function of the RTP destinations.

Properties of RTP Destinations

This section is devoted to RTP destinations.

- **Destination name** – this parameter shows the name of the RTP destination and can be changed.

- **Audio codec** – allows you to select the required audio codec . You can choose one of the following options:

    - G711 u-Law

    - G711 A-Law

    - Liner PCM 16 bit Stereo

    - Liner PCM 16 bit Mono

- **IP Address** – set the multicast IP address.

- **Port** – set the port.

- **TTL** – set the TTL (Time to Live) for multicast packets: -1 means system default and we recommend you not to change this parameter. This parameter may cause network overload if not used in cooperation with a network specialist.

- **Licence owned** – shows if a valid licence was successfully added.

Change the settings for several audio devices simultaneously

If you select multiple audio devices you can save the settings for several audio devices. A grey strip indicates that you can change settings for more audio devices. Select a parameter, change it and click **Save**.

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Audio Sources

An audio source is the type of audio input that is used with AXIS Audio Manager Pro. Before an audio source is used it must be configured. AXIS Audio Manager Pro has the following type of audio sources:

- Hardware Inputs

- Playlist

- VoIP

- Internet radio

- RTP

- SIP Mic

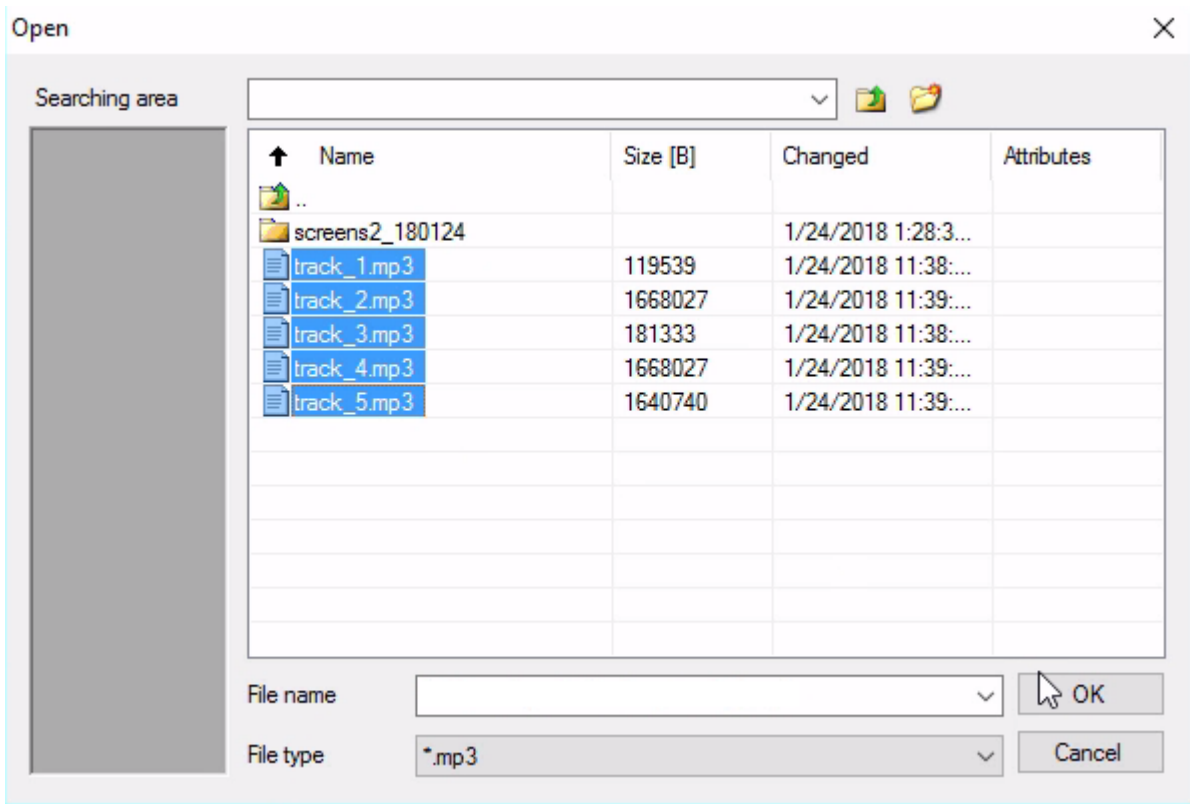- Audio bridge

Hardware Inputs

HW inputs are inputs of your server PC on which AXIS Audio Manager Pro is running.

Playlists

Click the **+** button and add audio files to create a new playlist. You can select several files at once by using Shift/Ctrl.

Internet Radio

Use an internet radio channel:

1. Go to **Audio sources** > **Internet radio**.

2. Enter a URL for and internet radio channel.

3. Set the radio delay in milliseconds.

Generic RTP Input

Generic RTP input is a common audio input. All the devices that can transmit audio using RTP protocol and codec G711, L16 can serve as a sound source for the session. The sound source is defined by the listening port, the IP address (except for unicast, i.e. IP address of the server). If AXIS Audio Manager Pro listens on the defined multicast address, it is necessary to enter this address. You can define the source IP address to make the sound source secure.
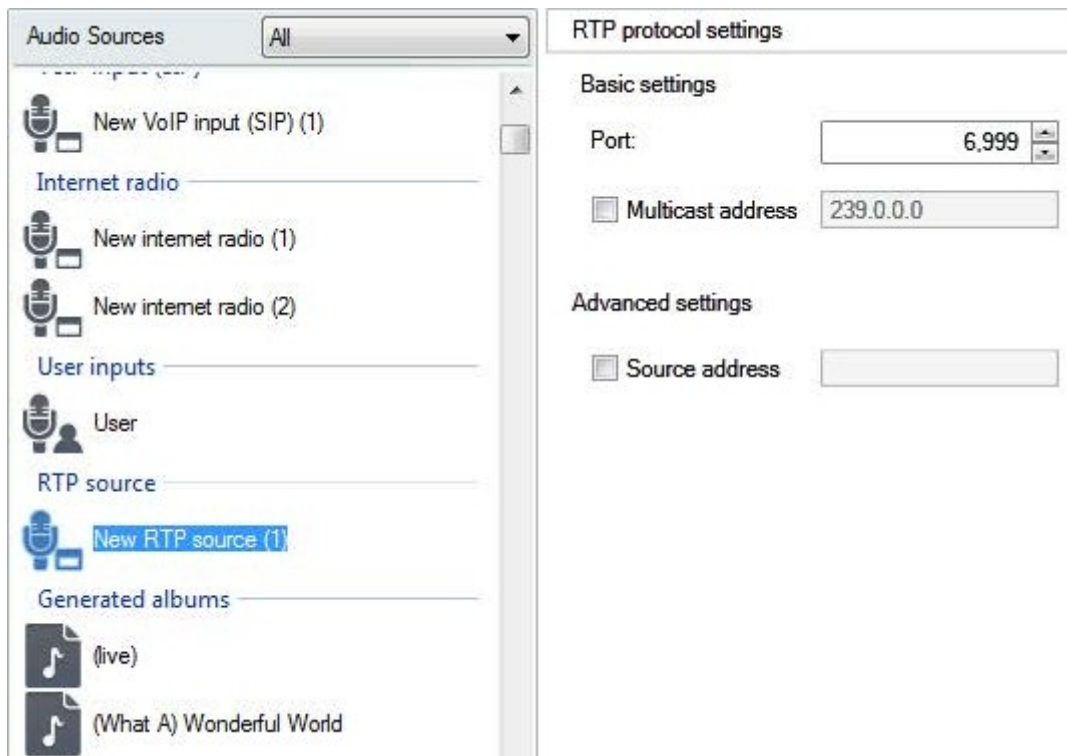
SIP Mic

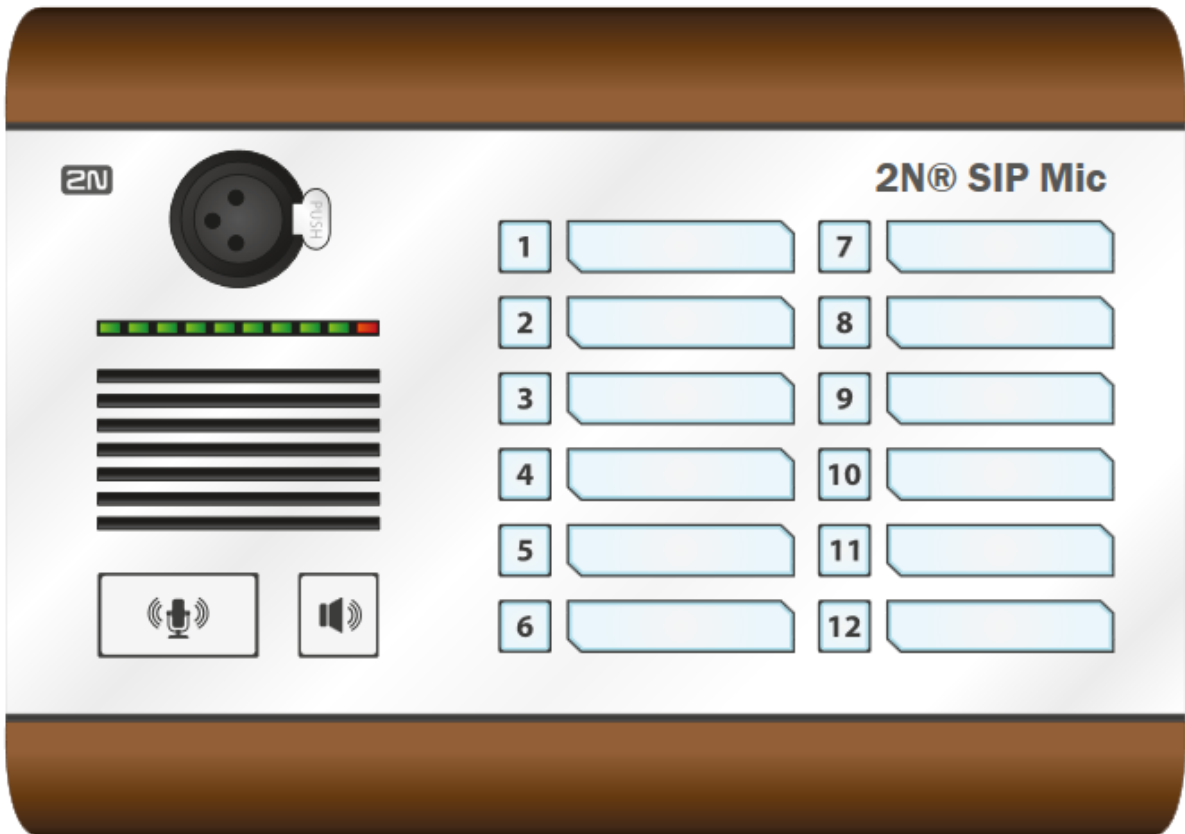Go to **Audio sources** > **SIP mic** to connect and set up the 2N SIP Mic directly from AXIS Audio Manager Pro.

SIP Mic 1 ✎

Click on the buttons to set their function

SIP Mic properties

State: Not connected

MAC address: 7C:1E:B3:02:70:AB

IP address   192.168.188.59

Audio parameters

Sound path delay [ms]:   100

Audio codec   Linear PCM 16 bit Mono (32kHz, 512kbps)

Priority

Priority: 10

Audio bridge

To connect and set up an audio bridge directly from AXIS Audio Manager Pro, go to **Audio sources** and select the audio bridge.

AXIS C8033 (ACCC8EADE594) ✎

AXIS C8033 Network Audio Bridge properties

State:	Connected
MAC address:	AC:CC:8E:AD:E5:94
IP address	192.168.188.44

Audio parameters

Sound path delay [ms]:	100
Audio source:	Microphone
Microphone power:	☐
Input gain [dB]:	9
Sampling frequency [Hz]:	44100
Channels:	Stereo
Audio codec:	MP2
Bit rate [kbit]:	192

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User Interface

VoIP Input

Go to **Audio sources** > **VoIP input (SIP)** connect an audio device to a VoIP (Voice over IP) PBX as a VoIP station via the SIP or a VoIP trunk.

Enter the PBX IP address, username, password (if requested) and the port to establish a connection to the PBX.

You can also allow direct VoIP calls to AXIS Audio Manager Pro by selecting **Local VoIP input (SIP)** and assign the source to a session. AXIS Audio Manager Pro can then receive peer-to-peer VoIP calls from VoIP phones.

Local Settings

- **Listening port** – is a local port of AXIS Audio Manager Pro which the given gateway is communicating with the counterparty.

- **Realm (Domain)** – defines the domain over which this gateway is communicating. The domain and port specified here are relevant for subsequent call routing to AXIS Audio Manager Pro. The Request–URI field including Realm (Domain) + port are checked for incoming INVITE messages. If the values match the SIP GW setting, the packets are routed to AXIS Audio Manager Pro. The INVITE messages are served to the Request–URI values included in the Aliases field.

- **Via/Contact header** – defines the contents of the **Via** and **Contact** headers. The following options are available:

    - **IP address** – fill in the unique PBX IP address.

    - **FQDN** – the header includes the PBX Host Name, which can be filled on the PBX IP interface.

    - **NAT** – fill in the fixed public IP address and NAT port to which signalling messages for the PBX should be sent by the counterparty. Packets are routed to the PBX based on the set port routing IP address on the router.

    - **STUN** – fill in the STUN server address and port to identify the current address behind the NAT router.

- **Authorisation required** – activate authorisation request for incoming calls from the counterparty. User login data are used for call authorisation. All login are always searched through.

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User Interface

Remote SIP Server Parameters

- **Connect to** – set the IP address or DNS name of the counterparty (operator or another PBX) to which you want to connect AXIS Audio Manager Pro via a trunk (where call and registration requests shall be routed). To use a port other than 5060, specify the port behind the colon (192.168.122.43:5071).

- **Protocol** – specify whether UDP and/or TCP shall be used for transmission. If you choose NAPTR (Name Authority PoinTeR), a query to the DNS is made first and, depending on the reply, the proper transmission protocol is set.

- **Register line** – enable registration with the counterparty and specify the gateway registration number (Caller ID). No call setup requests can be resent to an unregistered gateway.

Trustful IP Addresses

The parameter helps secure AXIS Audio Manager Pro against undesired call attempts via the given SIP gateway. Tick off this option to only let through requests from IP addresses included in the list. Add, remove or modify an IP address in the list using the buttons to the right of the IP address list or the context menu that you get when right-clicking in the IP address list.

Advanced VOIP (SIP) settings

Click on **Advanced** to go to the **Advanced VOIP (SIP) settings**.

SIP
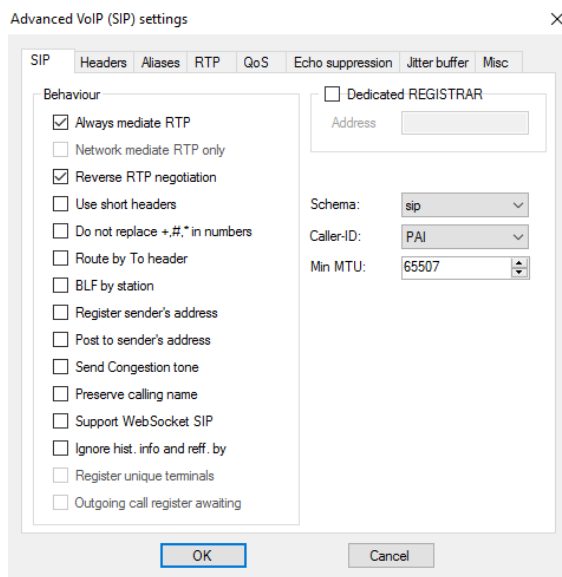
- **Always mediate RTP** – route the RTP stream via the PBX VoIP card in all cases. Otherwise, the RTP stream might be routed outside the PBX (for VoIP – VoIP connections) and the PBX processes only signalling for such a call.

- **Reverse RTP negotiation** – set the codec negotiating method. If you do not tick off this option, the PBX offers codecs in the INVITE message.

- **Use short headers** – use abbreviated header items for outgoing SIP packets. Example: From = f, To = t, Via = v. This optimisation helps minimise data to be transmitted.

- **Do not replace +,#,* in numbers** – the characters (+,#,*) will be replaced with adequate strings %xx in numbers.

- **Route by To header** – incoming call routing on the port will obey the header setting. Otherwise To (and by default!), calls are routed by the Request URI header.

- Register sender's address

- Post to sender's address

- Send Congestion tone

- Preserve calling name

- Ignore hist- info and reff. by

- **Schema** – sets the sip or tel scheme in the "To" and "From" headers of the SIP. tel is used for networks based on the numbering plan according to recommendation E.164.

- Caller-ID

- **Min. MTU** – the minimum packet length for obligatory use of TCP in the UDP & TCP mode. The recommended maximum value is 1448 bytes.

Headers

- **Complete domain** – specify the domain to be used within the **From**, **To** and headers.

- **Send information – P-Asserted-Identity** – activate the P-Asserted-Identity header for the INVITE message. This header is used for CLIR transmission to the counterparty, giving it information on the calling number even in the case of active CLIR (Calling Line Identification Restriction). By default, the header is enabled on the SIP Gateway and disabled on the SIP Proxy.

- **Complete users**

Aliases

Use this option to specify additional Realms (Domains) to be accepted on this port. Such incoming calls (their INVITEs) will be routed to this port whose Request-URIs match the given SIP GW or SIP Proxy settings and Domains or Aliases.

RTP

- **DSP** – optimise data to be transmitted. Packets are not sent while the user is not speaking. VAD stands for Voice Activity Detection.

    - VAD off

    - VAD according to G.729 Annex B

    - VAD light

- **Generate comfort noise** – enable comfort noise generating. As users of classic analogue lines are used to some background noise, this option simulates a similar call impression to them.

- **Mask lost packets** – enable optimised computing of probable contents of lost packets.

- **Send with period**

QoS

The TOS/DiffServ section helps you set outgoing packet parameters which define the packet priority for processing by network elements.

- **SIP** – hexadecimal priority value for SIP packets.

- **RTP** – hexadecimal priority value for RTP packets.

- **Default values** – restore the default values for the two parameters

Echo Suppression

Use this tab to activate various echo cancelling methods.

- Suppression off

- Profile G.168 8 ms

- Profile G.168 16 ms

- Profile G.168 32 ms

- Profile G.168 64 ms

- Profile G.168 128 ms

- Delay [ms]

- Adaptive suppression

- Non-linear processing

- Reuse of coefficients

- Automatic control

Jitter Buffer

Set the parameters in this tab to optimise packet delay fluctuation during network passage.

- Type: Fixed, Adaptive, Short-run Adaptive, Non-Managed Network

- Delay [ms]

- Depth [ms]

- Automatic adaptation

- Short adaptation parameters

- Low range [ms]

- High range [ms]

- Threshold

Miscellaneous

- **Send marks by INFO method**

    - **DTMF** – select one of the two DTMF sending modes using the SIP INFO method. The modes have different formats of the DTMF transmitting message.

- **Receive marks in call**

    - **Mode** – set the supported DTMF receiving mode for calls.

- **STUN server** – The STUN server helps NAT clients (i.e. PCs behind the firewall) set up telephone calls with the VoIP provider hosted outside the LAN.

    - **Address** – complete the STUN server address (IP or domain name) to be used if the STUN IP method is selected in the port RTP interface configuration. The default value is stunserver.org.

    - **Port** – set the port to be used for STUN. The default value is port 3478.

- KeepAlive

  - **Period** – define the KeepAlive packet sending interval. The default value is 10s.

Scheduler

Use the **Scheduler** to create events and actions for controlling playlists or sessions. A set of actions for a playlist is called an event, and you can control multiple playlists in multiple sessions within one event, see *Create a schedule on page 14*.

Create an event

Prepare a session with a playlist before creating an event.

To create an event, go to **Scheduler** and click **Create event**.

Event properties

Set the frequency of the event. The following options are available:

- **Not scheduled** – this event will never be performed.

- **Daily** – the event will start at the specified time every day (or Every xth day).

- **Weekly** – the event will start at the specified time of the set days every week or Every xth week.

- **Monthly** – select a day in the month for the action start and also in which months the action shall occur.

- **Once only**
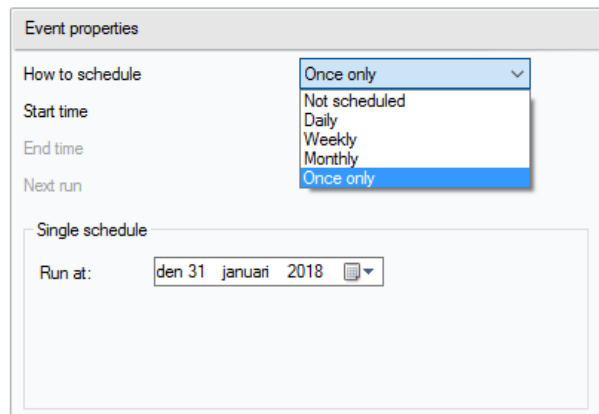
Set an action

Select a date where an event have been added. Drag and drop the required action. Click an action and edit its parameters in **Action properties**. You can set the following properties for actions:

- **Play, Stop Others** and **Previous and Activate/Deactivate session** - set the start time only.

- **Volume** – set the playlist volume – the volume level plus a smooth volume change.

- **Balance** – set the playlist balance – set start time, time for the balance change and amount of balance change.

Show events

There is an event list in the left part of the **Scheduler** tab. Display the list in any of the following views: **Month**, **Day** or **All events**.
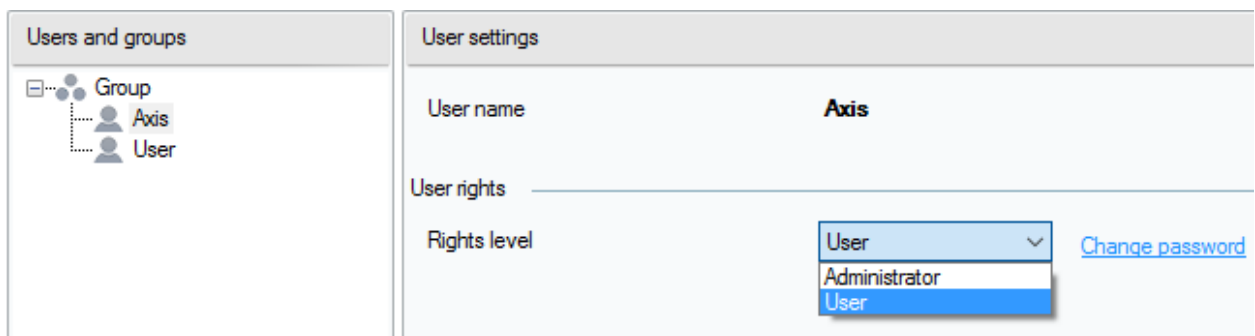
- **Month view** – displays a calendar for navigation. Click a date to run the events associated with the selected date.

- **Day view** – displays the details and what actions assigned to the selected date.

- **All events** – displays all the events available.

Users

Go to the **Users** tab to create and delete users, and change user rights.

Administration

The Administrator tab helps you set other system parameters and monitor the state of the system. The tab contains general settings and administration.

General settings – General

General allows you to set the language and external syslog server for logs.

The **SYSLOG server** is used for sending the AXIS Audio Manager Pro logs to an external Syslog server. After the IP address and destination port are set, the server messages will be sent automatically. The **TCP protocol** is used for packets and therefore it is necessary to enable TCP connection on your syslog server.

Log description:
```
1,RADIO_DISCONNECT,"New session (1)","http://api.play.cz/danceradio192.mp3.
m3u","New internet radio (1)"
Par1:  1 - id of message.  After server restart number is set to 0.
Par2:  RADIO_DISCONNECT - type of message (see below)
Par3:  "New session (1)" - source of message
Par4:  "http://api.play.cz/danceradio192.mp3.m3u" - subject of change
Par5:  "New internet radio (1)" - name of subject
```

The parameters are UTF-8 encoded and Type writer double quotes (" ") are used. If the text contains ' " ', then prefix '\' is used. It means that Speaker "Kitchen" will be encoded as "Speaker \"Kitchen\"" in the parameters.

Types of logs:
```
SPEAKER_LOGIN - NetSpeaker has been logged in; Par3=Speaker
SPEAKER_LOGOUT - NetSpeaker has been logged off; Par3=Speaker
SPEAKER_CONNECT - NetSpeaker has been connected to session;
Par3=Speaker,Par4=Session
SPEAKER_DISCONNECT - NetSpeaker has been disconnected from session;
Par3=NetSpeaker name
FILE_CONNECT - Session starts playing song; Par3=Session,Par4=name of
file with full path,Par5=Name of playlist
FILE_DISCONNECT - Session stops playing song; ,Par3=Session,Par4=name
of file with full path,Par5=Name of playlist
```

```
USER_CONNECT - User was connected to session; Par3=Session,Par4=User
USER_DISCONNECT - User was disconnected from session; Par3=Session,
Par4=User
VIRTUAL_CONNECT - Virtual input was connected to session; Par3=Session,
Par4=Virtual input
VIRTUAL_DISCONNECT - Virtual input was disconnected from session;
Par3=Session,Par4=Virtual input
RADIO_CONNECT - URL input (radio) was connected to session;
Par3=Session,Par4=URL address,Par5=Name of radio
RADIO_DISCONNECT - URL input (radio) was disconnected from session;
Par3=Session,Par4=URL address,Par5=Name of radio
HW_CONNECT - HW input was disconnected to session; Par3=Session,Par4=HW
input source's name
HW_DISCONNECT - HW input was disconnected from session; Par3=Session,
Par4=HW input source's name
VOIP_CONNECT - VoIP input was connected to session; Par3=Session,
Par4=Voip source's name
VOIP_DISCONNECT - VoIP input was disconnected from session;
Par3=Session,Par4=Voip source's name
RTP_CONNECT - RTP source was connected to session; Par3=Session,
Par4=Rtp source's name
RTP_DISCONNECT - RTP source was disconnected from session; Par3=Session,
Par4=Rtp source's name's name
SESSION_MUTED - Session was muted; Par1=Session source's name's name
SESSION_UNMUTED - Session was unmuted; Par1=Session source's name
```

General settings – Audio Devices

Audio devices allows you to select and route audio to a single a local audio device. It makes troubleshooting easier.

General settings – Licences

License administration. Licences are mandatory for advanced functions of AXIS Audio Manager Pro. The licence will be valid after PC Windows/HW upgrade.

## Interfaces

Gives an overview of available network interfaces.

Network settings – Multicast addresses

To set the multicast addresses, go to **Multicast addresses** under **Network settings**. Select multicast for the address range, multicast for the port range or single IP address transmission. The default address scope is 239.0.0.0/8. Each group of speakers with the same settings (session, delay, bitrate...) use one multicast stream.

You can change the parameter for TTL (time to live) multicast packets. Use -1 to ensure that the system default is used. Some applications may require a change of these settings, but it should be done only in cooperation with a network specialist.
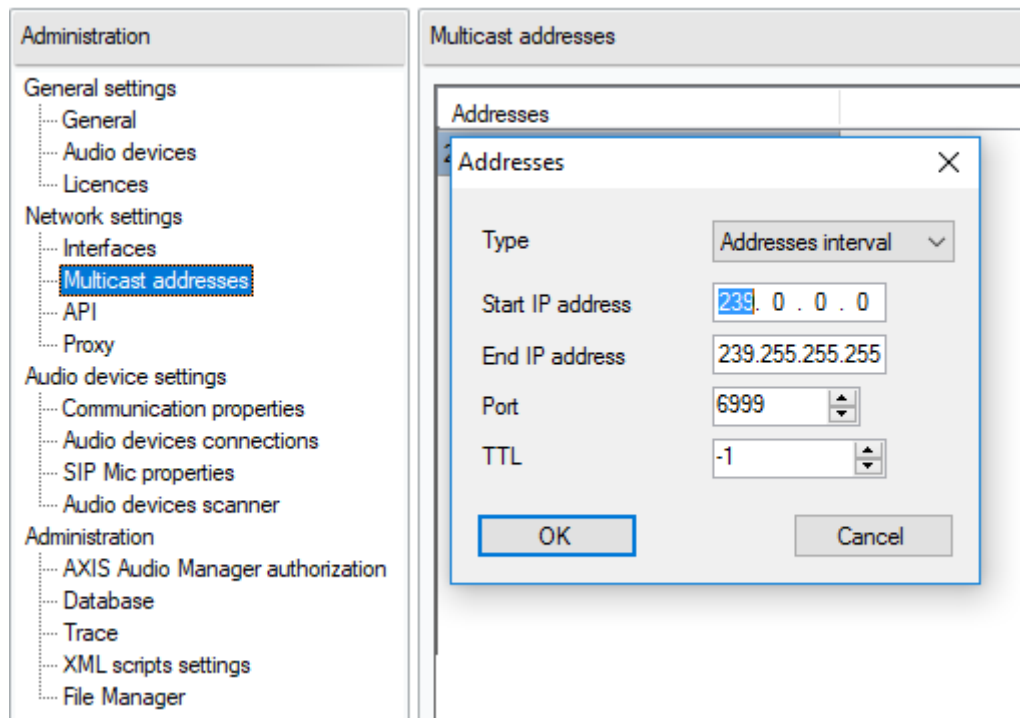
Network settings – API

Set port for AXIS Audio Manager Pro API connection and client restriction.

XML API allows an external 3rd party application (e.g. iPhone application, tablet application) to control the system. HTML API allows to send basic commands to the AXIS Audio Manager Pro and control sessions. HTML API is primary used by VoIP phones or intercoms but the command can be also sent from a web application.

Network settings – Proxy

Enable or disable the use of proxy. Enter the proxy host address.

Audio device settings – Communication properties

Set the manager domain for the connected audio devices to avoid connection to another AXIS Audio Manager Pro in the network.

Audio device settings – Audio devices connections

Set the connection parameters for audio devices in passive mode. Save the parameters to establish a connection to the audio devices.
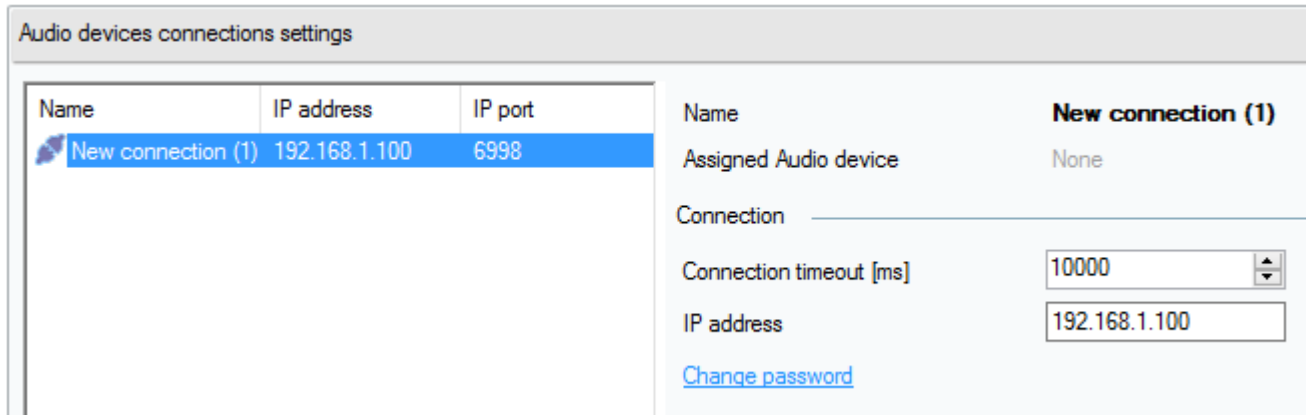
Go to **Audio device connections** under **Audio device settings** to set the parameters:

- **Connection timeout** – define the time interval between the connection making attempts.

- **IP address** – set the IP address for the audio device.

SIP Mic properties

Set the following parameters for the SIP Mic:

- **Server port**: Set the server port to use when AXIS Audio Manager Pro initiates a connection to the SIP mic.

- **Mic UDP min port**: Set minimum value for the port that will be used for announcement from the SIP mic to AXIS Audio Manager Pro.

- **SIP Mic UDP max port**: Set the maximum value for the port that will be used for announcement from the SIP mic to AXIS Audio Manager Pro.

- **Change password**: Set a new password for the communication between AXIS Audio Manager Pro and the SIP mic.

Note

The GUI does not indicate if a password is already set. If a new password is entered the old password will be overwritten.

Audio device settings – Audio devices scanner

Use the scanner to search for the audio devices and their settings in the network. All speakers available in the network are displayed to the left. Having selected a speaker, you can change its parameters.

Note

> If there is a yellow-highlighted audio device in the scanner, please check the domain (both on the manager and end device sides), or check the license for excessive count of devices for trial use.

Go to **Audio devices** and select **Audio devices scanner** to set the parameters:

- **Description** – enter optional details on the selected audio device.

- **Domain** – Audio device with the set domain is assigned to the server with the same domain.

- **Serial number**

- **MAC address**

- **Mode** – select the passive/active/search mode.

    - **Passive mode** necessary to set connection to an audio device from the server side in the **Net Audio device settings & Net Audio devices connections** section.

    - **Active mode** device gets connected to AXIS Audio Manager Pro automatically using the Server IP address parameter.

    - **Search mode** means that the audio device finds AXIS Audio Manager Pro in the network automatically, takes over the domain and connects to the server.

- **IP address** – actual IP address for an audio device.

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User Interface

- AXIS Audio Manager Pro IP address

- AXIS Audio Manager Pro TCP port

Administration – AXIS Audio Manager Pro authorization

- AXIS Audio Manager Pro authentication to audio device – set the password for manager authentication to the audio device.

- Audio device authentication to AXIS Audio Manager Pro – set the password for the audio device authentication to AXIS Audio Manager Pro.

- Administrator login authentication password – change the main administrator password of the Administrator user.

Note

The GUI does not indicate if a password previously has been set. If a new password is entered the old password will be overwritten.

Only update the password for **AXIS Audio Manager Pro authentication to audio device** and **Audio device authentication AXIS Audio Manager Pro** when trouble shooting. Make sure telnet has been enabled before.