

Exstreamer M400/MR400/MPA400, IPFormer

IP Audio Client user manual

Introduction

The aim of this document is to guide the user through the functionalities of the IP Audio Client application, which is the software running on the following Barix Products:



PRODUCT	ORDERING PART NUMBER	DESCRIPTION
Exstreamer M400	2020.9291 EU Package 2020.9292 US Package 2020.9293 UK Package 2020.9294 No PSU Package	IP Audio Decoder with stereo line level audio output on RCA connectors
Exstreamer M400 InformaCast	2020.9295 EU Package 2020.9296 US Package 2020.9297 UK Package 2020.9298 No PSU Package	IP Audio Decoder with stereo line level audio output on RCA connectors - Including Singlewire's InformaCast license
Exstreamer M400 Dante®	2023.9403P EU Package 2023.9404P US Package 2023.9405P UK Package	IP Audio Decoder with stereo line level audio output on RCA connectors - Including Dante® license



	2023.9406P NoPSU Package	
Exstreamer M400 Dante® + InformaCast	2023.9407P EU Package 2023.9408P US Package 2023.9409P UK Package 2023.9410P NoPSU Package	IP Audio Decoder with stereo line level audio output on RCA connectors - Including Dante® license + Singlewire's InformaCast license
Exstreamer MPA400	2020.9317P Package	IP Audio Decoder, PoE or externally powered with up to 40W amplified output on screw terminal block to drive loudspeakers
Exstreamer MPA400 InformaCast	2020.9325P Package	IP Audio Decoder, PoE or externally powered with up to 40W amplified output on screw terminal block to drive loudspeakers - Including Singlewire's InformaCast license
Exstreamer MPA400 Dante®	2023.9411P Package	IP Audio Decoder, PoE or externally powered with up to 40W amplified output on screw terminal block to drive loudspeakers - Including Dante® license
Exstreamer MPA400 Dante® + InformaCast	2023.9412P Package	IP Audio Decoder, PoE or externally powered with up to 40W amplified output on screw terminal block to drive loudspeakers - Including Dante® license + Singlewire's InformaCast license
Exstreamer MR400	2023_9418P Package	IP Audio Decoder, 24VDC externally powered with up to 1W amplified output, one relay
Exstreamer MR400 InformaCast	2023_9431P Package	IP Audio Decoder, 24VDC externally powered with up to 1W amplified output, one relay - Including Singlewire's InformaCast license
Exstreamer MR400 Dante®	2023_9432P Package	IP Audio Decoder, 24VDC externally powered with up to 1W amplified output, one relay - Including Dante® license
Exstreamer MR400 Dante® + InformaCast	2023_9433P Package	IP Audio Decoder, 24VDC externally powered with up to 1W amplified output, one relay - Including Dante® license + Singlewire's InformaCast license
IP Former	2020.9299P Package	IP Audio Decoder, PoE powered with up to 6W amplified output on screw terminal block to drive loudspeakers

IP Former InformaCast	2020.9301P Package	IP Audio Decoder, PoE powered with up to 6W amplified output on screw terminal block to drive loudspeakers - Including Singlewire's InformaCast license
IP Former Dante®	2023.9401P Package	IP Audio Decoder, PoE powered with up to 6W amplified output on screw terminal block to drive loudspeakers - Including Dante® license
IP Former Dante® + InformaCast	2023.9402P Package	IP Audio Decoder, PoE powered with up to 6W amplified output on screw terminal block to drive loudspeakers - Including Dante® license + Singlewire's InformaCast license
Annunicom MPI400	2021.9330P Package	IP Audio Encoder and Decoder, PoE power with up to 1W amplified output, balanced audio out, 3x audio inputs, digital inputs and relays
Annunicom MPI400 InformaCast	2022.9372P Package	IP Audio Encoder and Decoder, PoE power with up to 1W amplified output, balanced audio out, 3x audio inputs, digital inputs and relays - Including Singlewire's InformaCast license
Annunicom MPI400 Dante®	2023.9413P Package	IP Audio Encoder and Decoder, PoE power with up to 1W amplified output, balanced audio out, 3x audio inputs, digital inputs and relays - Including Dante® license
Annunicom MPI400 Dante® + InformaCast	2023.9414P Package	IP Audio Encoder and Decoder, PoE power with up to 1W amplified output, balanced audio out, 3x audio inputs, digital inputs and relays - Including Dante® license + Singlewire's InformaCast license
Annunicom AHE YA404	2025.9463P Package	IP Audio Intercom Station, 24VDC, with (16x) 25/70V intercom station connections (2/4 wires).
Annunicom AHE YA404 InformaCast	2025.9464P Package	IP Audio Intercom Station, 24VDC, with (16x) 25/70V intercom station connections (2/4 wires) - Including SingleWire's InformaCast license
Annunicom AHE YA404 Dante®	2025.9465P Package	IP Audio Intercom Station, 24VDC, with (16x) 25/70V intercom station connections (2/4 wires) - Including Dante® license
Annunicom AHE YA404 Dante® + InformaCast	2025.9466P Package	IP Audio Intercom Station, 24VDC, with (16x) 25/70V intercom station connections (2/4 wires) - Including Dante® + SingleWire's InformaCast license

Throughout this document these devices are referred to as **IPAC devices**

The document describes the elements and the functionalities of the application in detail. It doesn't cover arguments related to the physical connections of the IPAC devices, delegating this description in the Quick Install Guide of each one of them.



i Dante® is a technology developed and owned by Audinate Pty Ltd - Any reference throughout the manual have the sole purpose of educating the user of IPAC in how to integrate with the Dante®.

What is IP Audio Client?

IP Audio Client is a flexible application that controls the audio play-out of the devices on which it runs. It allows the possibility to set up multiple "sources" from where getting the audio and organize them in a multi-layer priority hierarchy, meaning that the source with an higher priority will always play over sources with lower priority.

Thus IP Audio Client serves multiple application needs, it can work as:

- a mass-emergency notification endpoint
- a background music player with embedded scheduling functions
- a paging endpoint
- a RTP decoder
- a VoIP / SIP endpoint
- a pre-recorded message player
- a Dante® receiver, transmitter and gateway device

■ Standard player

IP Audio Client can be configured to run in 3 modes:

1. Standard mode
2. AES67 mode
3. Dante Exclusive mode

One mode excludes the other, meaning that if the application is set to operate in "IP Audio Client" mode it cannot work in AES67 mode nor Dante mode. For this reason the entire user manual on AES67/Dante is detached from this manual, which describes solely the IP Audio Client application mode.

i IP Audio Client Certified for Mass Emergency Notifications

IP Audio Client is a certified **Intrado SynApps Revolution** endpoint

IP Audio Client is a certified **SingleWire InformaCast** endpoint

About this manual

This manual is intended for distributors, installers and end users of the IP Audio Client solution. It will cover first configuration steps plus details on how to use the applications after the installation.

This manual assumes that the firmware version installed on the IPAC devices is the latest one released. (check out the [release notes](#) for information on the release history and change log). Barix always recommends to run the latest firmware available and provides a dedicated download area for registered users on www.barix.com/downloads.

IP Audio Client offers the possibility to update its firmware locally from its web user interface (more details in the Update Page section).

Additional support material

The following additional documents are available for the IP Audio Client application:

- [Exstreamer M400 - Quick Install Guide](#)
- [Exstreamer MPA400 - Quick Install Guide](#)
- [Exstreamer MR 400 - Quick Install Guide](#)
- [IP Former - Quick Install Guide](#)
- [Annunicom MPI400 - Quick Install Guide](#)
- [Annunicom AHE YA404 - Quick Install Guide](#)
- [IP Audio Client - Knowledge Base articles](#)
- [Youtube Video Tutorials](#)
- Beside checking regularly the Barix website and the Barix help, the user has access to support in several ways:
 - Check with your local distributor for the technical support they offer. A list of distributors is also available on the Barix website [at this link](#)
 - Email directly a Barix support technicians at support@barix.com

Quick Start

All IPAC devices offer the possibility to connect to the network via ETH (Ethernet) wired interface (10/100Mbps). If the IPAC device supports PoE, this is the only cable required to get the device powered, otherwise you would need to power the device using an external power supply.

IP Audio Client, by default, is configured to be a DHCP client, it acquires an IP address from a DHCP server reachable in the network.



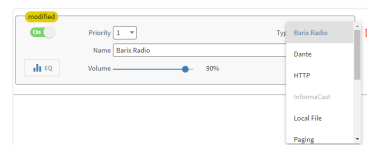
The first configuration of the device involves the following steps:



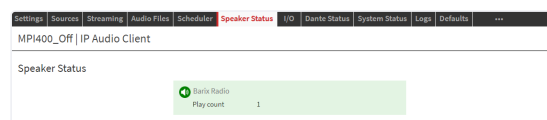
- 1 Make all the necessary connections: audio, network and then power up the unit
- 2 During boot the device will announce its IP Address (Sonic IP function), make sure to hear the audio on the output and note the IP address on paper. An IPAC device takes between 40s to 50s to finish the boot process depending on the hardware on which is running
- 3 Connect to the IP Address of the device via web browser by typing in the URL the IP address just heard or type in the browser *"http://barix-<MACLast6Digits>.local"* (e.g. *"http://barix-09c2a1.local"*)
- 4 Enter username and password (*username: admin - password: printed on the backside of the device*) - After login, please change the default password immediately under SECURITY in the Settings Page and make sure to store it in a safe place!



- 5 Locate the SOURCES tab. ENABLE any one of the sources and select the type of Source desired among the available ones. To enjoy the best music over internet simply play Barix Radio 😊 which is a pre-configured source type available in the drop down



- 6 Click SUBMIT - After a few seconds the music starts



This was a quick start chapter to get the device up and running in less than 5 mins playing background music. The rest of the manual details every function available within the application.

Connect to the device

In this chapter are detailed all the methods that can be used to retrieve an IPAC device IP Address.



Retrieve the IP address using the Sonic IP® function

All Barix devices are equipped with a function called “Sonic IP®”, meaning that when you boot the device and this is connected to a network where a DHCP server is present, the device will acquire an IP address and announce it over its audio output. This function is useful when you connect to the device for the first time and need to access its web interface for performing the first configuration.

 **IMPORTANT:** Before connecting to the amplification system, make sure the volume of your system is lowered at minimum or completely switched off to avoid potential damage to the system.

Follow the below steps to take advantage of the SonicIP function:

1. Connect the audio output to the receiving system, this varies depending on the IPAC device being used
2. Make sure the receiving system works fine so that when the IPAC device boots you are able to hear the IP Address
3. Power up the device. If the IPAC device supports PoE, this is the only cable you need
4. After 20s listen to the Sonic IP® announced and note it somewhere
5. After 35s the device is ready

Retrieve the IP address using the Discovery Tool

Other than the Sonic IP® function, there are other ways to find out a Barix device's IP address. One is by using the Discovery Tool from Barix.

The Discovery Tool is freely available for download from www.barix.com/downloads.

The Discovery Tool is written in the Java programming language, so it requires a Java Runtime Environment (JRE) installed on your computer. If you do not have a JRE installed, you can download and install it from: <https://www.java.com/en/download/> Java run time environments are available for many major operating systems. If you are running the Discovery Tool on a Linux or UNIX platform, the Discovery Tool also requires the X-window graphical user interface.

The Discovery Tool is distributed in a Java Archive (.jar) file. On most operating systems you can run the Discovery Tool by simply double-clicking on the discover.jar file. When the discovery tools open it will show the following interface:

Discovery Tool								
Set reply	Ethernet address	IP address	YR type	IRAM type	Product ID	Firmware version	DHCP name	
	00-08-e1-25-93-a3	192.168.3.37	Amuncom P31	IPAM 102	b1	v0.1.4		
	00-08-e1-23-cc-0b	192.168.3.35	Prosumer	IPAM 102	Prosumer	v0.0.0		
	00-08-e1-08-06-a9	192.168.3.31	Barix M500	IPAM 000	IP-Speaker	v0.0.0		
	00-08-e1-08-06-73	192.168.3.31	Barix S400	IPAM 000	Retal Player	v0.0.0		
	00-08-e1-08-0a-63	192.168.3.32	Barix S400	IPAM 000	Retal Player	v0.0.0		
	00-08-20-20-72-43	192.168.1.1	Barionet	generic	Barionet	v0.0.0		
	00-08-e1-07-06-0a	192.168.3.17	Barionet	generic	Barionet	v0.0.0		

Discovery Tool

Click the "Get" button to initiate a search. The program will scan the network and list all the Barix devices found with its current IP address, MAC address (labeled "Ethernet Address" in the Discovery Tool), firmware version, and other information. The Discovery Tool will find any device that is on the same network as the computer running the tool, regardless of their current IP address setting. The tool will not search through a router to another subnet. Use the address that appears in the Discovery Tool to access the IPAC device via a web browser.

Setting a static IP Address using the Discovery Tool

1. In the discovery tool window Double click the IP Address field of the device which you want to change set the Static IP Address
2. Enter the IP address you want to assign.
3. Click SET in the lower right corner - the tool will reply with a "No error" in the "Set reply" column.
4. Reboot your device and access the web interface through the newly assigned IP Address. Make sure to finalize the network settings configuration (Subnet, Gateway, DNS), otherwise the device might not work correctly.

Connect using the mDNS/DNS-SD hostname

Every IPAC device comes with Avahi zero conf service enabled by default. One of the feature of Avahi is that it provides a default hostname to the device which can be solved to an IP Address in a local area network. With this is possible to access the web interface of IPAC without knowing the IP Address of the same. Simply type the following in your browser URL: **"http://barix-<MACLast6Digits>.local"** (e.g. **"http://barix-09c2a1.local"**).

NO DHCP REACHABLE

When the device boots for the first time and there is no DHCP reachable the unit cannot be used on the network. It will assign itself the placeholder IP address

0.0.0.0 (which doesn't allow the usage of the unit on the network).

The boot process takes longer when this condition presents because the device tries multiple times to reach the DHCP server before executing the application.

The first time a device is powered it needs to find a DHCP server.

Application Settings

With the IP address or the hostname of the device is possible to connect to the web interface served by the the web server of any IPAC device.

Open a web browser and type the IP address or the hostname of the device in the URL field.

Default Password

By default new devices or devices that are accessed after a reset to defaults (soft or hard) will ask for credentials to login:

user: admin

password: *printed on the back-side of the device*

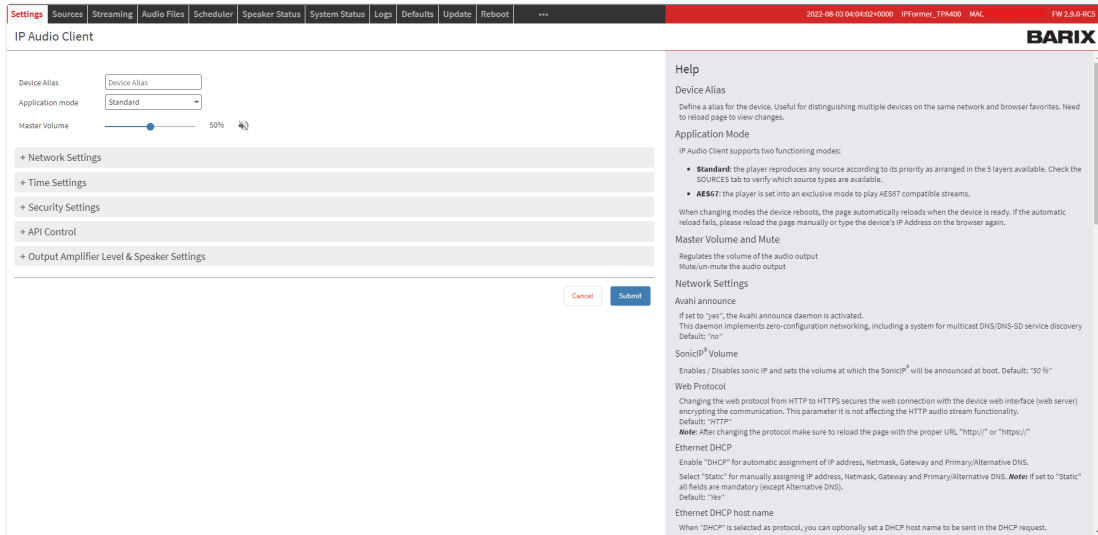
MAKE SURE TO CHANGE THE DEFAULT PASSWORD AFTER THE FIRST LOGIN AND STORE IT IN A SAFE PLACE

USE A PASSWORD WITH AT LEAST 10 CHARACTERS INCLUDING SPECIAL CHARACTERS, NUMBERS, LOWERCASE AND UPPERCASE LETTERS.

The settings page offers: network settings, time settings, security settings, API control settings and for IPAC devices with a built-in amplifier the possibility to set the power source and speaker settings.

The window is organized in:

1. The top navigation bar. In addition on the top right corner there are the MAC address and current firmware installed on the device
2. The operational area. Where the user input values and set parameters
3. The help section on the left side of the window



IPAC web UI - settings tab

Main application controls

Device Alias

Application mode

Standard

Master Volume

50%



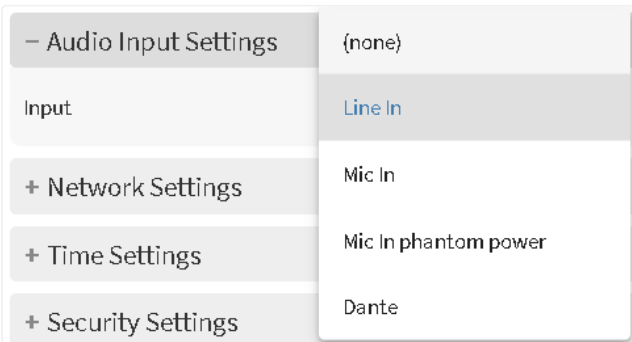
Item	Description
Name	In the top bar it is visualized the name of the Firmware and the "Device Alias"
Device alias	A custom name that can be assigned to the device. This name shows up also in the remote configuration tool <i>Default value is: " "</i>
Application mode	Select the main application mode between Standard, AES67 and Dante exclusive mode (only licensed devices) <i>Default value is: Standard</i>
Master volume	The main output volume of the device. From 0% to 100% <i>Default value is: 50%</i>



Item	Description
Mute/unmute button	Mute the main audio output. <i>Default value: false</i>

Audio Input Settings

 The Audio Input Settings varies depending on the IPAC device in use.



In this section it is possible to select which audio input to consider to feed the SIP Client of the device (for bi-directional audio calls). It is an exclusive selection, only one audio input at a time can be used.



Item	Description
Input	Allows to select the desired audio input interface among: - Line In: selects the line level input - Mic In: selects the Mic Input (this refers to the standard mic input biased at 2.1V) - Phantom Mic In: selects the Phantom Mic Input (this refers to the microphone input biased at 18V) - Dante® (only licensed devices): selects the Dante® received stream (Dante® streams are controlled and managed by the Dante® Controller) <i>Default: "None" - (M400, TPA400, MPA400, MR400)</i> <i>Default: "Line In" - (MPI400)</i>
Gain (dB)	Sets the gain of the selected input (only available for mic Phantom and mic in) <i>Default: 0dB</i>



Network Settings

– Network Settings

Avahi announce

SonicIP

SonicIP Volume

50%

Web Protocol

HTTP

Ethernet

DHCP

No

IP Address

192.168.11.223

Netmask

255.255.255.0

Gateway IP Address

192.168.11.251

DNS

Manual

Primary DNS

192.168.11.251

Alternative DNS

8.8.8.8

Syslog Address

192.168.11.135:58882

In this area, it is possible to configure the device’s network interface and few other functions.



IP Audio Client, like all other Barix products, supports the Dynamic Host Configuration Protocol (DHCP), which allows the device to acquire an IP address available from a pool of addresses hosted by a DHCP server. It is the server that offers an IP address to the device when discovered on the network. The IP address offered by a DHCP server comes with a lease time, meaning that when the leased time is over the DHCP server re-issues the IP addresses to the devices on the network, with the possibility that it might assign a different address than the one assigned previously. In addition, the DHCP server provides all associated information to the requesting device, such as subnet, gateway, and DNS addresses.



Item	Description
Avahi announce	This enables/disables the Avahi service, which is the network discovery daemon based  mDNS/DNS-SD used by one IP Audio Client device to discover in the network other IP

	Audio Client devices. Avahi assigns a default hostname to the IPAC device which can be used to access the device's web interface, the hostname is unique per each IPAC device and resides on the .local domain: <i>"barix-<MACLast6Digits>.local"</i> (e.g. <i>"barix-091f09"</i>). When enabled the device sends a query packet on the network at the Multicast IP address: 224.0.0.51 on port 5353 as defined by the mDNS protocol. The interval at which these query packets are sent doubles as time passes and no other devices are found in the network. <i>Default: Yes</i>
Sonic IP/Sonic IP volume	Enable/disable the Sonic IP function so the device announces its IP Address during boot time. Set the volume of the SonicIP with the corresponding slide. <i>Default: Yes</i>
Web protocol	Configure the protocol used to access the web interface of IP Audio Client. It can be HTTP or HTTPS. When set in HTTPS the communication between the browser and the device is encrypted. The application uses a self-signed certificate for the SSL/TLS authentication, not one from a CA (Certificate Authority) considered safe from web browsers. For this reason when using HTTPS the browser will warn that the certificate used is not a trusted one (NET::ERR_CERT_AUTHORITY_INVALID). To continue and open the web interface it is necessary to advance manually, "forcing" the browser to trust the self signed certificate. The connection and data communication will be fully encrypted. <i>Default: HTTP</i>
DHCP	Set the DHCP (Yes) or Static IP addressing mode (No). <i>Default: DHCP</i>
DHCP hostname only available in DHCP mode	Provides the DHCP with an hostname to easily identify the device in the DHCP list of leased IP addresses. <i>Default: " "</i>
Static IP mode	IP Address It is the fixed IP address of the RetailPlayer device. The IP address is the unique identifier of a device on a network, in case of static assignment make sure that no other device on the same network has the same IP address assigned to the RetailPlayer Netmask In addition to the IP address, it is mandatory to provide information about the subnet mask. The subnet mask indicates which bits in the IP address are used for the network portion and which bits for the host portion. It is common in Class C LAN to see 255.255.255.0, this

	configuration indicates that the first 3 octets of the IP address (e.g. 192.168.1.100) are dedicated to the network identification, while the last octet (e.g. 192.168.1.100) is dedicated to the host identification on the network Gateway IP address The Gateway IP is the address of the device that bridges your Local Area Network with the Internet. Typically is a router where devices on a LAN are connected. For using the RetailPlayer in all its cloud-based features, but also to receive a stream from an internet radio in stand-alone mode, for example, it is essential to provide this address. If you are not sure about the Gateway address in your network contact your network administrator
DNS auto	Automatic Domain Name System resolution. If set in Auto the DHCP server will provide the address of a DNS server to contact for the IP name resolution process.
DNS manual	If DNS is set to Manual the device expects the Primary and Alternative DNS server fields to be filled with the related information. When Ethernet IP method is set to Static, DNS will always be set Manual and it is mandatory to provide the address information
Primary DNS and Alternative DNS	This address is for the Domain Name Server. It tells the RetailPlayer which DNS server to contact to resolve domain names into IP addresses. I.e the address of an internet radio like http://barix.streamguys.net/Barix_hi has to be translated into an IP address, this is the job of the DNS server. The alternative DNS is a backup resolution in case the primary DNS server is down. The alternative DNS is not a mandatory field. If you are not sure about the DNS Server address to be used, contact your network administrator.
Syslog Address	Enables the remote syslog to send system data to a remote server over UDP. Provide a valid <IP_ADDRESS:PORT> to start logging data to the remote server. After enabling remote syslog, the internal logging function is disabled. <i>Default: Disabled (empty)</i>

Time Settings



– Time Settings

Timezone

UTC

NTP Server 1

1.barix.pool.ntp.org

NTP Server 2

2.barix.pool.ntp.org

NTP Server 3

3.barix.pool.ntp.org

Manual

 Set Date and Time

IPAC web UI - Time Settings

It's important that the IP Audio Client device operates running the correct date and time, especially if a scheduler is configured and the device is not installed in a UTC timezone. IP Audio Client uses an NTP daemon to set its system time. The NTP protocol is used on the standard port 123. The time set by NTP is in UTC, to configure the local time it is necessary to select the correct timezone from the “*Timezone*” drop down.

On the web interface, the time settings allow the user to change the default NTP pools used by the device. This is because in some situations, customers might be using their internal NTP servers, or they simply prefer to use others. The fields accept either a domain name (which requires a valid connection to the DNS to be able to associate an IP Address to the DN) or an IP Address. It is possible to check the system time in the Status page of the device.

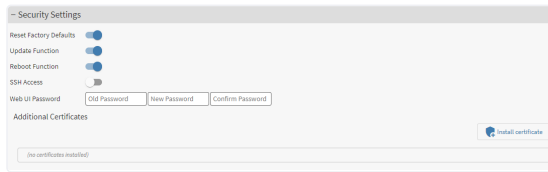
NTP sends packets once every 4mins to the same NTP server to check for clock updates. By leaving the configuration fields empty it is possible to disable the NTP function.

In case an NTP server is not reachable (e.g. device must work offline) IPAC also offers the possibility to set its own time manually. Click on “*Set Date and Time*” button to enter the information.

 When setting the Date and Time manually the configuration is lost after a reboot. For this reason, where possible, always try to adopt NTP to set the time automatically at every boot.

Security Settings

This area is dedicated to secure the login to the device web interface device and to enable or disable the possibility to run critical operations: reboot, reset to factory defaults and update



IPAC - Security Settings



Item	Description
Reset factory defaults	Enable or disable the possibility to reset the device to factory default from the web UI <i>Default: Enabled</i>
Update function	Enable or disable the possibility to update the device from the web UI <i>Default: Enabled</i>
Reboot function	Enable or disable the reboot function of the device from the web UI <i>Default: Enabled</i>
SSH access	Enable or disable SSH service. Only enable SSH Access upon request of Barix support, otherwise leave the option disabled. <i>Default: Disabled</i>
Web UI password	Set the password to access the web interface of the device. If a password is already set another field is displayed: OLD PASSWORD, which is required to modify the current password <i>default username: admin (it cannot be changed)</i> <i>default password: printed on the backside of the device</i>
Additional certificates	Allows the possibility to upload custom .crt certificates to the device. Useful when the device is used in private enterprise networks where a private CA is in place to validate secure connections <i>Default: None</i>

API Control

IPAC devices offers a REST API that can be used to set and retrieve configuration parameters from the device. This enables (i.e.) users the possibility to build personalized web applications, like CMS (Content Management System) to interact remotely with multiple IPAC devices or to integrate IPAC in thrid party solutions able to send API requests over HTTP(s).



Please find at this link the API documentation to start programming your custom web app: [IP Audio Client API](#)

– API Control

Enable

Authentication

API Key

Current API Key

.....

Generate new

Last time generated: Aug 2, 2022, 2:11:17 PM

IPAC - API Control Settings

The settings available to configure the access to the API are:

Item	↕	Description	↕
Enable		Enable or disable the API endpoint <i>Default: Enabled</i>	
Authentication		Choose the desired authentication method to be adopted by the API between <i>API Key</i> or <i>None</i> . Refer to the IP Audio Client API documentation to know how to use the API Key <i>Default: API Key</i>	
Current API key		Currently configured API Key. It is possible to copy the key in the clipboard, modify it by clicking on its field or re-generate the key with the corresponding “Generate new” button. The timestamp of the last generated key is available. When a new API Key is generated the previous one becomes invalid. <i>Default: randomly generated API Key</i>	

SUBMIT

By clicking on the button SUBMIT all the changes made in the SETTINGS page are applied in the configuration.

By clicking on CANCEL all the changes are not applied and the fields return to their previous inserted value.

Sources

A source is an originating source of audio, the application allows to arrange sources in 5 priority layers where the layer with the lowest index assigned has more importance compared



to the layer with the highest index. Meaning that Layer 1 is the most important and will play on top of any other layer, layer 5 is the least important and will play only in absence of incoming audio in any one of the other layers.

MPA400 | IP Audio Client

Sources

Input Sources Priority Table

[Import](#) [Export](#)

[+ New source](#)

On

Priority 1

Type Barix Radio

Name BR

EQ

Volume 90%

On

Priority 2

Type Dante

Name Dante

EQ

Volume 90%

On

Priority 3

Type InformaCast

Name InformaCast

InformaCast URL Server URL

Volume 90%

On

Priority 4

Type Local File

Name LF

Trigger UDP Port 8000

Loop ☐

EQ

Volume 90%

On

Priority 5

Type Scheduler

Name Scheduler

Each layer represents a priority level and a source type can be assigned to it. Use the “New Source” button to add another source in the configuration, the maximum number of sources supported is 5.

Use the *BIN* icon to remove that source from the configuration.

On the top side of the page there are *IMPORT* and *EXPORT* functions used to import / export the current source configuration in JSON format. This can be used to easily replicate the same configuration on multiple IPAC devices or to prepare the configuration to be sent in the API call used to [configure all sources](#).

i It is possible to import .json files exported by other IPAC devices running the **same** firmware version



The common functions available in each layer/source are:



ON/OFF	Allows to switch the corresponding layer on or off
Volume	Set the volume for the corresponding layer relatively to the master volume configured in the settings page. The change is applied in real time
Name	Each layer can be assigned a name that is the name of the layer

Below are all the source types available for each layer/source:

- InformaCast (only licensed devices)
- SynApps
- Dante® (only licensed devices)
- SIP
- Paging (IC Paging / Simple Paging)
- RTP
- UDP
- HTTP(s)
- Local Files
- USB
- Scheduler
- Barix Radio

** Default source configuration**

1 or 2 sources are ENABLED by default, depending on the IPAC device purchased: SynApps and InformaCast. Where InformaCast is only enabled and available on devices running the Singlewire's license.

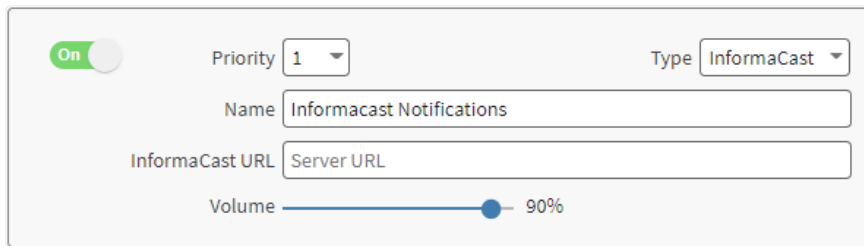
This allows the device to automatically register with the corresponding mass notification servers as soon as they are plugged. Read the following chapters for more information.

InformaCast

SingleWire's InformaCast is a mass notification platform that streams audio to certified endpoints registered with the InformaCast server. This is how the configuration of an



InformaCast source looks like:

A configuration interface for an InformaCast source. It includes a green 'On' toggle switch, a 'Priority' dropdown menu set to '1', and a 'Type' dropdown menu set to 'InformaCast'. Below these are two text input fields: 'Name' containing 'Informacast Notifications' and 'InformaCast URL' containing 'Server URL'. At the bottom is a 'Volume' slider set to 90%.

InformaCast source

i InformaCast License

Singlewire's InformaCast requires a license in order to run on IP Audio Client, just like on any other Singlewire certified endpoint. Make sure to order the right PN when purchasing the IPAC device. When your device is running the correct InformaCast license its **Feature** will appear as "InformaCast" in the System Status → Licenses as per below screen

Features	InformaCast
Signature	Valid

IPAC implements the InformaCast API that governs how the IPAC device:

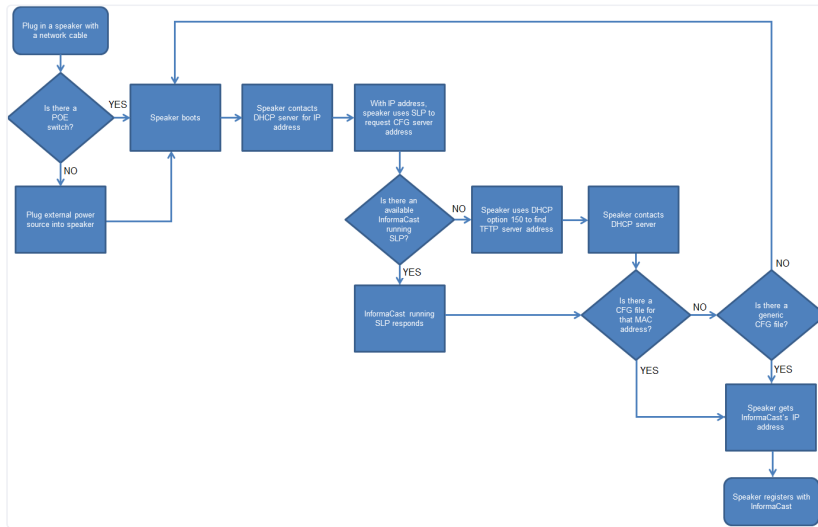
1. Registers with the InformaCast server
2. Receives commands and notifications

For this reason the configuration is easy and straightforward.

InformaCast registration

The registration process with an InformaCast server doesn't require any configuration if the server is reachable from the network where the device sits. IP Audio Client devices implement registration over SLP (Service Location Protocol), if that fails TFTP (Trivial File Transfer Protocol) is used (requires DHCP option 150 to be configured in the DHCP). If none of the mentioned protocols are available in the network or failing it is possible to specify manually the URL in the "InformaCast URL" field in the Source configuration.

The automatic registration process is well described in the below logic diagram (picture taken from [InformaCast IP Speaker registration notes](#)):



InformaCast Registration Process

For more information and troubleshooting steps about the InformaCast registration visit this link provided by SingleWire: [IP-Speaker-Registration-and-Troubleshooting-Guide](#)

In case your InformaCast server and the IPAC device is behind a firewall, check this link: [Informacast Inbound / Outbound connections](#)

i InformaCast Manual registration

When entering the URL manually in the source configuration of IP Audio Client the URL should be: ***http://<ICast_server_IP>:8081/InformaCast/admin***

After the registration the device heartbeats the server once every 5 minutes to keep the server updated about its existence. This interval value cannot be configured by InformaCast (despite there is a function on InformaCast that allows it), it is fixed in the configuration file of the device. If the device doesn't send the heartbeat to the server the server will consider the device "expired" and it is not possible to send notifications to it.

InformaCast IDLE Audio

IDLE Audio is a function within the InformaCast source available only when the InformaCast URL is explicitly specified (the IDLE Audio configuration fields appear when the user starts typing in the InformaCast URL field).

This function allows to specify an RTP stream to play as background music when no notifications are sent. It can be seen as a "source-in-a-source". Of course it plays in the same priority level of the InformaCast source (but not over InformaCast notifications), thus if Background music is playing on Priority 1, lower level priority sources will never play.

The url RTP scheme should be as follow: ***rtp://<IP_Address>:[Port]***

It can be a unicast or multicast IP Address:

- Unicast example: `rtp://0.0.0.0:12345` (the `rtp://@:12345` schema is not supported for InformaCast Idle audio)
- Multicast example: `rtp://239.0.0.1:12345`

IDLE Audio format specifies the decoding format to be used for IDLE audio. Available formats are

- PCM @ 44.1kHz 16bits mono
- G711 uLaw 8kHz mono

How to setup InformaCast on IP Audio Client and receive notifications



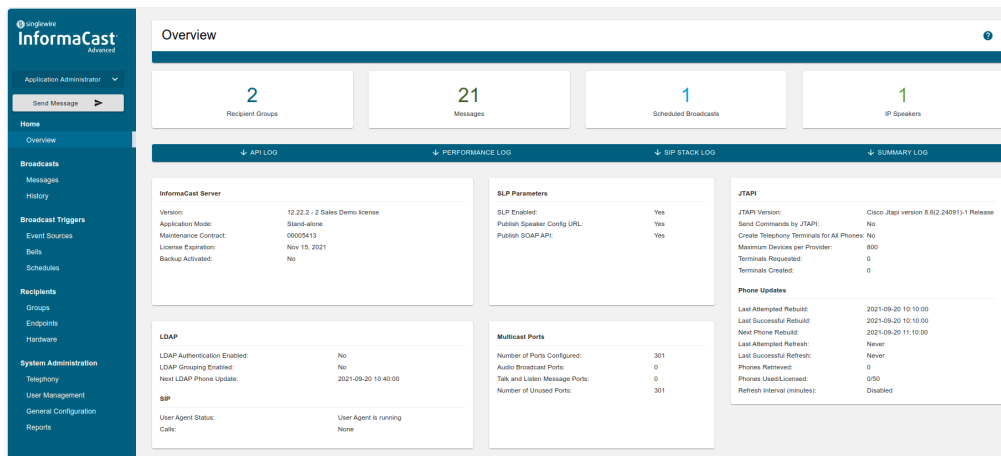
- 1 Make sure that your InformaCast Server is running and that a network communication with the IPAC endpoint can be established. By default IPAC licensed devices will boot with InformaCast already enabled in the Priority source 1. The highest priority source. Boot your IPAC device.

IPAC devices will try to automatically register with the InformaCast server using SLP (Service Location Protocol), if this fails it will try with TFTP (DHCP Option 150 configured and available in the DHCP of the network), if this fails the device will not be able to register automatically with the server, thus it will be necessary to enter the URL of the InformaCast server manually in the source configuration. The speaker status will confirm the success of the registration when the device is registered with the ICast server providing the time of the last successful registration process.

Speaker Status - InformaCast registered

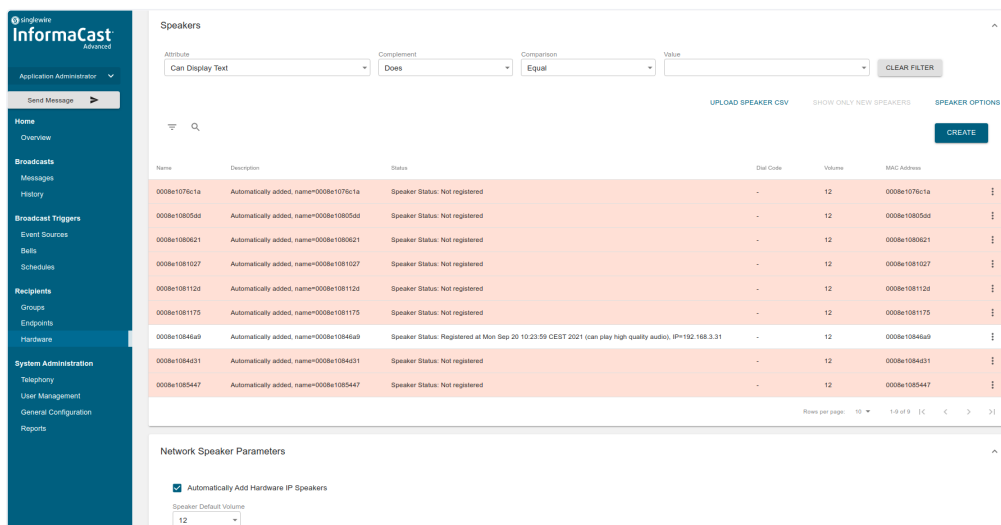
- 2 Navigate to the InformaCast dashboard typing the URL of your server in the browser (typically the administration dashboard can be reached at: **`https://<ICast_IP>:8444/InformaCast/admin`**). In the dashboard it is possible to check the number of speakers currently registered with the server and ready to receive notifications.





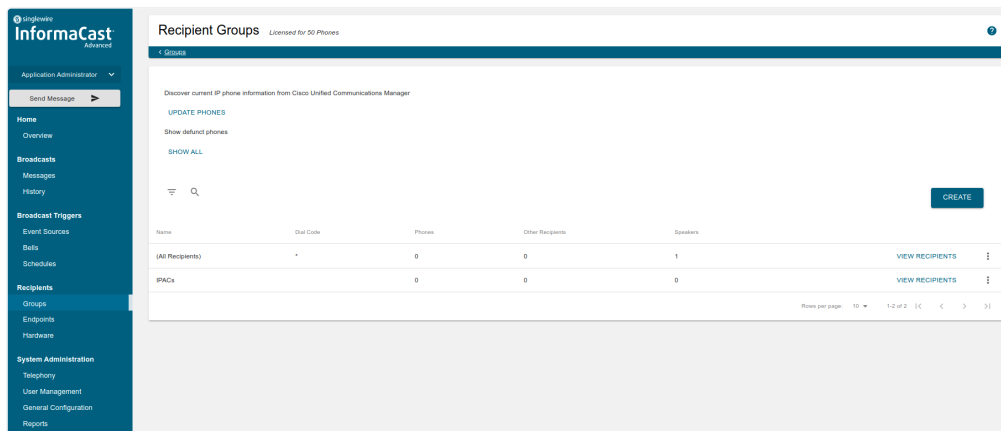
InformaCast Server Dashboard

- Clicking on the IP Speakers icon in the top right corner of the dashboard opens the list of IP Speakers with all the details. Underneath the IP speaker list, there is an important flag "**Automatically Add Hardware IP Speakers**", if this option is not flagged IP Speakers will not be listed automatically, but have to be entered manually following the IP Speaker creation procedure (refer to InformaCast user guide)



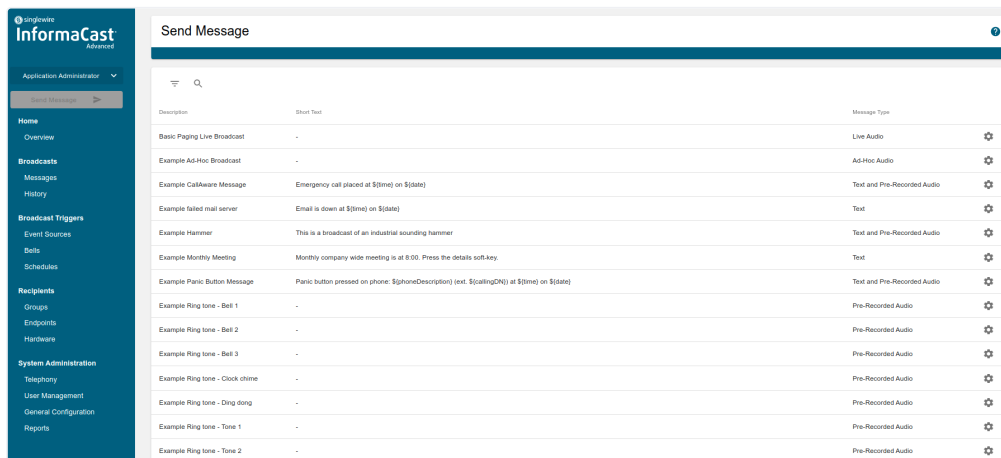
InformaCast IP Speaker List

- Create a recipients group in **Groups** → **Recipient groups** accessed from the left side bar menu. Follow the on screen instructions to add IPAC devices to the group and confirm the group creation. This step is not mandatory as you can also send notifications to a pre-defined group called "all recipients" but it's useful if you want to test only one IP Speaker or a group of IP Speakers.



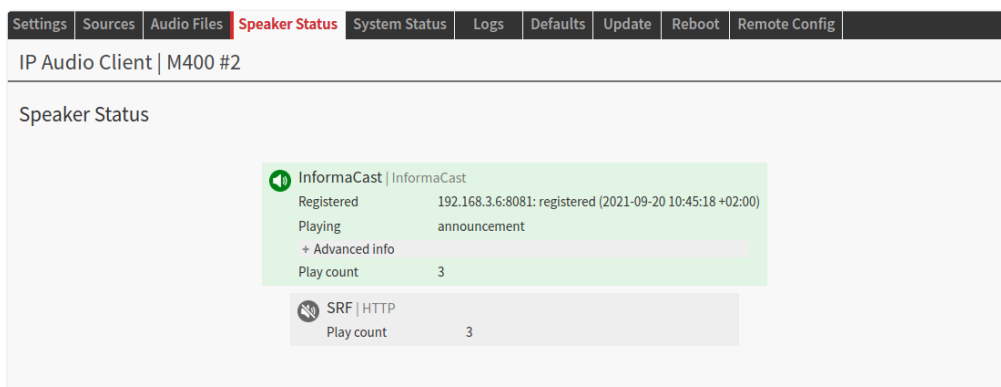
InformaCast - Create Recipient Group

- 5 Test an audio notification: click on "**Send Message**" in the left side bar, select a "Pre-Recorded" audio notification type, select the recipient group that has to receive the notification and click on **SEND**



InformaCast - Send Notification

- 6 While the notification is received on IPAC device it is possible to monitor the status in real-time. Check the "Advanced" tab to view more details such as the Multicast address used by InformaCast to broadcast the notification (this address is configurable in the InformaCast server)

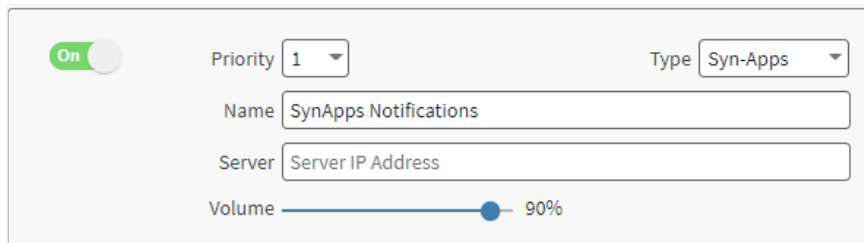


Speaker Status while Informacast plays



SynApps

[SynApps Revolution by Intrado](#) is a mass notification platform that streams audio to certified endpoints registered with the SynApps Revolution server. This is how the configuration of a SynApps source looks like on the IPAC device:

The image shows a configuration window for a SynApps source. It features a green 'On' toggle switch, a 'Priority' dropdown menu set to '1', and a 'Type' dropdown menu set to 'Syn-Apps'. Below these are two text input fields: 'Name' containing 'SynApps Notifications' and 'Server' containing 'Server IP Address'. At the bottom, there is a 'Volume' slider set to 90%.

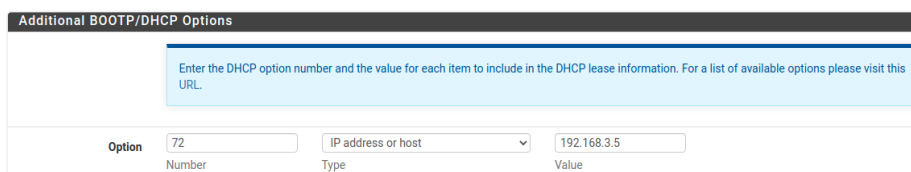
SynApps source

IPAC implements the SynApps API to: register with the SA server, receive commands and notifications. For this reason the configuration is easy and straightforward. SynApps is switched ON by default at priority 1 on any new IPAC device. If the device is licensed for InformaCast as well, it is ON by default but in the priority layer 2.

SynApps registration

The registration process with a SA server from an IPAC can be automatic or manual.

Automatic registration requires the DHCP server in the network to have enabled and configured DHCP Opt 72 ([more information on DHCP options here](#)). Below is an example of a router where it is possible to specify the DHCP option and associate the IP Address of the SA server to it.

The image shows the 'Additional BOOTP/DHCP Options' configuration page on a pfSense router. A light blue box contains instructions: 'Enter the DHCP option number and the value for each item to include in the DHCP lease information. For a list of available options please visit this URL.' Below this, there is a table with three columns: 'Option Number', 'Type', and 'Value'. The first row shows 'Option Number' as 72, 'Type' as 'IP address or host', and 'Value' as 192.168.3.5.

DHCP Opt. 72 on a pfSense router

Once this is configured the DHCP will release this information to the IPAC device when it will try to find in the network the service provided by the SA server during boot time. If your device was already powered when the DHCP option was configured, reboot it to make sure it will find the information it needs to connect automatically with the SA server.

On IPAC devices the automatic registration is the default option. Leave the "Server" IP Address field empty in the source configuration to benefit from the automatic registration process.

Manual registration is an option to use when for example DHCP Opt 72 is not available in the DHCP server present in the network. Manual registration means that the IP Address of the SA Server IP address has to be manually entered in the SynApps source configuration in the IPAC device. See below example:

On

Priority1

TypeSyn-Apps

NameSynApps Notifications

Server192.168.3.5

Volume90%

SynApps manual registration

After the first registration the device heartbeats the server once every minute to keep the server updated about its existence. This interval value cannot be configured on the device nor on SA Server, it is fixed in the configuration file of the device. If the device doesn't send the heartbeat to the server the server will consider the device "expired" and it is not possible to send notifications to it.

How to setup SynApps on IP Audio Client and receive notifications



- 1
- Make sure that your SA Server is running and that a network communication with the IPAC endpoint can be established. By default IPAC devices will boot with SynApps already enabled in the Priority source 1 (Priority source 2 if InformaCast is also running). If, after boot, the connection with the server can be established and the DHCP Server in the network provides DHCP Opt 72 the device will register automatically, if this fails or if DHCP Opt 72 is not configured or available the device will not be able to register automatically with the server, thus it will be necessary to enter the URL manually in the source configuration. Here is an example of what is shown in the speaker status after the device successfully registered with a SA server.

SettingsSourcesAudio FilesSpeaker StatusSystem StatusLogsDefaultsUpdateRebootRemote Config

IP Audio Client | M400 #2

Speaker Status

SynApps | Syn-Apps

RegisteredYes

Playingnothing

Play count1

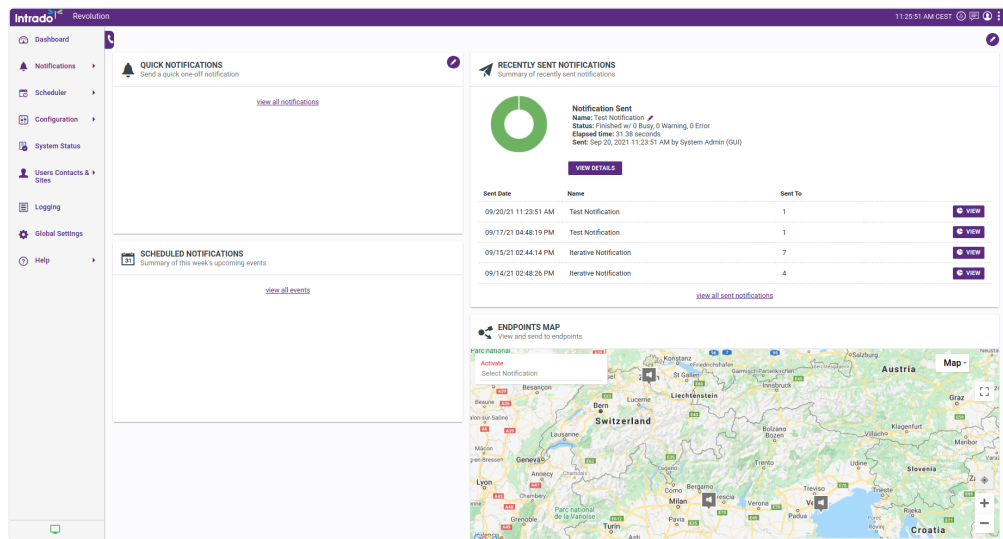
SRF | HTTP

Play count2

SynApps Registered

2

Navigate to the SynApps dashboard typing the IP Address of your server in the browser (typically the administration dashboard can be reached at: **http://<SA_IP>**



SynApps Dashboard

- Navigate in the **Configuration** → **IP Devices** - Here is where all the active / inactive IP Devices appear, including IPAC devices. By clicking on the settings icon on a specific device it is possible to edit its details, i.e. assigning a name.

IP Address	Status	MAC	Manufacturer	Model	Firmware	Last Seen	
192.168.3.30	Inactive	00:0E:41:07:6C:7a	Barix	IP Speaker	2.7.0	9/15/21 5:44 PM	Settings
192.168.3.113	Inactive	00:0E:41:08:06:6d	Barix	IP Speaker	2.7.0	9/15/21 3:07 PM	Settings
192.168.3.115	Inactive	00:0E:41:08:06:21	Barix	IP Speaker	2.7.0	9/15/21 5:13 PM	Settings
192.168.3.21	Inactive	00:0E:41:08:10:27	Barix	IP Speaker	2.6.2-RC10	7/21/21 4:33 PM	Settings
192.168.3.22	Inactive	00:0E:41:08:11:2d	Barix	IP Speaker	2.6.1	9/15/21 3:36 PM	Settings
192.168.3.123	Inactive	00:0E:41:08:11:75	Barix	IP Speaker	2.7.0	9/15/21 3:08 PM	Settings
192.168.3.31	Active	00:0E:41:08:46:a9	Barix	IP Speaker	2.7.0	9/20/21 11:25 AM	Settings
192.168.3.25	Inactive	00:0E:41:08:46:31	Barix	IP Speaker	2.7.0	9/15/21 3:07 PM	Settings
192.168.3.20	Inactive	00:0E:41:08:54:47	Barix	IP Speaker	2.7.0	9/15/21 3:59 PM	Settings
192.168.3.16	Inactive	00:0E:41:08:55:a0	Barix	Barionet400	Barionet FW v2.17.0.11	7/21/21 4:43 PM	Settings

End of records found.

SynApps IP Devices List

- IPAC Devices are registered as Endpoints in the SA platform, devices capable of receiving audio notifications. Navigate in **Configuration** → **Endpoints**. All endpoints available and ready to receive notifications are listed. If it's the first time the device registers with the SA Server it might take some time to appear in the endpoint list. In some cases the time could be up to 10-12 minutes.

Module	Status	Name	URN	Site	IP Address	Licensed
IPDevice	Inactive	Barix - 192.168.3.113	@IPDevice0009e108056d1	All	192.168.3.113	✓
IPDevice	Inactive	Barix - 192.168.3.115	@IPDevice0009e10806211	All	192.168.3.115	✓
IPDevice	Inactive	Barix - 192.168.3.123	@IPDevice0009e10811751	All	192.168.3.123	✓
IPDevice	Inactive	Barix - 192.168.3.16 - GPO1	@IPDevice0009e10855a01GPO1	All	192.168.3.16	✓
IPDevice	Inactive	Barix - 192.168.3.16 - GPO2	@IPDevice0009e10855a01GPO2	All	192.168.3.16	✓
IPDevice	Inactive	Barix - 192.168.3.16 - GPO3	@IPDevice0009e10855a01GPO3	All	192.168.3.16	✓
IPDevice	Inactive	Barix - 192.168.3.16 - GPO4	@IPDevice0009e10855a01GPO4	All	192.168.3.16	✓
IPDevice	Inactive	Barix - 192.168.3.20	@IPDevice0009e10854471	All	192.168.3.20	✓
IPDevice	Inactive	Barix - 192.168.3.21	@IPDevice0009e10810271	All	192.168.3.21	✓
IPDevice	Inactive	Barix - 192.168.3.22	@IPDevice0009e10811261	All	192.168.3.22	✓
IPDevice	Inactive	Barix - 192.168.3.25	@IPDevice0009e10844311	All	192.168.3.25	✓
IPDevice	Inactive	Barix - 192.168.3.30	@IPDevice0009e1076c1a1	All	192.168.3.30	✓
IPDevice	Active	Barix - 192.168.3.31	@IPDevice0009e10844a91	All	192.168.3.31	✓
Mobile & 3rd Party	Active	Mobile Gateway	@MobileNotifur0	All	n/a	✓

SynApps Endpoint List

5 Prepare a notification: Locate **Notifications** → **Manage** → **New Notification**

Notification Name	Site	Tags	Notification Type
Emergency alert	All	Show	Stored Audio
Iterative Notification	All	Show	Stored Audio
Siren	All	Show	Stored Audio
Test Notification	All	Show	Stored Audio

SynApps Notification List

Under **General** → **Notification** Type select "**Stored Audio**"

GENERAL

Notification Name: Test Notification

Notification Type: Stored Audio

Activation Type: Iteration

Iteration: 1

Priority: 5

Dashboard Icon: None

Available in All Sites: ☒

Enable Time Limit: ☐

SELECT TRIGGERS

Select what activates the notification

MESSAGE DETAILS

Content to send to the endpoints

ACTIONS

Configure buttons for responses and actions, as well as triggering other notifications

ENDPOINT & CONTACT SELECTION

Devices & Contacts that the notification will be sent to

NOTIFICATION DETAILS

Confirm your notification setup

GENERAL

Notification Name: Test Notification

Notification Type: Stored Audio

Priority: 5

Dashboard Icon: None

SELECTED TRIGGERS

MESSAGE DETAILS

Content to send to the endpoints

Audio ID: Show

Stored Audio: Air-Raid-Siren.mp3

Volume: 10

Images: None

Title: None

Body: None

ACTIONS

ENDPOINT & CONTACT SELECTION

Barix - 192.168.3.31

CLOSE SAVE & SEND SAVE

SynApps Notification - General Settings

Under **Message Details** → **Select audio**: pick an audio file from the ones available. i.e "**Air-raid-siren.mp3**"

Intrado

Revolution

Dashboard

Notifications

Scheduler

Configuration

System Status

Users Contacts & Sites

Logging

Global Settings

Help

GENERAL

Notification level settings

SELECT TRIGGERS

Select what activates the notification

MESSAGE DETAILS

Content to send to the endpoints

ACTIONS

Configure buttons for responses and actions, as well as triggering other notifications

ENDPOINT & CONTACT SELECTION

Devices & Contacts that the notification will be sent to

NOTIFICATION DETAILS

Confirm your notification setup

GENERAL

Notification Name

Test Notification

Notification Type

Stored Audio

Priority

5

Establishment Name

SELECTED TRIGGERS

MESSAGE DETAILS

Caller ID

Show

Select Audio

As-Raid-Siren.mp3

✕

Volume

10

Use device default

Select Image

Form Color

Devices without font color support will use their default color

Clear notification from display on completion

Title

Body

CLOSE

SAVE & SEND

SAVE

SynApps Notification - Message Details

Under **Endpoint and Contact Selection** select the IPAC devices that need to receive the notification

intrado

Evolution

Dashboard

Notifications

Scheduler

Configuration

System Status

Users Contacts & Sites

Logging

Global Settings

Help

GENERAL

Notification level settings

SELECT TRIGGERS

Select what activates the notification

MESSAGE DETAILS

Content to send to the endpoints

ACTIONS

Configure buttons for responses and actions, as well as triggering other notifications

ENDPOINT & CONTACT SELECTION

Endpoints & Contacts that the notification will be sent to

Allow users to add endpoints dynamically

None

SELECT YOUR DEVICES & CONTACTS

Search

Endpoints

180 Contacts

User Tags

System Tags

Barrio - 192.168.3.119

Barrio - 192.168.3.115

Barrio - 192.168.3.123

Barrio - 192.168.3.16 - GP01

Barrio - 192.168.3.16 - GP02

Barrio - 192.168.3.16 - GP03

Barrio - 192.168.3.16 - GP04

Barrio - 192.168.3.20

Barrio - 192.168.3.21

Barrio - 192.168.3.25

Barrio - 192.168.3.25

Barrio - 192.168.3.20

Barrio - 192.168.3.21

CHOOSE CONTACT METHODS

If contacts are selected you must choose a communication method. Only contact methods with configured endpoints are selectable. Tap Contacts may be included inside User Tags.

Internal

Mobile & 3rd Party

Email

SMS

WhatsApp

NOTIFICATION DETAILS

Confirm your notification setup

GENERAL

Notification Name

Test Notification

Notification Type

Stored Audio

Priority

5

Dashboard Icon

SELECTED TRIGGERS

MESSAGE DETAILS

Content ID

Show

Stored Audio

Are Read Stored.mp3

Volume

10

Images

Title

Body

ACTIONS

ENDPOINT & CONTACT SELECTION

Barrio - 192.168.3.21

Barrio - 192.168.3.20

Barrio - 192.168.3.25

Barrio - 192.168.3.113

CLOSE

SAVE & SEND

SAVE

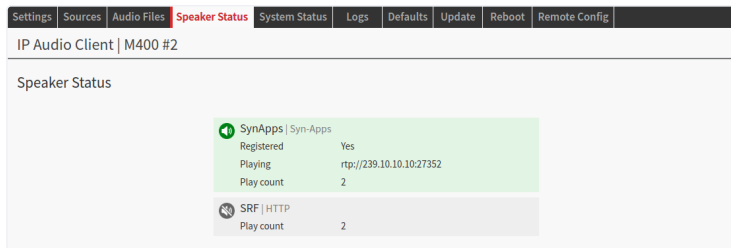
SynApps Notification Endpoint Selection

Now you can **SAVE** or **SAVE & SEND**.

Clicking **SAVE & SEND** will send the notification to the selected endpoints

- 6 Monitor the notification receipt in the Speaker Status of the IPAC device while it is playing. The status will show the RTP Multicast address used by SynApps platform (this address is configurable in the SynApps server)





SynApps Notification while it plays

i For more information on SynApps requirements for firewall configuration visit Intrado SynApps official user manual: https://www.syn-apps.com/downloads/User%20Guides/Revolution_Help/en-us/Content/Installation/System_Requirements.htm

Paging

IPAC devices support 2 paging modes:

1. IC Paging
2. Simple Paging

Both paging solutions are made by Barix and allow any IPAC device to receive audio from any one of the below Paging Master stations:

Paging Master 	Paging Mode 
Paging Gateway M400	IC Paging
Annunicom PS Touch	IC Paging / Simple Paging
Annunicom PS1	IC Paging / Simple Paging
Annunicom 100	IC Paging
Annunicom 200	IC Paging
Simple Paging iOS / Android app	Simple Paging (requires an Exstreamer 100 to be configured as a Gateway in the system)
IC Graph	IC Paging



IC Paging

IC Paging is the Barix Paging solution that works relying on the [Barix Paging Protocol](#), abbreviated BARP. Barix developed a proprietary protocol to allow "BARP devices" to communicate between each other providing audio, control and status messages over a Broadcast or Multicast addresses. An IC Paging system includes:

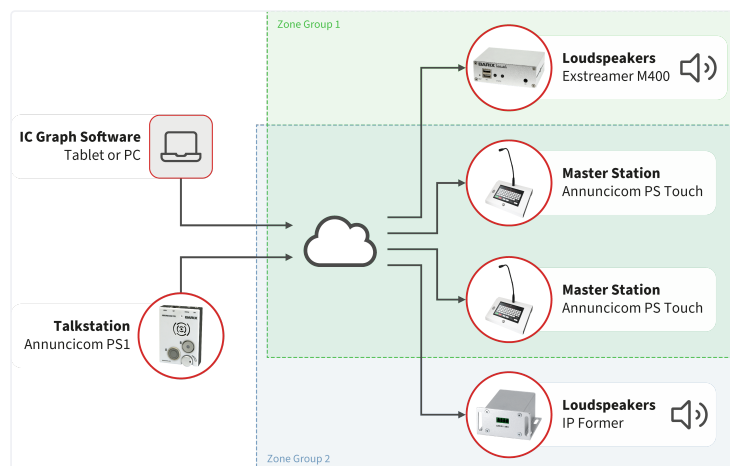
- one or more Master station(s) or ICMaster running [IC Paging firmware](#)
- one or more Endpoint(s) or ICClient running [IC Paging firmware](#)
- one or more Endpoint(s) running IPAC

Firmware Note

IC Paging firmware running on Annunicom or Exstreamer 1xx / 2xx devices is not the same as the one running on IPAC devices. IP Audio Client is a self-contained firmware that implements compatibility with IC Paging devices, it is not an IC Paging firmware like the one deployed on legacy Annunicom or Exstreamer 1xx / 2xx devices

IC Paging implementation on IPAC devices allow IPAC to become endpoints, devices able to receive a stream coming from a master station. A typical example and use case is running the IC Paging firmware on an Annunicom [PS Touch](#) or [PS1](#) or any other Annunicom device supporting IC Paging firmware and configure multiple endpoints to be part of different zones, so that it is possible from a button on a master station (correct name: PTT or Push-To-Talk) to page in the desired endpoints while pressing the button.

Below is a diagram showing the configuration of an IC Paging based solution:



IC Paging Example Layout

IC Paging Source Configuration:



The default configuration of IC Paging when first enabled matches the default configuration of an IC Paging Master device. If nothing is touched the IPAC device would be able to receive streams from the master immediately after submitting the configuration to enable IC Paging

On

Priority2

TypePaging

NameIC Paging

Paging modeIC-Paging

BARP Station ID1

BARP IP Address0.0.0.0

BARP Audio Port5555

BARP Control Port5556

BARP Status Port5557

Sample rate24KHz

Audio Formatμ-law

EQ

Volume90%

IC Paging Source configuration

Item	Description
Name	Modify the name of the priority layer
Paging Mode	Select between IC Paging / Simple Paging (it is not possible to use the 2 modes in different sources, one excludes the other)
BARP Station ID	Unique identifier of the paging endpoint (also called Zone ID in an IC Paging system). Different endpoints can be assigned the same Station ID. In the IC Master device is possible to assign every ID to GROUPS to page different groups of IDs using different calling buttons (number of buttons depends on the capabilities of the IC Master device) <i>Default: empty</i>
BARP IP Address	IP Address used by the endpoint to listen/send on the BARP ports. Modify the BARP IP address to use multicast if communication across different networks is desired, or if your LAN policy is not allowing broadcast. This field cannot be Unicast. The BARP address used by the endpoints should match the one set on the Master station and vice versa. <i>Default: 0.0.0.0 (which corresponds to the broadcast address of the network where the device is installed)</i>



Item 	Description 
BARP Audio Port	Endpoint local port used to listen for incoming audio from the master. It is possible to configure any unused port between 5555-5999 <i>Default: 5555</i>
BARP Control Port	Endpoint local port used to listen for incoming control messages from the master. It is possible to configure any unused port between 5555-5999 <i>Default: 5556</i>
BARP Status Port	Destination port number to send status messages to the master. It is possible to configure any unused port between 5555-5999. <i>Default: 5557</i>
Sample Rate	Configure the sample rate according to the selection applied in the Paging Master station. It offers: 8kHz or 24kHz <i>Default: 8kHz</i>
Audio Format	Configure the audio format according to the selection applied in the Paging Master station. It offers: uLaw or aLaw <i>Default: uLaw</i>

How to setup IC Paging system



- 1 After switching IC Paging Source ON fill in the Station ID field with the desired number. The Station ID is the Zone Group ID used in the BARP protocol to identify one or more endpoint devices. In this example ID 1 is used, all other IC Paging parameters remain with default settings (meaning that BARP will use the Broadcast IP of the network where the device is installed to propagate audio, control and status messages). Click on ***SUBMIT***



On

Priority 2

Type Paging

Name IC Paging

Paging mode IC-Paging

BARP Station ID 1

BARP IP Address 0.0.0.0

BARP Audio Port 5555

BARP Control Port 5556

BARP Status Port 5557

Sample rate 24KHz

Audio Format μ-law

EQ

Volume 90%

IC Paging Source configuration

- On the IC Paging master station (a PS1 device in this example) navigate to **Configuration → Basic Settings**

HOME	CONFIGURATION	STATUS	DEFAULTS
------	---------------	--------	----------

Intercom Master Station

Current Profile

PS1(Ann100/200) / FD

Basic Settings

Advanced Settings

Apply

Cancel

BASIC SETTINGS

BARP Station ID

1001

Duplex Mode

☐ Half
☒ Full

Hands Free

☐ Yes
☒ No

Group 1 Address Map

View/Modify

2 clients selected

IC Paging Master - Basic Settings

On Annunicom PS1 only one button / group can be configured, thus only **Group 1 Address Map** is shown. On a PS Touch or Legacy PS16 Annunicom this would be from 1 to 8. By default each group contains 10 station IDs or zones. So by default our IPAC endpoint will end up being part of Group 1, since its station ID is set to be “1”. Up to 128 station IDs can be selected to be part of a group. To double check which Station IDs are called from a specific group click on **View/Modify** related to the group required, the below station selection window appears. It is possible to add/remove IDs from the group.

HOME	CONFIGURATION	STATUS	DEFAULTS	UPDATE
------	---------------	--------	----------	--------

Intercom Master Station

Current Profile

PS1(Ann100/200) / FD

GROUP 1 SETUP

1

2

3

4

5

6

7

8

9

10

11

12

13

14

15

16

17

18

19

20

21

22

23

24

25

26

27

28

29

30

31

32

33

34

35

36

37

38

39

40

41

42

43

44

45

46

47

48

49

50

51

52

53

54

55

56

57

58

59

60

61

62

63

64

65

66

67

68

69

70

71

72

73

74

75

76

77

78

79

80

81

82

83

84

85

86

87

88

89

90

91

92

93

94

95

96

97

98

99

100

101

102

103

104

105

106

107

108

109

110

111

112

113

114

115

116

117

118

119

120

121

122

123

124

125

126

127

128

Select All

Clear All

Apply

Close



IC Paging Master Group Station Assignment

There are no steps required to perform as by default the correct station ID is part of the Zone Group 1. So when pressing the PTT button 1 on the PS1 the IPAC endpoint is called and automatically plays the audio received by the master station, typically from its microphone input.

- 3 Check the BARP settings on the IC Paging Master station. They can be found under “**Configuration** → **Advanced Settings** → **BARP Control**”. The Station ID of Master stations can be any number as far as it is not one already taken by another device in the system. It is used to notify endpoints about which master is sending the audio and control messages in a multi-master environment. The BARP broadcast address is the BARP IP Address to be used by master stations and endpoints. It has to match on both configurations. When it's 0.0.0.0 means that the Broadcast IP address of the network will be used to propagate audio and control messages and from the master to fetch the status of the endpoints. Multicast can also be used. Unicast is NOT supposed to be used for the purpose of the application.

HOME	CONFIGURATION	STATUS	DEFAULTS
Intercom Master Station			
Current Profile PS1(Ann100/200) / FD			
BARP CONTROL			
BARP Station ID		1001	
Paging Priority		127	
BARP Broadcast Address		0 . 0 . 0 . 0	
BARP Audio Port		5555	
BARP Control Port		5556	
BARP Status Port		5557	
Basic Settings			
Advanced Settings			
Network			
BARP Control			
Audio			
Incoming Call			
Paging			
I/O Control			
Security			
Apply Cancel			

IC Paging Master BARP Control settings

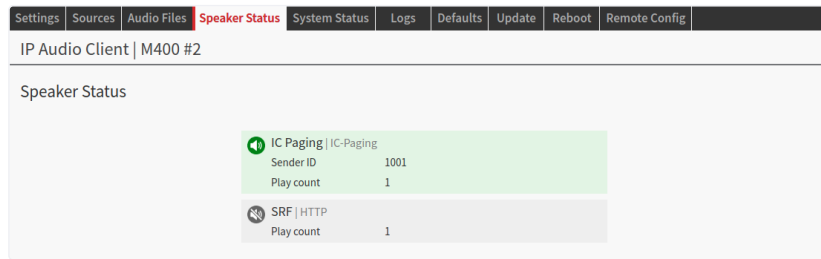
For our example there is no need to change this configuration as it is correct by default.

- 4 Important is to double check the audio format selected in the IC Paging Master. It can be found on the IC Paging Master station under “**Configuration** → **Advanced Settings** → **Audio**”. This is also the default setting on an IC Paging Master device.

HOME	CONFIGURATION	STATUS	DEFAULTS
Intercom Master Station			
Current Profile PS1(Ann100/200) / FD			
AUDIO			
Audio Format		μ-law / 8 kHz (G.711)	
Acoustic Echo Cancellation		☐ On ☒ Off	
Input Source		☐ Line ☒ Mic	
Duplex Mode		☐ Half ☒ Full	
Hands Free		☐ Yes ☒ No	
Microphone Gain		21 dB	
A/D Amplifier Gain		0 dB	
Default Volume		50 %	
Basic Settings			
Advanced Settings			
Network			
BARP Control			
Audio			
Incoming Call			
Paging			
I/O Control			
Security			
Apply Cancel			



- 5 The IPAC endpoint is now ready to receive audio from a master station. To summarize, using a **DEFAULT** configuration on a master station and on an IPAC device, it is only necessary to configure the STATION ID on the IPAC device to be any number from 1 to 10, since the configuration is already applied on IC Paging Master station to page Station ID from 1 to 10 pressing the first button (in case of PS1 device it corresponds to Button “B”) using BARP address 0.0.0.0 and ports 5555 for audio, 5556 for control, 5557 for status, which are the same on all the devices when using the default configuration. When the IPAC device receives audio the Speaker Status changes as follow:



IC Paging Speaker Status while paging

IC Paging Network requirements

IC Paging, by default configuration, uses the network’s broadcast address to:

- Discover other IC Paging devices
- Send control commands (e.g. start / stop page)
- Send status information (e.g. zone busy)
- Audio transport over RAW UDP

In case the broadcast address of the network cannot be used, it is possible to configure IC Paging with multicast. Make sure the network equipment within the subnet where the solution is deployed supports **IGMP** (Internet Group Management Protocol) for multicast to work efficiently.

The ports used are specified in the configuration and can be adapted according to the installation and networking needs. The default ones are:

- Port 5555 used for the audio transport
- Port 5556 used for control messages
- Port 5557 used for status messages

In a multi-master setup, make sure to use different ports for each IC Master.

Simple Paging

Simple Paging is a simplified version of IC Paging, always relying on the [Barix Paging Protocol](#), abbreviated BARP, the Barix proprietary protocol to allow "BARP devices" to communicate between each other providing audio, control and status messages over a Broadcast or Multicast addresses. A Simple Paging system includes:

- one or more Master station(s) or master devices running [Simple Paging firmware](#)
- one or more Endpoint(s) running [Simple Paging firmware](#)
- one or more Endpoint(s) running IPAC

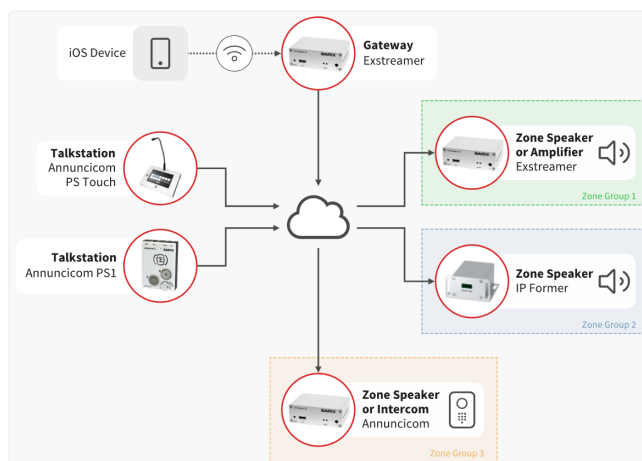
Firmware Note

Simple Paging firmware running on Annunicom or Exstreamer 1xx / 2xx devices is not the same as the one running on IPAC devices. IP Audio Client is a self-contained firmware that implements compatibility with Simple Paging devices, it is not a Simple Paging firmware like the one deployed on legacy Annunicom or Exstreamer 1xx / 2xx devices

Simple Paging implementation on IPAC devices allow IPAC to become Simple Paging endpoints, devices able to receive a stream coming from a master station. A typical example and use case is running the Simple Paging firmware on an Annunicom [PS Touch](#) or [PS1](#) and configure multiple endpoints to be part of different zones, so that it is possible from a PTT button on a master station to page in the desired endpoints while pressing the button.

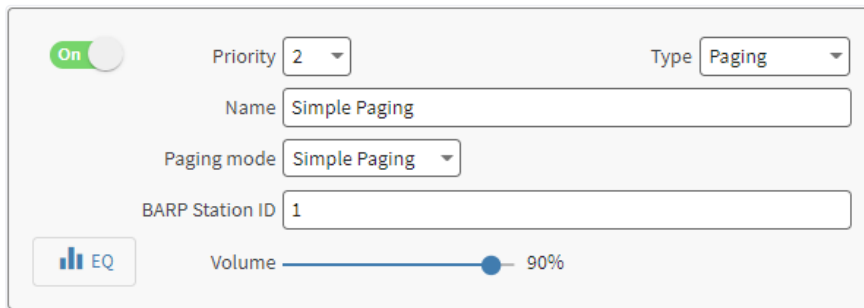
Unlike IC Paging, Simple Paging is easier to configure. It just requires the device to be connected and a page can be immediately established. If "zoning" is required, then it is necessary to modify one field only to assign the devices to their relative groups.

Below is a diagram showing the configuration of an IC Paging based solution:



Note that compared to IC Paging, Simple Paging also support the possibility to page from an app installed on Android or iOS devices. This requires the presence of a [Simple Paging Gateway](#) device in the system running on an [Exstreamer 100](#).

The setup of a Simple Paging source is straight forward:

The image shows a configuration window for a 'Simple Paging' source. At the top left is a green 'On' toggle switch. To its right is a 'Priority' dropdown menu set to '2'. Further right is a 'Type' dropdown menu set to 'Paging'. Below these is a 'Name' text field containing 'Simple Paging'. Underneath the name is a 'Paging mode' dropdown menu set to 'Simple Paging'. Below that is a 'BARP Station ID' text field containing '1'. At the bottom left is an 'EQ' button with a bar chart icon. To its right is a 'Volume' slider set to 90%.

Simple Paging source

There are not parameters beside the BARP Station ID, which is the Zone ID where the endpoint is assigned to receive a page. On the master device it is possible to assign groups of multiple Zone IDs to PTT to perform calls.

Simple Paging Network requirements

A Simple Paging system runs over an IP Network with the followings architecture:

- A single segment Local Area Network
- All nodes on the system has to be on same subnet
- Smart devices has to be part of same subnet of the same local area network

Simple Paging, by default configuration, uses the network's broadcast address to:

- Discover other Simple Paging devices
- Send control commands (e.g. start / stop page)
- Send status information (e.g. zone busy)

While the audio transport relies on a randomly picked multicast address. Make sure the network equipment within the subnet where the solution is deployed supports [IGMP](#) (Internet Group Management Protocol) for multicast to work efficiently.

The ports used are:

- Port 5555 used for the audio transport
- Port 5556 used for control messages
- Port 5557 used for status messages

Simple Paging System Limits

- Maximum 128 end-points in one system
- Maximum 10 masters in one system
- Maximum 1 gateway in one system
- Maximum 5 iPhone/Android Paging Application can page simultaneously deploying the Simple Paging Gateway device (not to be confused with the Paging Gateway M400, which works only with IC Paging)
- Maximum 14 groups can be created on Annunicom PS Touch
- Maximum 2 groups can be created on Annunicom PS1

SIP

SIP stands for [Sessions Initiation Protocol](#) and it is the most used protocol for Unified Communication Systems or VoIP (Voice over IP) solutions. One (not more than one) of the priority layers of any IPAC device can be configured to turn the device into a SIP client via the SIP source and benefit from the broad compatibility with many third party vendors to integrate IPAC devices in infrastructures that are already using SIP as their main communication backbone.

A SIP source on IPAC supports 3 operational modes:

1. SIP Server Mode
2. SIP Peer to Peer Mode
3. Both modes combined

SIP Server Mode

In Server mode the IPAC device allows to enter the values required to register with a SIP server that is used to route and handle the connections and calls between the SIP endpoints. IPAC devices support SIP and SIPS (Secure SIP) protocols (TLS + SRTP) When switching in Server mode the SIP source displays the following options:

On

Priority 1

Type SIP

Name SIP

SIP mode Server

Server Domain 192.168.11.27

Username 102

Password

Advanced Settings

Transport & Security UDP

Port 5060

Half duplex

Registration Timeout (s) 30

Registration Retry Interval (s) 30

Playback Buffer Latency (ms) 100

Capture Buffer Latency (ms) 100

Audio Input Level 50 %

Inbound Calls

Auto Answer

Auto Answer Timeout (s) 0

Hang Up Calls

Call Timeout (s) 0

Stream Timeout (s) 0

Volume 100%

SIP server mode - settings



Item	Description
Server Domain	The server url or IP Address that the IPAC device has to register with. It can be the server hostname or the IP Address <i>Default: "empty"</i>
User Name	The user name for this IPAC endpoint. It is normally the extension number that must be called to reach this extension <i>Default: "empty"</i>
Password	The password for the user name entered and that has to be used to authenticate the registration of the user name with the SIP Server. When it is optional leave this field empty <i>Default: "empty"</i>
Advanced Settings	



Transport & Security	Allows to configure the transport and security protocols to be used to establish the connection and transport data: 1. UDP: Connection establishment (SIP) and media transport (RTP) between the server and the endpoint is transparent and relies on UDP User Datagram Protocol (port 5060) 2. TCP: Connection establishment (SIP) between the server and the endpoint is transparent and relies on TCP User Datagram Protocol (port 5060) 3. TLS: Connection establishment (SIP) between the SIP Server and the endpoint is encrypted, performed using Transport Layer Security. Media transport (RTP) not over TLS, thus un-encrypted (port 5061) 4. TLS + SRTP: Connection establishment (SIP) and media transport (RTP) between the server and the endpoint is encrypted using TLS and SRTP (Secure Real-time Transport Protocol) (port 5061) <i>Default: UDP</i>
Port	Allows to configure the SIP negotiation Port <i>Default: 5060</i>
Half Duplex*	Enable or disable the half-duplex functionality which allows the device to only "talk" or "listen" and not both at the same time. Level: Only available when Half Duplex is enabled. This is a value between 0 and 1 that helps to fine tune the level at which the device enters in "listen" mode. When set to 1, the behavior is similar to a FDX communication. <i>Default: 0.05</i> Timeout: After entering in "listen" mode this is the timeout that can be configured to go back to "talk" mode if no incoming audio is detected. It accepts values from 10 to 1000 ms. <i>Default: 600ms</i>
Registration Timeout	Configure the interval used for the registration with the SIP Server, in seconds. Accepted range: 5 to 300 seconds. <i>Default: 30</i>
Registration retry interval	Configure the registration retry interval in case of registration failure, in seconds. Set to 0 to disable auto re-registration. The re-registration will be done at the retry timeout plus a random time (between 0 to 10 seconds to avoid all clients re-registration at the same time). <i>Default: 30</i>

Playback Buffer (ms)	Sets the desired target buffer for the SIP playback interface. An higher value increases the overall delay gaining on quality, a lower value decreases the delay but the quality of the play-out might be affected by drops or errors. <i>Default: 100</i>
Capture Buffer (ms)*	Sets the desired target buffer for the SIP capture interface. An higher value increases the overall delay gaining on quality, a lower value decreases the delay but the quality of the play-out might be affected by drops or errors. <i>Default: 100</i>
Audio Input Level*	Fine tunes the audio input level to increase or reduce the input level into the SIP call. <i>Default: 50%</i>
Auto Answer*	Enables or disables the possibility to auto answer an incoming call. When auto answer is disabled it is only possible to pick up calls when a Digital Input is closed. <i>Default: enabled</i>
Auto Answer Timeout*	It sets a timeout to automatically pick up an incoming call. Only used when Auto Answer is enabled.
Call Timeout	Configure the maximum duration of an established call, in seconds. Set to 0 to disable. <i>Default: 0</i>
Stream Timeout	Configure the timeout that the SIP client should wait to close the current call in the case it does not detect audio from the remote party. Set to 0 to disable. <i>Default: 0</i>

 *: Parameter only available on IPAC devices supporting bidirectional SIP communication.

SIP Server Response Codes

When in Server mode the Speaker status provide useful feedback on the server replies when the device attempts to register with it, below is an example of successful registration



Settings Sources Audio Files **Speaker Status** System Status Logs Defaults Update Reboot Remote Config

IP Audio Client | M400 #2

Speaker Status

SIP | SIP
Registered Yes
Response Code 200 OK
Play count 0

SRF | HTTP
Play count 1

More information on SIP Response codes [here](#) - Those codes are used to inform the user about connection status between the endpoint and the server.

SIP peer-to-peer Mode

It is the most simple SIP mode, which doesn't require a server for the nature of its working principle. A [peer-to-peer communication](#) is a connection established between 2 endpoints only (the latency in this mode is approximately 147 ms, using an Annunicom 100 as encoder). When in peer-to-peer mode the IPAC device can be called (it is working as a SIP receiver) by any other SIP capable device, the entire connection establishment, setup and dismounting is performed using SIP standards. Below a representation of a source configured in SIP peer-to-peer mode:

On

Priority

1

Type

SIP

Name

Test

SIP mode

Peer-to-Peer

+ Advanced Settings

Volume

100%

To call an IPAC device in P2P mode it is enough to:

- 1 Switch ON one source to be SIP in peer-to-peer mode and click **SUBMIT**

On Priority 1 Type SIP

Name Test

SIP mode Peer-to-Peer

+ Advanced Settings

Volume 100%

- 2 1. On the calling station dial the IP Address of the IPAC device (URL scheme is “**sip:<IPAC_IP>**”). Below is a configuration example of an Annunicom PS1 device running **SIP Client firmware** calling an IPAC device in p2p mode, note the IP Address typed for the “**Input 0 Call ID**” parameter, this means start a call to **sip:192.168.3.31** when the Button “A” (index 0 on a PS1 device) is pressed:

SIP CLIENT

SIP Phone

BASIC SETTINGS

SIP PROTOCOL SETTINGS

Peer to Peer ☐ No ☒ Yes

SIP ID (username) PS1

OUTBOUND CALL SETTINGS

Call on Device Inputs

Input 0 Call ID 192.168.3.31

INBOUND CALLS

Phone pickup mode autohang up after timeout

Pick/hang up time 20 seconds

SIP Client on Annunicom PS1

When pressing the button “A” on the PS1 the call is started to the IP Address of the IPAC device. Pressing the button again terminates the call.

- 3 During the call the speaker status changes indicating the caller details

Settings Sources Audio Files Speaker Status System Status Logs Defaults Update Reboot Remote Config

IP Audio Client | M400 #2

Speaker Status

Icon	Protocol	Current Call	Security	Play count
	SIP	<sip:PS1@192.168.3.37>	No	2
	SRF HTTP			2

Speaker Status on-going SIP call in p2p mode

This source type allows the device to receive an audio stream using the [Real-Time Transport Protocol](#). RTP Playback allows any IP based audio device that is able to stream over RTP to feed audio to IPAC devices. Examples are:

1. BGM Distribution where a [Barix Instreamer](#) feeds one or more IPAC device(s) streaming audio to an RTP multicast address (instead of an Instreamer it is possible to use also VLC or FFMPEG to stream audio over RTP)
2. Audio Distribution where a [Barix Multicoder](#) feeds one or more IPAC device(s) streaming audio to an RTP multicast address
3. [Annunicom PS1](#) running an [Any to All](#) or [IP Intercom](#) firmware streaming voice announcements to one or more IPAC devices
4. [Barionet](#) devices running FFmpeg to stream RTP every time a contact is closed (i.e. to stream an alarm, a pre-recorded message triggered by a switch or a panic button...)

Sources

Input Sources Priority Table Import Export

[+ New source](#)

☒ On

Priority 1 Type RTP

Name

Decoder mode URL

URL

EQ Volume 90%

☒ On

Priority 2 Type RTP

Name

Decoder mode SDP

SDP file

EQ Volume 90%

☒ On

Priority 3 Type RTP

Name

Decoder mode URL

URL

EQ Volume 90%

RTP streaming sources possibilities



Item	Description
Decoder mode	1. URL: setup RTP streaming (unicast, multicast or broadcast) from a URL 2. SDP file: setup RTP streaming from an SDP file (Session Description Protocol)
URL	The RTP Address where the device has to connect to in order to receive the audio stream. It supports multicast or unicast addresses. Multicast URL scheme: <code>rtp://<ip>:<port></code> Unicast URL scheme: <code>rtp://@:port</code> - When using unicast the device waits for incoming audio on the specified port. The “@” symbol is used to indicate the self-IP address of the IPAC device. • When using any RTP static payloads the URL schema is a standard one, there is no need to specify decoder settings: <code>rtp://<IP_ADDRESS>:<PORT></code> (e.g.: <code>rtp://239.0.1.39:5004</code>) • For non-standard payloads the format can be manually set using the following URL schema: <code>rtp://<IP_ADDRESS>:<PORT>?format=[FORMAT]&channels=[CHANNELS]&rate=[RATE]&pt=[PAYLOAD_TYPE]</code> where any one of the following values can be assigned to the fields above: • <i>format</i> = <i>mulaw, alaw</i> • <i>channels</i> = <i>1, 2</i> • <i>rate</i> = <i>8000, 16000, 24000, 32000, 48000</i> • <i>payload type</i> = <i>integer value (i.e. pt=100)</i> Example: play a G711 uLaw 32kHz Mono stream generated by an Instreamer Classic (payload type = 100) use the following URL scheme: <code>rtp://<IP_ADDRESS>:<PORT>?format=mulaw&channels=1&rate=32000&pt=100</code> When the manual format setting is used it overrides the automatic format detection.  HINT: use an application like Wireshark to identify the data format being transmitted by the sending device <i>Default: “empty”</i>
SDP	Uploading an SDP file offers an easy way to setup the decoder to play RTP streams, especially the ones that do not use a standard payload type. When uploading an SDP file ensure that the mandatory attributes are provided. Below is an example of SDP file from a Barix Multicoder M400



Item

Description

SDP File

Upload file

or

s=Stream1
t=0 0
m=audio 5004 RTP/AVP 96
c=IN IP4 239.30.22.1
a=rtpmap:96 L24/48000/2
a=recvonly
a=ptime:1
a=ts-refclk:ptp=IEEE1588-2008:01-23-45-67-89-AB-CD-EF:127
a=mediaclk:direct=0
a=sync-time:0

CancelSave

Default: "empty"

Synchronized Play-out and Delay

The RTP source in IPAC offers a reliable way to play audio synchronized among different devices part of the same installation, with an end to end delay of approx. 190ms when decoding a PCM stream (with MP3 ithe delay is around 240ms)

RTP static payload types

The following is a table listing the Static Payload types, the ones supported without the need to specify the codec to be used in the URL field of the RTP source. For more info on [RTP payload types](#)

Payload type	Name	#Channels	Sampling Frequency	Description	References (RFC)
0	PCMU	1	8000	ITU-T G.711 PCM μ-Law audio 64 kbit/s	RFC 3551
8	PCMA	1	8000	ITU-T G.711 PCM A-Law audio 64 kbit/s	RFC 3551
9	G722	1	8000	ITU-T G.722 audio 64 kbit/s	RFC 3551 - Page 14
10	L16	2	44100	Linear PCM - 16-bit Stereo audio 1411.2 kbit/s, uncompressed	RFC 3551, Page 27

Payload type ↕	Name ↕	#Channels ↕	Sampling Frequency ↕	Description ↕	References (RFC) ↕
11	L16	1	44100	Linear PCM - 16-bit audio 705.6 kbit/s, uncompressed	RFC 3551, Page 27
14	MPA	1,2	32000...48000	MPEG-1 or MPEG-2 audio only	RFC 3551, RFC 2250

Instreamer Classic/100 RTP formats supported:

IPAC devices supports the following formats streamed by an Instreamer Classic/100:

↗

Codec ↕	IPAC Support ↕	URL schema ↕
MPEG1 48kHz (MP3)	supported	auto
MPEG1 44.1kHz (MP3)	supported	auto
MPEG1 32kHz (MP3)	supported	auto
MPEG2 24kHz	supported	auto
MPEG2 22.05kHz	supported	auto
MPEG2 16kHz	supported	auto
G711 uLaw 24kHz mono	supported	?format=mulaw&channels=1&rate=24000&pt=97
G711 uLaw 8kHz mono	supported	auto
G711 aLaw 24kHz mono	supported	?format=alaw&channels=1&rate=24000&pt=98
G711 aLaw 8kHz mono	supported	auto
PCM 24kHz 16bit mono big endian	not supported	
PCM 8kHz 16bit mono big endian	not supported	
PCM 24kHz 16bit mono little endian	not supported	

≡

Codec ↕	IPAC Support ↕	URL schema ↕
PCM 8kHz 16bit mono little endian	not supported	
G711 uLaw 32kHz mono	supported	?format=mulaw&channels=1&rate=32000&pt=100
G711 uLaw 12kHz mono	supported	?format=mulaw&channels=1&rate=12000&pt=109
G711 aLaw 32kHz mono	supported	?format=alaw&channels=1&rate=32000&pt=101
G711 aLaw 12kHz mono	supported	?format=alaw&channels=1&rate=12000&pt=110
PCM 32kHz 16bit mono big endian	not supported	
PCM 12kHz 16bit mono big endian	not supported	
PCM 32kHz 16bit mono little endian	not supported	
PCM 12kHz 16bit mono little endian	not supported	
PCM 44.1kHz 16bit mono big endian	supported	<i>auto</i>
PCM 44.1kHz 16bit stereo big endian	supported	<i>auto</i>
PCM 44.1kHz 16bit stereo little endian	not supported	
PCM 48kHz 16bit stereo little endian	not supported	
PCM 48kHz 16bit stereo big endian	not supported	

Playback latency synchronization

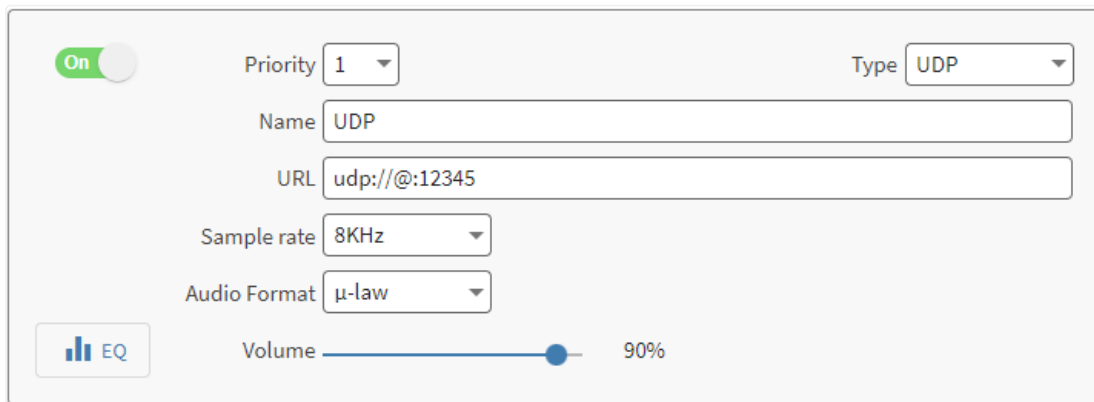


- The playout across multiple IPAC units is synchronized and the end to end delay is approximatively 190ms.

UDP

The UDP source type allows to receive a RAW UDP stream, i.e. to receive the stream generated from a legacy [Annunicom](#) device.

The configuration is quite straightforward:



UDP Source configuration

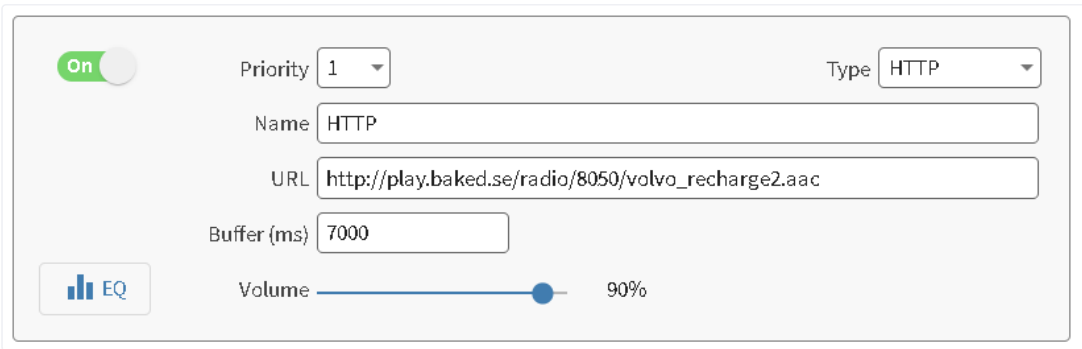


URL	The UDP Address where the device has to connect to in order to receive the audio stream. It supports multicast or unicast addresses. Multicast URL scheme: <i>udp://<ip>:<port></i> Unicast URL scheme: <i>udp://@:port</i> - When using unicast the device waits for incoming audio on the specified port. The “@” symbol is used to indicate the self-IP address of the IPAC device.
Sample rate	The sample rate of the stream to be played. It must match the sender setting. It allows: • 8 kHz • 24 kHz
Audio format	The audio format of the stream to be played. It must match the sender setting. It allows: • uLaw • aLaw



HTTP(S)

The most used protocol adopted by web radios to stream music via Internet. Enabling an IPAC device to connect with a music streaming provider via HTTP(S) transforms the device into a compact and reliable music streaming player. IP Audio Client supports also HTTPS connections to guarantee a secure connection with the music streaming server. Icecast or Shoutcast generated streams are fully supported.



HTTP source configuration (now with buffer)



Item	Description
Protocol	<code>http://</code> and <code>https://</code> HLS streams served over HTTP are supported
Buffer	Sets the playout buffer. (configure a range between 2000 ms-120000 ms) <i>Default - 7000 ms</i> Note: playback starts only after filling the buffer. I.e. if the buffer is set to 120000ms the playback starts only after 2 minutes after applying the setting.

Configuring an HTTP Source is as simple as entering the URL of the web streaming server the IPAC device has to connect to.



Supported stream playlist types	Description
M3U / M3U8	Web streaming server providing an M3U or M3U8 (used mainly to play HLS streams) playlists containing stream URLs are supported. IPAC will parse the M3U text file and will play the URL available in the first line of the file



Supported stream playlist types ↕	Description ↕
PLS	Web streaming server providing an PLS playlist containing stream URLs are supported. IPAC will parse the M3U text file and will play the URL available in the first line of the file

When an HTTP(S) source is playing the Speaker Status reflects the playback status. The play count is an indication of how many times the device reconnected with the streaming server.

MPA400 | IP Audio Client

Speaker Status

 HTTP

Play count 1

USB

This source is only available on devices with a USB port available. Its goal is to allow an easy way to play music (background music) from files stored on the USB pen-drive or stored and signed on a .m3u playlist.

On

Priority

1

Type

USB

Name

USB

Path

folder inside usb device i.e. this/is/my/folder

EQ

Volume

90%

USB audio source configuration

Path	path to a specific music folder inside the pen-drive i. e. <i>/my_music/pop_playlist/</i> . Leave this empty to use the pen-drive base path (the root folder of the pen-drive). In absence of a .M3U file in the path the playback sequence is SHUFFLED. When an .M3U file is present, it is possible to specify the sequence of playback, where each line in the .M3U corresponds to the title of the song to be played, the sequence of playback is from top to bottom. The title must include the extension (i.e. "mySong.mp3").
------	---

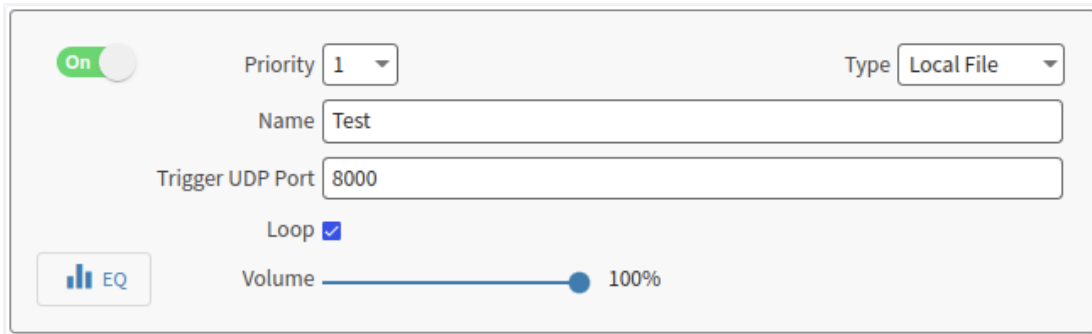


 Note: the application will be restarted if the pen-drive is unplugged. Reboot the device before insert the pen-drive again, hot plug n play is not supported.

Local File

This source enables the user to play a locally stored file to play when the IPAC device receives a command via UDP at the configured port or via the [IP Audio Client API](#) call to play a local file.

On an IPAC device it is possible to upload individual audio files in the Audio Files tab or to connect a USB stick, in this case the files must be stored in a folder called **“triggered_files”** in the USB memory stick (if the IPAC device in use offers physical USB ports). Every audio file stored in the internal memory or from a USB external memory can be triggered to play using the “Local File” source. The working principle of this source type is to configure a port number between 8000 and 8999, IPAC will listen for a command on that port, when received it will immediately play the audio file specified in the command.



Trigger an Audio File from UDP

The datagram of the message required to play the file is:

CODE

```
FILEPLAY=fileName.xxx
```

This command has to be sent at the **<IPAC_IP>:[Configured_Port]** via UDP.

Replace “fileName.xxx” with the name of the file that has to play, “xxx” is the extension format e.g. “mp3”.

To STOP the playback send this command

CODE

```
FILESTOP
```



The commands must always be sent to the configured UDP port in the Local File source.

When the command is sent from a script running on a third party machine double check if it's required to add the escape character “\n” at the end of the command. E.g. when sending the command from a [Barionet](#) device running a Lua script the command written in the script has to be: **FILEPLAY=fileName.xxx\n**

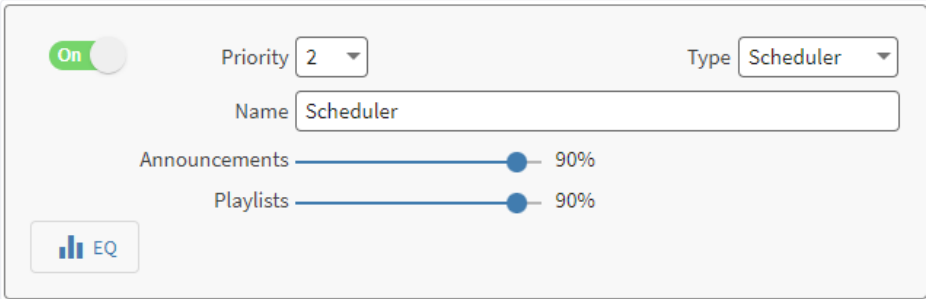
[Here](#) is a nice example on how to trigger the playback also from Windows and Linux machines.

⚠ Supported audio file names: "myFileName.ext" or "myFile Name.ext" - special characters are not allowed and maximum one space character in the name. Two or more spaces are not allowed. Files will fail to upload when not respecting the format expected

📁 Supported audio files formats are: **.mp3, .aac, .flac, .ac3, .dts, .wav, .ogg**

Scheduler

The scheduler source is used to run the scheduled items configured in the Scheduler page



The screenshot shows the configuration for a 'Scheduler' source. It includes a green 'On' toggle switch, a 'Priority' dropdown set to '2', and a 'Type' dropdown set to 'Scheduler'. The 'Name' field contains 'Scheduler'. Below these are two volume sliders: 'Announcements' and 'Playlists', both set to 90%. An 'EQ' button with a bar chart icon is located in the bottom left corner of the configuration box.

Scheduler source

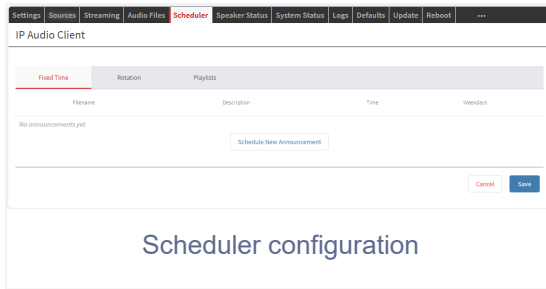
The scheduler embeds 2 different types of assets:

- Announcements: short audio clips, either fixed time or rotational
- Playlists: typically music tracks, arranged in playlists

In the source configuration it is possible to set the volume of the 2 asset classes independently. The change is applied in real time.

Scheduler Configuration

 Before tests **always** check that date and time are correct.



Fixed Time	Rotation	Playlists
Filename	Description	Time
No announcements yet		
Schedule New Announcement		
Cancel Save		

Scheduler configuration

In order to run a scheduler, there must be one configured. Locate the Scheduler tab in the top navigation bar. IP Audio Client offers 3 types of scheduler:

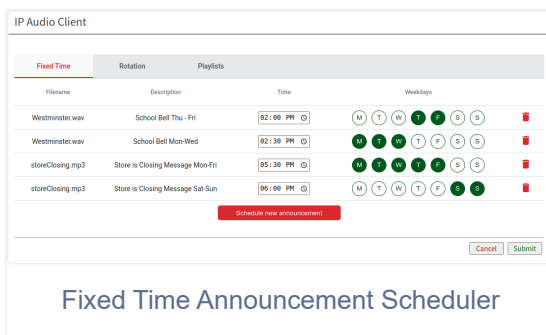
1. Fixed Time
2. Rotation
3. Playlist

All the schedulers work on a weekly basis.

The 3 types of schedulers can play simultaneously by the same scheduler source, for this reason they respect an intrinsic priority:

- Fixed Time and Rotation: in case there is a Fixed Time announcement and a Rotation message set to play at the same time, the files are queued, the fixed time announcement plays first and the rotation plays after.
- Playlist: music scheduled in the playlist scheduler plays always in absence of any announcement scheduled. Announcements always play on top of any scheduled playlist. Meant to be used as a BGM material.

Fixed Time



Fixed Time	Rotation	Playlists
Filename	Description	Time
Westminster.wav	School Bell Thu - Fri	02:00 PM
Westminster.wav	School Bell Mon-Wed	02:30 PM
storeClosing.mp3	Store is Closing Message Mon-Fri	05:30 PM
storeClosing.mp3	Store is Closing Message Sat-Sun	06:00 PM
Schedule new announcement		
Cancel Submit		

Fixed Time Announcement Scheduler

This type of scheduler is suitable for announcements that must be scheduled to play at a specific fixed time of the day. E.g. bells or informative messages.



To create an new announcement:

1. Click on the "*Schedule New Announcement*" button.
2. The dialog the shows up allows to select a file from the ones already uploaded in the Audio Files tab OR to upload a new file to be used right in the Fixed Time Scheduler.
3. A description can be added (not mandatory).
4. Set the Play Time by typing the time manually or using the time picker, select the days of the week when the announcement must play and SAVE.

Announcements created appear as a list. They can be modified at any time using the provided controls:

- Time: change the time when an announcement is scheduled to play.
- Weekdays: change the weekly days when the announcement must play.
- Edit: used to change a scheduled file and the description.
- Copy: copy and paste an announcement on the next line to quickly setup the same announcement to play on different time and days.
- Remove: remove a scheduled announcement.

Once the scheduler is complete use the SUBMIT button to confirm and apply the scheduler configuration.

Rotation

The screenshot shows the 'IP Audio Client' interface with the 'Rotation' tab selected. It features a table for 'Rotation Clips' with columns for Filename, Description, Enable, and Playback order. Below this is a 'Schedule Rotation' section with a table for 'Rotation weekly time frames' showing start and end times for each day of the week. There is also an 'Interval' dropdown set to 20 minutes and a 'Restarts daily' checkbox. At the bottom, there are 'Cancel' and 'Submit' buttons.

Filename	Description	Enable	Playback order
Conv02.wav	Covid distance reminder	☒ Yes	1
Conv03.wav	Wash your hands reminder	☒ Yes	2
Imperial_march.mp3	Promo Message	☒ Yes	3

[Add clips to the Rotation](#)

	Start	End
Monday	08:00 AM	08:00 PM
Tuesday	09:00 AM	07:00 PM
Wednesday	08:00 AM	06:00 PM
Thursday	09:00 AM	07:00 PM
Friday	08:00 AM	06:00 PM
Saturday	07:00 AM	12:00 PM
Sunday	09:00 AM	12:00 PM

Interval: 20 minutes

Restarts daily: ☒ Yes

[Cancel](#) [Submit](#)

Rotation Advertisement Scheduler

This type of scheduler is suitable for advertisements or messages to be played following a rotational pattern. To create a new rotation:

1. *Add clips to the Rotation* using the corresponding button



2. Configure the order of playback for the files added to the rotation, which file has to be the first, the second, etc.
3. Use the enabled/disabled flag to temporary disable a clip in the rotation (i.e. if an ad for chocolate bars is playing in the shop but the inventory for chocolate bars is out of stock).
4. Any clip in the rotation can be modified any time with the pencil icon or removed with the bin icon.

Once clips are in the rotation it's time to schedule the rotation:

- It is possible to set a time window for each day of the week when the rotation starts and ends.
- Use the copy buttons if the time frame is the same across multiple days.
- Set the interval of playback, which is the time in between every clip part of the rotation. E.g. Interval is set to 15 minutes. The first clip will play, the second clip will play after 15 mins, the third clip after 15 mins, etc.
- Restarts daily forces the rotation to start always from the first clip every day.

Once the scheduler is complete use the *SUBMIT* button to confirm and apply the scheduler configuration.

Playlist

Name	Description	Start Time	End Time	Weekdays
Morning playlist	Morning music to start the day wonderfully	08:00 AM	01:00 PM	Mon, Tue, Wed, Thu, Fri, Sat, Sun
Afternoon Playlist	Enjoy working while listening to your favorite music	01:00 PM	06:00 PM	Mon, Tue, Wed, Thu, Fri, Sat, Sun
Evening playlist	Have fun!	06:00 PM	11:00 PM	Mon, Tue, Wed, Thu, Fri, Sat, Sun
Rock Music	Best rock of 2007	08:00 AM	11:00 PM	Mon, Tue, Wed, Thu, Fri, Sat, Sun

[Schedule a Playlist](#)

[Cancel](#) [Save](#)

Playlist scheduler

This type of scheduler allows the possibility to schedule playlists of files.

- Before scheduling a playlist it is necessary to create one in the Audio Files Playlist tab.

How to create a playlist schedule:

1. Click on "Schedule a Playlist" to create a new playlist.
2. Select the playlist (retrieved from the playlists created in the audio files playlist tab).

3. Set the time window when the playlist must start and when it has to stop. Use the time picker or manually type the time in the fields.
4. Select the days when the playlist must play and click on "SAVE" to confirm.

The list of scheduled playlists shows all the currently scheduled items.

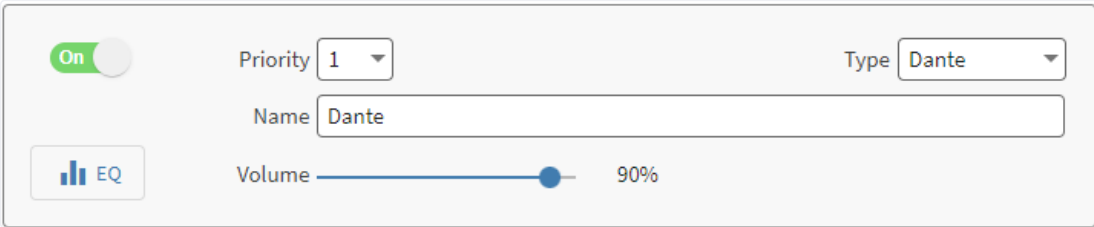
It is possible anytime to modify the start time, end time and the weekdays of the scheduled playlist.

Use the bin icon to remove a playlist from the scheduler configuration.

Once the scheduler is complete use the SUBMIT button to confirm and apply the scheduler configuration.

Dante® (source)

This source type allows to use the IPAC device as a 2 channels Dante® receiver.

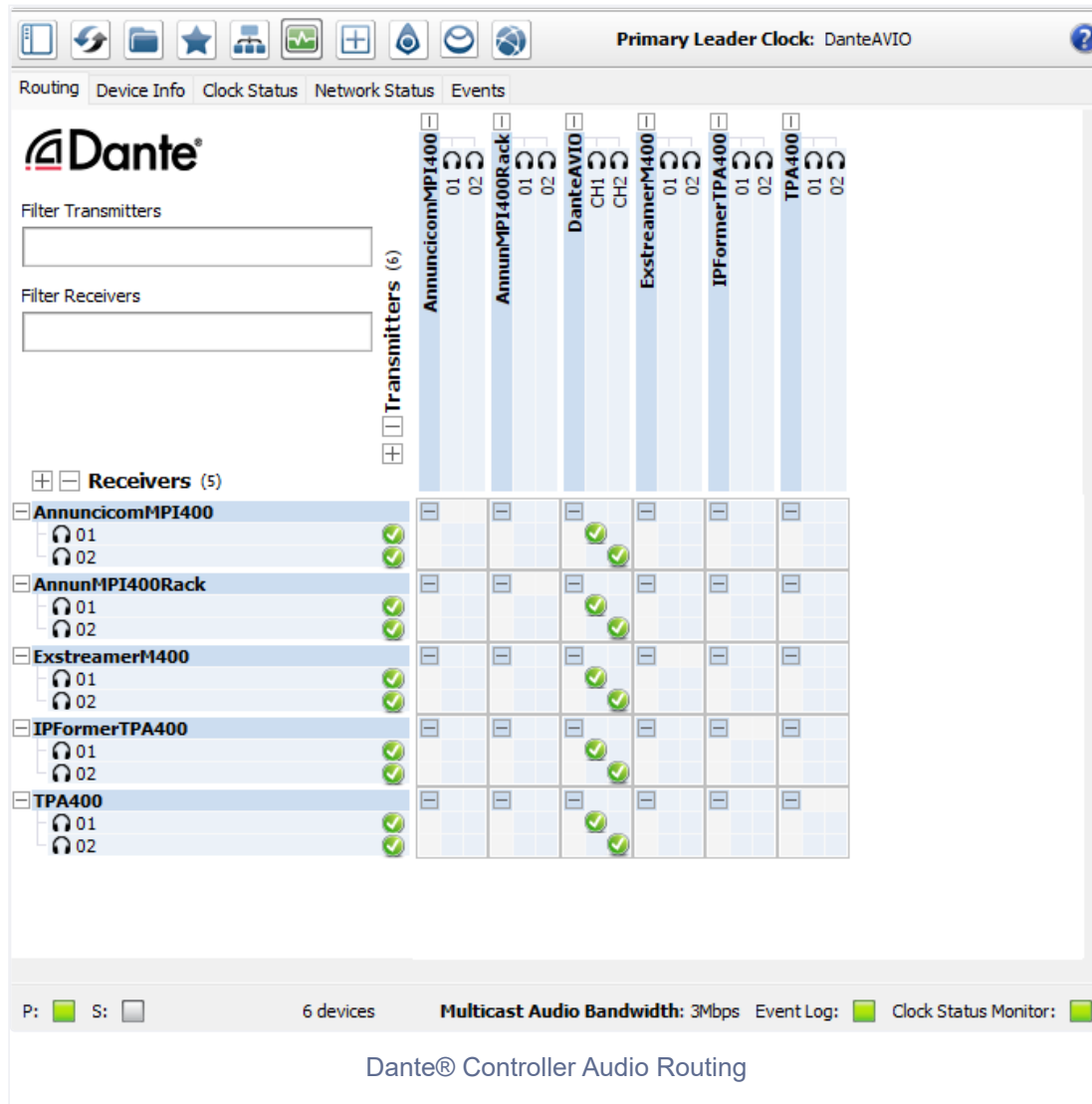


The screenshot shows a configuration panel for a Dante® Audio Source. It includes a green 'On' toggle switch, a 'Priority' dropdown menu set to '1', and a 'Type' dropdown menu set to 'Dante'. Below these is a 'Name' text input field containing the word 'Dante'. At the bottom left is an 'EQ' button with a bar chart icon. To its right is a 'Volume' slider set to 90%. The entire configuration area is titled 'Dante® Audio Source' at the bottom.

Dante® License

Dante® requires a license released by Audinate Pty Ltd. in order to run on IP Audio Client. Make sure to order the right PN when purchasing the IPAC device. It is not possible to install a Dante® license on a device that was manufactured without it.

The routing of the audio sent to IPAC is managed in the Dante® Controller where the device, by default configuration, appears with the name *"barix-last 6 digits MAC"* (i.e. "barix-09cfab"). It is possible to change the name in the device config section inside the Dante® Controller. Below is an example of IPAC devices receiving a 2 channels Dante® stream from a Dante® AVIO 2x0 (2 channels analog input adapter):



Most of the monitoring and reporting is also achieved inside the Dante® Controller in the Device Info, Clock Status and Network Status tabs.

Sample Rate and Encoding Format adjustments are performed on a device basis in the Dante® Controller. Double-Click on the interested device in the routing pane, a new window appears, locate the 'Device Config' tab:

TPA400

Rename Device

TPA400 Apply

Sample Rate

Sample Rate: 48k Sample Rate Pull-up:

This device does not support sample rate configuration. This device does not support sample rate pull-up configuration.

Encoding

Encoding: PCM 24

This device does not support encoding configuration.

Clocking

Unicast Delay Requests: Disabled

Device Latency

Latency: 4.0 msec

Reset Device

Reboot Clear Config

IP Former TPA400 Dante® configuration

IPAC devices allow a sample rate of 48 kHz, while the encoding format is 24 bits.

Warning about Latency

The Dante® implementation on IPAC targets a flexible usage of the device which allows to combine multiple sources assigned at different priority layers. Due to the nature of this architecture the IPAC Dante® receiver plays with a higher latency compared to other Dante® enabled products, the latency reported in Dante® Controller doesn't include the latency added by the IPAC decoder chain. Key applications for IPAC Dante® devices: BGM distribution, paging over Dante®, Dante®-SIP bridging, Dante®-RTP bridging, all 'non-live' applications.

For more information on Dante® Controller and all the features offered please check the [Audinate's website here](#)

To download Dante Controller visit [Audinate's website here](#)

Useful Links from Audinate's website: [Audinate FAQs - Network Requirements and other useful information](#)

Audio Files can be stored in the internal memory of the device or they can be played from an external USB memory storage connected to the front panel USB ports (for IPAC devices supporting the USB feature). The audio files page is where the user can upload files or check files present on an external USB memory.

SettingsSourcesStreamingAudio FilesSchedulerSpeaker StatusSystem StatusLogsDefaultsUpdateRebootRemote Config2022-08-05 09:50:16+0300M400_FLEMAHACFW 2.9.0-RC5

IP Audio Client

BARIX

Audio Files

Storage

Used: 17.8 MB

Available: 5.8 GB

Total: 5.8 GB

Upload Audio Files

Audio Files

Search

☐	Name	Notes	Duration	Size
☐	Summer of Love (Shawn Mendes & Taylor)		00:02:59	2.7 MB

Items per page: 51 - 1 of 1< > >>

Playlists

Name	Description	Shuffle	Number of files	Total duration
Rock Music	Best rock of 2000'	No	0 file	0s
Morning playlists	Morning music to start the day wonderfully	No	0 file	0s
Afternoon breeze	Enjoy working while listening to your favorite music	No	0 file	0s
Evening playlist	Have fun!	No	0 file	0s

Create New Playlist

USB Files

02_Torredor.mp3

Assets

Music

Triggered_files

Help

Audio Files

In this tab it is possible to upload audio files in the local memory of the IP Audio Client device. A friendly widget at the top of the page informs about the total space used and available to be allocated for audio files storage.
Supported audio formats: mp3, aac, flac, ac3, dts, wav, ogg and m4a. Each file doesn't have to exceed 256MB.
Click on "Upload Audio Files" to select the files to be uploaded in the device. It is allowed multi-selection using the CTRL key to upload multiple files at once.
Once an audio file is uploaded it becomes available in the "Audio Files" section. This list is arranged in blocks of 5 files, use the arrows in the lower right corner to scroll through all the files when there are more than 5 files uploaded.
Use the search function on the top left corner to filter files containing characters typed here.
To remove a file click on the checkbox at its left and on the bin icon that appears.
For each file it is possible to edit the name, add notes, check its duration and size.
Audio Files uploaded in the internal memory can be used by the LOCAL FILE and by the SCHEDULER source.
Playlist
In this area it is possible to arrange audio files into playlists that can be later scheduled.
Click on "Create New Playlist" to create a playlist. Assign a name and a description (not mandatory) then Add Files To Playlist.
It's possible to select files already uploaded or upload files in the playlist directly from here. All files part of the playlist are shown in a list made up of block of max. 5 files each, use the arrows to scroll through all the files added to the playlist.
When files are added it is possible to modify the playback order using the UP / DOWN arrows on the right side. Use the "Shuffle Playback" option to randomly play files from the playlist.
Use the SAVE button to confirm the creation of the playlist. Once created it is possible to edit it again using the pencil icon or remove it using the bin icon.
Playlists are meant to be used by the SCHEDULER. Once a playlist is created it is possible to schedule its playback in the scheduler playlist tab.
Make sure to SUBMIT to apply the changes after a playlist has been created / modified.
USB
When a USB pendrive is inserted in one of the USB ports of the IP Audio Client device (only devices supporting USB ports) here are visualized the files contained in it.
The USB audio files can be used by the USB source or by the LOCAL FILE source.
In case they must be used by the LOCAL FILE source, to successfully trigger the playback of a file, they have to be stored in a folder named "triggered_files" saved in the base path of the pendrive.

The page is organized as follow:

- **Audio Files:** List of audio files already stored in the local memory of the device. The maximum number of files that can be uploaded depends on the available memory available. A useful widget in the top of the page helps monitoring the amount of space being used
- **Playlists:** Here is where audio files can be arranged into playlists, or group of files that can be playback following a sequence
- **USB Files:** If a USB pen-drive is attached to one of the USB ports of the IPAC device, its content is listed here.

Internal Audio Files

In order to upload audio files click on "Upload Audio Files" to select the files to be uploaded in the device. It is allowed multi-selection using the CTRL key to upload multiple files at once.

Audio Files

Search

☐	Name	Notes	Duration	Size
☐	Summer of Love (Shawn Mendes & Taylor)		00:02:59	2.7 MB
☐	Spring Love		00:00:01	9.6 KB
☐	Along Along Love		00:00:01	14.2 KB
☐	Love Love		00:00:02	15.7 KB
☐	Love Love		00:00:02	15.7 KB

Items per page: 51 - 5 of 5< > >>

Audio Files List



- Once an audio file is uploaded it becomes available in the "Audio Files" list. This list is arranged in blocks of 5 files, use the arrows in the lower right corner to scroll through all the files when there are more than 5 files uploaded. Use the search function on the top left corner to filter files containing characters typed here
- To remove a file click on the checkbox at its left and on the bin icon that appears right after
- For each file it is possible to edit the name, add notes, check its duration and size
- Preview function: clicking on the PLAY button on the left side of the filename will preview the file, the playback will start immediately on top of anything that is being currently playing by the IPAC device.

 Individual file size when uploading in the internal memory: 256MB

 Supported audio files formats are: **.mp3, .aac, .flac, .ac3, .dts, .wav, .ogg**

Playlist

In this area is possible to arrange audio files into playlists that can be later scheduled.

Create New Playlist

Name

My Playlists

Description

Morning glory - best music for morning times

Shuffle files

Search

☐	Name	Notes	Duration	Size	Playback order
☐	2:08 Previously On Other People's Heartache...m4a		00:01:06	2.7 MB	↓
☐	1:11 The API Song.m4a		00:02:54	2.9 MB	↑ ↓
☐	01:21 Things I Want In A Lover.m4a		00:03:28	3.3 MB	↑ ↓
☐	02 Narcissus.m4a		00:03:38	3.4 MB	↑ ↓
☐	1:04 Hey Mama.m4a		00:03:34	3.6 MB	↑ ↓

Items per page: 5 1 - 5 of 5 |< < > >|

+ Add File To Playlist

Cancel Save

Playlist creation

To create a playlist:

1. Click on "Create New Playlist" to create a playlist.
2. Assign a name and a description (not mandatory)

3. Add Files To Playlist. It's possible to select files already uploaded or upload files in the playlist directly from here.
4. All files part of the playlist are shown in a list made up of block of max. 5 files each, use the arrows to scroll through all the files added to the playlist.

When files are added is possible to modify the playback order using the UP/DOWN arrows on the right side. Use the "Shuffle Playback" option to randomly play files from the playlist.

Use the SAVE button to confirm the creation of the playlist. Once created it is possible to edit it again using the pencil icon or remove it using the bin icon.

Playlists				
Name	Description	Shuffle	Number of files	Total duration
Rock Music	Best rock of 2007	No	0 file	0s
Morning playlist	Morning music to start the day wonderfully	No	0 file	0s
Afternoon playlist	Enjoy working while listening to your favorite music	No	0 file	0s
Evening playlist	Have fun!	No	0 file	0s
My Playlists	Morning glory - best music for morning times	Yes	6 files	10m 15s

[Create New Playlist](#)

Playlist List

Playlists are meant to be used by the [SCHEDULER](#). Once a playlist is created it is possible to schedule its playback in the scheduler playlist tab.

Make sure to SUBMIT to apply the changes after a playlist has been created / modified.

USB Files

This section in the Audio Files tab is present only if a USB pen drive with valid content is attached to the IPAC device.

USB Files	
- Assets - Music 01 That's Not Her Style.mp3 02 The State of Massachusetts.m4a - album.wav - triggered_files - Cont02.wav	

USB Files content

The list of folders and files is available for the user to scroll.

To play USB files using the LOCAL FILE source, which triggers the playback via UDP or REST API, ensure that the files are stored in a folder called: **triggered_files**

Check [Local File Source](#) and [USB source](#) details for more information

Handle USB pen drives

When using external USB memories it's possible to attach them to the USB port while the device is either powered on or off but to eject or remove a USB there is

no “Eject” function available, thus is necessary to power off the IPAC device and remove the USB afterwards, this will ensure no files on the USB will be corrupted.

Streaming

In the streaming tab are present the configuration options to stream the MAIN audio output (with all intrinsic source configuration) to network destinations via RTP or Dante® (only for licensed devices).

The screenshot shows the 'MPI400 | IP Audio Client' interface. Under the 'Streaming Options' section, there is a 'Forwarding Enable' toggle switch which is currently turned off. Below it, the 'Protocol' dropdown menu is open, showing 'RTP' as the selected option and 'Dante' as an alternative. The 'Destination URL' field is empty. The 'Audio Format' dropdown menu is also open, showing 'PCM 16bit 44.1kHz ste...' as the selected option. At the bottom of the interface, the text 'IPAC - streaming options' is displayed.

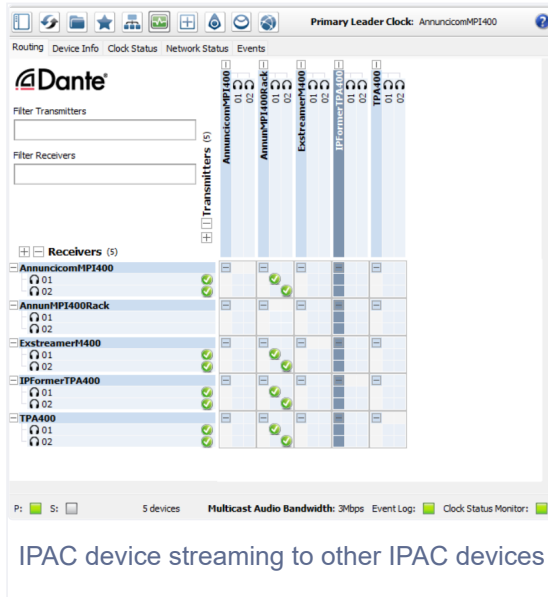
Dante® Forwarding

It allows to stream the main output to Dante®, thus enabling the IPAC device to become a 2 channels Dante® transmitter.

Dante® License

Dante® requires a license released by Audinate Pty Ltd. in order to run on IP Audio Client. Make sure to order the right PN when purchasing the IPAC device. It is not possible to install a Dante® license on a device that was manufactured without it.

When the Dante® Forwarding is enabled on an IPAC device, its local audio output is muted and the audio routing is fully managed by the Dante® Controller. In the picture below is an Annuncicom MPI400 streaming to other IPAC devices with a Dante® source configured to receive the stream.



This functionality allows to use an IPAC device as a 'Dante® gateway' since it is possible to setup multiple sources like: Paging, InformaCast, SynApps, SIP, Scheduler etc.. and send the output to a Dante® system.

These are all the possibilities available:

- RTP: IPAC to receive an RTP stream and forward it to Dante®
- InformaCast: IPAC to receive an InformaCast notification and forward it to Dante®
- SynApps: IPAC to receive a SynApps notification and forward it to Dante®
- Scheduler: IPAC to run a message/bell/ads/BGM scheduler and forward it to Dante®
- SIP: IPAC to receive a SIP call and forward it to Dante® / IPAC to feed a SIP call with Dante® audio => This allows for remote bidirectional interconnection of 2 Dante® systems.
- UDP: IPAC to receive an UDP stream and forward it to Dante®
- USB: IPAC to play from USB and send its audio to Dante® (simple USB BGM solution to Dante®)
- HTTP: IPAC to receive an HTTP stream from a web radio and forward it to Dante®
- Paging: IPAC to receive an IC Paging / Simple Paging stream and forward it to Dante®
- Local Files: IPAC to play local files triggered via API and forward to Dante®

Of course it's possible to combine multiple sources (up to 5 layers can be configured in IPAC) and the whole output can be forwarded to Dante®.

RTP Forwarding

This function forwards the MAIN audio output of the IPAC device to a configured destination via RTP. Other IP Audio Client devices in the same installation can play the content forwarded



by this IP Audio Client. This enables the creation of setups where one IPAC device (e.g.) operate as a “gateway” running multi-source configurations like: InformaCast, SIP, Paging, RTP, HTTP and all its traffic is then routed back in the network to other IPAC client devices.

– Streaming Options

Forwarding Enable

Protocol

RTP

Destination URL

rtp://239.0.1.1:5004

Audio Format

PCM 16bit 44.1kHz ste...

Streaming options - RTP Forwarding

If the destination URL is a mutlicast address the IPAC device will become a client of itself, it will join the same RTP stream it forwards and will play it at the analog audio output.
If the destination URL is a unicast address this IPAC device will go silent and will only stream its output to the destination URL.

Item	↕	Description	↕
Enable/Disable		Enables or disables the RTP Forwarding function	
Destination URL		The destination IP address where to forward the stream over RTP. It can be a Multicast/Unicast IP Address with the "rtp://" and the port number specified i.e. "rtp://239.0.1.1:5004"	
Audio Format		It is possible to choose the audio format of the forwarded stream among: • PCM 16 bit 44.1kHz stereo • MP3 128kbps 44.1kHz stereo • G722 16kHz mono • G711 uLaw 8kHz mono • G711 aLaw 8kHz mono	

 When playing an RTP stream forwarded by an IPAC device on other IPAC units in the same installation, the end to end delay is approximatively 190ms and the playout across different devices is in sync.



Inputs/Outputs

i This page is presented only on devices offering digital inputs and digital outputs or relay outputs hardware support. (e.g. Annunicom MPI400).

SettingsSourcesStreamingAudio FilesSchedulerSpeaker StatusI/O Dante StatusSystem StatusLogsDefaults...

MPI400 | IP Audio Client

Input/Output

– Inputs

Input 1

Operational ModeSIP

Peer to Dial101

Input 2

Operational ModeSIP Press-Hold-Release

Peer to Dial102

– Relays

Relay 1

API control

Relay 2

API control

Enable relay on

☐ SIP

☒ Local File

☐ Dante

☒ UDP

☐ Simple Paging

Cancel

Submit

Annunicom MPI400 IO tab

Inputs

This section allows the configuration of the Digital Inputs (dry contacts). For each Digital Input physically detected on the device it is possible to independently set:

- **Operation Mode**
 - **SIP**: When the contact is closed, the input state is lowered and a call to the configured peer is initiated. The input can then go back to its initial state. To hangup the call the input must be closed one more time.
 - **SIP Press-Hold-Release**: When the contact is closed, the input state is lowered and a call to the configured peer is initiated. The call is kept open until the input state doesn't change, when this happen the call is closed.



- Calling:
 - [Press] Input goes high (1) => Call is started towards the destination defined
 - [Hold] Input stays high (1) => Call is held
 - [Release] Input goes low (0) => Call is closed
- Receiving:
 - [Press] Input goes high (1) => Call is picked up
 - [Hold] Input stays high (1) => Call is held
 - [Release] Input goes low (0) => Call is closed
- **SIP Refer Blind Transfer:** In this mode the Input can be used to transfer the ACTIVE call to the configured "Peer to Dial". For this to happen the call must be ACTIVE. SIP Blind Transfer (RFC5589) doesn't check for the availability of the remote peer where the call is being transferred, it simply transfers the call as soon as instructed by the Digital Input state. When the Input is closed and used to Blind Transfer the call, the ACTIVE call is closed and transferred.

 If your PBX or SIP Server already have Forwarding rules configured they can affect the functionality of SIP Refer Blind Transfer.
Not all the SIP Servers support this mode.

Peer to Dial: Is the remote party, the peer to be called when the input is closed. When SIP Source is configured in "Peer to Peer" or "Server and Peer to Peer" it's possible to dial an IP ADDRESS. When in "Server" or "Server and Peer to Peer" mode it is possible to dial a SIP Extension that belongs to the same domain.

- **Receiving Calls**

1. When a Call is received and SIP Auto Answer is ENABLED (see SIP Source Advanced configuration): any Digital Input can be used to close the active call
2. When a Call is received and SIP Auto Answer is DISABLED (see SIP Source Advanced configuration): any Digital Input can be used to pick up the call, consequently any Digital Input can be used to close the active call.

This section allows the configuration of the trigger detection to change the current state of the relays. Consult the User Guide of the HW device being used to verify the default state of each relay.

E.G. on [Annunicom MPI400](#) the default condition is:

- Relay 1: Normally Open
- Relay 2: Normally Closed

For each Relay is possible to specify the trigger condition as follow:

- **Enable Relay On:** this option offers an easy way to select which source, among the ones configured in the sources tab, will control the relay when it is activated (i.e. when there is an incoming audio)
- **API Control**
 - It allows the possibility to enable the control of the corresponding relay via REST API. The API control is independent from any other control that might be already assigned to the Relay.
Check the [API documentation](#) for more details.

Intercom

 This tab is presented only on devices supporting Intercom Stations functionalities.
E.g. [Annunicom AHE YA404](#)

IPAC devices implementing this feature present multiple “Station” outputs (up to 16 for the Annunicom AHE YA404), each one supporting an Intercom panel. Typical applications:

- Paging / intercom device for classrooms, hospitals, elevators, factories, public transportation, prisons etc.
- Door Intercom station
- Help Point/SOS call box device
- Generic, bidirectional half-duplex VoIP module for Paging and Intercom applications



Example Intercom Panel supported - VRIC-3G-4525

The system operates using two distinct audio channels, each serving specific communication functions:

1. **Paging Channel (the output of the sources configuration)**

The Paging Channel serves as the default audio output for all stations when no active SIP intercom calls are in progress. This channel continuously broadcasts audio content configured through the **SOURCE** tab in the web interface. Common applications include background music, announcements, or ambient audio distribution to all connected stations.

2. **Intercom Channel**

The Intercom Channel activates when a station initiates communication with the central station. This channel takes priority over the Paging Channel and is configured through the Intercom tab in the web interface. The system supports a maximum of one station on the intercom channel at any given time and uses SIP to establish connections with a remote central.

Intercom Operation Characteristics

Audio Direction Control

When a station is connected via the intercom channel, audio transmission operates in half-duplex mode. The direction of audio flow is automatically determined by audio presence detection at the central station:

- **Station-to-Central:** When no audio is detected from the central station, the station's speaker functions as a microphone, transmitting room audio to the central station.



- **Central-to-Station:** When audio is detected from the central station, it takes priority and is transmitted to the station's speaker.

Dynamic Role Switching

The station's speaker unit dynamically switches between output (speaker) and input (microphone) modes based on the communication direction, enabling natural two-way conversation flow.

Typical Operation Workflow

Standard Operation Sequence

1. **Default State:** All stations receive audio from the Paging Channel (e.g., background music from an HTTP / RTP source or announcements from a Scheduler source).
2. **Call Initiation:** A user activates the station's input mechanism to initiate communication with the central station.
3. **SIP Call Establishment:** The system establishes a SIP call between the requesting station and the designated central extension.
4. **Initial Communication Phase:**
 - The central station is initially muted upon call connection
 - The station's speaker operates as a microphone, transmitting room audio to the central station
 - Central personnel can hear audio from the requesting station's location
5. **Bidirectional Communication:**
 - When central station personnel speak, the station's speaker returns to output mode
 - Audio from the central station is transmitted to the requesting station
 - Communication continues with automatic direction switching based on audio presence
6. **Call Termination:** Upon call completion, the station automatically returns to receiving audio from the Paging Channel.

Group Paging

By configuring a SIP Source in the SOURCES tab it will be possible to use a dedicated extension for an ALL station page.



When doing so, the channel used is the paging channel which drives ALL stations with the same signal.

It is NOT possible to perform a group page for a subset number of stations.

Intercom Settings Configuration

SettingsSourcesStreamingAudio FilesSchedulerSpeaker StatusIntercomSystem StatusLogsDefaultsUpdateReboot

AHE YA404 | IP Audio Client

– Intercom SIP Settings

Server Domain

pbx.zadarma.com

Port

5060

Transport & Security

UDP

Registration Timeout (s)

30

Registration Retry Interval (s)

30

Playback Buffer Latency (ms)

100

Capture Buffer Latency (ms)

100

Audio Input Level

50 %

Half duplex Mode

Auto

Half duplex Level

0,05

Half duplex Timeout (ms)

600

Stations Configuration

Station	Enable	Extension	Password	Auth ID	Destination	Description
1		6001	*****		6000	Station 1
2		6002	*****		6000	Station 2
3		6003	*****		6000	Station 3
4		6004	*****		6000	Station 4
5		6005	*****		6000	Station 5
6		6006	*****		6000	Station 6



Item	Description
Intercom SIP Settings	
Server Domain	The server url or IP Address that the IPAC device has to register with. It can be the server hostname or the IP Address Default: “empty”
Port	Allows to configure the SIP negotiation Port Default: 5060



Transport & Security	Allows to configure the transport and security protocols to be used to establish the connection and transport data: 1. UDP: Connection establishment (SIP) and media transport (RTP) between the server and the endpoint is transparent and relies on UDP User Datagram Protocol (port 5060) 2. TCP: Connection establishment (SIP) between the server and the endpoint is transparent and relies on TCP Transport Control Protocol (port 5060) 3. TLS: Connection establishment (SIP) between the SIP Server and the endpoint is encrypted, performed using Transport Layer Security. Media transport (RTP) not over TLS, thus un-encrypted (port 5061) 4. TLS + SRTP: Connection establishment (SIP) and media transport (RTP) between the server and the endpoint is encrypted using TLS and SRTP (Secure Real-time Transport Protocol) (port 5061) <i>Default: UDP</i>
Registration Timeout	Configure the interval used for the registration with the SIP Server, in seconds. Accepted range: 5 to 300 seconds. <i>Default: 30</i>
Registration retry interval	Configure the registration retry interval in case of registration failure, in seconds. Set to 0 to disable auto re-registration. The re-registration will be done at the retry timeout plus a random time (between 0 to 10 seconds to avoid all clients re-registration at the same time). <i>Default: 30</i>
Playback Buffer (ms)	Sets the desired target buffer for the SIP playback interface. An higher value increases the overall delay gaining on quality, a lower value decreases the delay but the quality of the play-out might be affected by drops or errors. <i>Default: 100</i>
Capture Buffer (ms)	Sets the desired target buffer for the SIP capture interface. An higher value increases the overall delay gaining on quality, a lower value decreases the delay but the quality of the play-out might be affected by drops or errors. <i>Default: 100</i>
Audio Input Level	Fine tunes the audio input level to increase or reduce the input level into the SIP call. <i>Default: 50%</i>
Half Duplex Mode	The Intercom Channel only operates in Half Duplex, that means it doesn't allow to transfer audio in both directions at the same time.



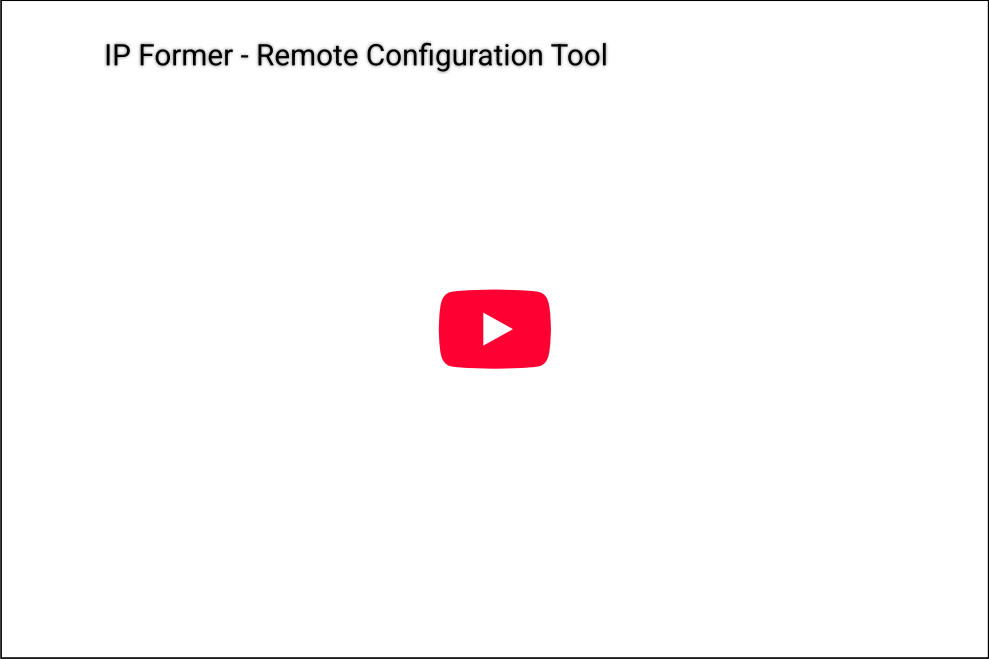
	Auto: The “direction” of the audio in the call is prioritized based on audio detection of the remote central. When no audio is detected, the direction is “station” → “central”, when audio is detected and goes above the specified “Level” (see below) the audio direction switches and is from “central” → “station” Manual: The direction of the audio is controlled by the digital input (button pressed) on the station. As far as the button is pressed the direction is “station” → “central” when button is released the audio is from “central” → “station”. in this configuration the call can only be ended by the remote central. <i>Default: “Auto”</i>
Half Duplex Level	This is a “threshold” value between 0 and 1 that helps to fine tune the level at which the audio detection on the remote central happens to switch the direction of the audio. When set to 1, the behavior is similar to a FDX communication. This is not used when Mode is set to “Manual”. <i>Default: 0.05</i>
Half Duplex Timeout	After remote central takes priority in the call (audio goes above the specified level) this is the timeout to be configured before the direction is switched back when audio goes below that same level. It accepts values from 10 to 1000 ms. <i>Default: 600ms</i>
Station Configurations - each station is registered in the SIP server with a dedicated extension	
Enable	Enable / Disable the corresponding station
Extension	The extension / username to be used for the corresponding station
Password	The password to be used for the corresponding extension
Auth ID	The Auth ID to be used for the corresponding extension
Destination	The extension to be dialed when the input is closed (button pressed)
Description	A brief description of the station

Remote Configuration Tool

A useful tool to manage multiple IPAC devices installed in the same network. The remote configuration tool is a server-less program. Every IPAC device runs a backend able to automatically find other IPAC devices on the network using the [Avahi service](#) and list them in this page. Avahi is built relying on [mDNS](#) and [DNS-SD](#) which takes advantage of multicast properties to advertise services on the network and enable connections between machines. IPAC devices advertise the services on the IP address: 224.0.0.251 (IPv6 ff02::fb) port 5353 (as per mDNS specifications). Below is a video introduction and explanation on the



configuration tool (it refers to IP Former but the video is valid for all other IPAC devices too), where it is possible to see it in action.



❏ The Remote Configuration Tool lists only IPAC devices running the same firmware version

The goal of the remote config tool is to facilitate and speed up the configuration of an entire system, particularly useful in case of IP Former installations, where each (IP Former) loudspeaker is an endpoint of the system and it might be easy to have tens of devices into the same installation. Thus it avoids the need to log into every single web interface to perform a configuration but access only one device to configure the entire system. On any IPAC device when opening the Remote Config tab will show:

SettingsSourcesAudio FilesSpeaker StatusSystem StatusLogsDefaultsUpdateRebootRemote ConfigM400_FLEXA MAC 00:08:E1:08:46:A9 FW 2.7.0

IP Audio Client | M400 #2

BARIX

Total devices discovered: 6

Filter

Reboot AllRead AllSave AllCancel changes

Name	Zone	Settings	Prio 1	Prio 2	Prio 3	Prio 4	Prio 5	Audio Files	System	Control	Copy / Push
Car Park	Car Park	192.168.3.30	Informacast	Synapps	IC Paging	RTP BGM	Barix Radio	0 Files	00:08:e1:07:5c:1a		CopyPush
Ceiling 1	Lobby	192.168.3.115	Informacast	Synapps	IC Paging	RTP BGM	Barix Radio	0 Files	00:08:e1:08:06:21		CopyPush
Ceiling 3	Lobby	192.168.3.22	Informacast	Synapps	IC Paging	RTP BGM	Barix Radio	9 Files	00:08:e1:08:11:2d		CopyPush
Ceiling 2	Lobby	192.168.3.20	Informacast	Synapps	IC Paging	RTP BGM	Barix Radio	0 Files	00:08:e1:08:54:47		CopyPush
Ceiling 4	Lobby	192.168.3.31	Informacast	Synapps	IC Paging	RTP BGM	Barix Radio	0 Files	00:08:e1:08:46:a9		CopyPush
Meeting room	Rooms	192.168.3.25	Informacast	Synapps	IC Paging	RTP BGM	Barix Radio	0 Files	00:08:e1:08:4d:31		CopyPush

Items per page: 101 - 6 of 6<>>

Remote Configuration Tool

The table is organized in columns, some of them are dynamic, meaning it is possible to open their content to change configurations (note the small arrow on the right-side of the header). Above the table, in the upper section of the screen there is an indication on the **Total number of devices discovered** and a search bar that can be used to **filter** the content of the table. By default the table visualized sorts the devices by their **Zone name**. A **green** marker around a source (the one in the RTP BGM in the screen above) indicates that the source is (or was) playing the last time the Remote Config Tool table was refreshed, the marker is not updated in real-time but after a table reload (or clicking on the READ ALL button).

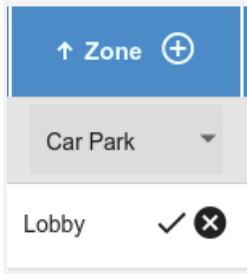


Function	Description
Name	This is the name used in the web interface and in the configuration tool to identify a device. By default is set to “Empty”. To change it click on the pencil tool that appears in proximity of the name cell when overlaid with the mouse. Once modified SAVE. Change table sorting: in the Name column header there is an arrow, click on it to change the sorting using the device’s name from A -> Z to Z -> A.
Zone	It is possible to assign devices to Zones. This makes it easy to apply the same configuration only to devices that are part of the same Zone. By default this field is set to “Zone 0”, all new discovered devices are part of this default zone and can be controlled in bulk immediately. Add a new Zone: click on the plus-circled button on the header of the zone column. This will open a dialogue window where you can check the list of the already existing zones and add a new one simply typing the name of the new zone to be added and click on Add Create New Zone Type the name of the new zone (case sensitive): New zone name Add Warning: If you don't assign any device to the new zone after closing this popup, the new zone will disappear after reloading the Remote Config webpage. Zone List Zone 0 Lobby Rooms Car Park Create a new Zone
	Rename a Zone: simple overlay with the pointer the zone name of any device part of the Zone and click on the pencil tool that appears then confirm the change by pressing the “V” confirmation button, the new name will be applied automatically to all devices belonging to that zone.

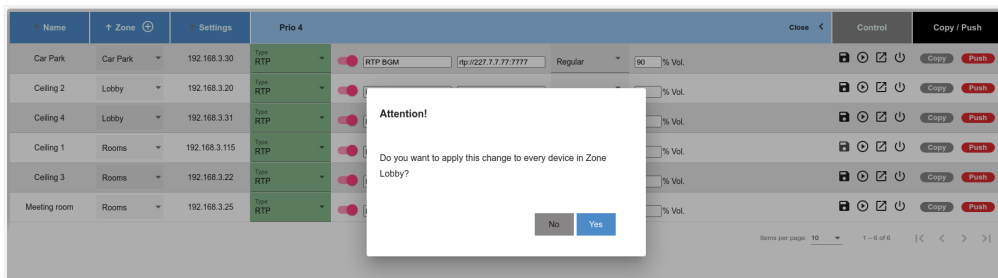


Function

Description



Assign devices to Zone: select the zone from the drop-down menu in correspondence of the device required, after selecting the new zone a dialogue window asks if the change has to be applied only to that device or all devices part of the same Zone. Selecting NO will only assign that device. Selecting YES will change all devices.



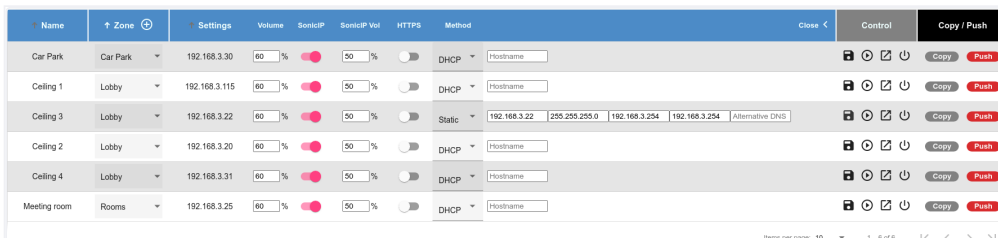
Assign device to a new Zone

Remove a Zone: there is no such function in the Remote Config Tool. A zone is removed automatically when no devices are part of it. If it's still available in the drop down is because the web page needs to be refreshed, the zone containing no devices will disappear from the list right after.

Change table sorting: in the Zone column header there is an arrow, click on it to change the sorting from A -> Z to Z -> A.

Settings

This is a dynamic column. Click on the arrow in proximity of the Settings header to open the device's settings configuration

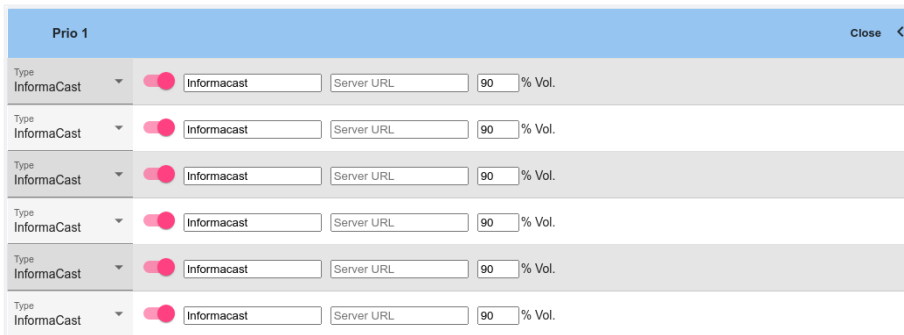


Remote Config Tool - Device Settings

For each device it is possible to (from left to right):

- Visualize the IP Address currently assigned
- Change the Master Volume in percentage: it can be applied in bulk or individually



Function 	Description 
	• Enable / disable Sonic IP and set its volume: it can be applied in bulk or individually • HTTPS It enables the HTTPS protocol for the web interface access of each device. It can be applied individually or in bulk • IP Addressing method: DHCP or Static. It can be applied individually only. When using STATIC the device opens the manual IP configuration options (IP, Subnet, Gateway, DNS 1, DNS 2). When using DHCP the DHCP hostname can be changed. These are the same Settings available in the Settings page of each IPAC device. For more help topics on each field refer to the help section of the Settings page. When settings are changed it is important to SAVE them with the corresponding buttons in the device control section or using the SAVE ALL button available in the top right corner of the page.
Priorities	These are 5 dynamic columns reflecting the priority layers also available in the source tab. Click on the arrow in proximity of the header to expand the desired priority layer and configure the source, the parameters and values available are the same as the ones available for each source described in previous chapters. below is an example of InformaCast source expanded:  RCT - Example of Priority 1 with InformaCast Every priority layer will allow the user to: • Select the Source Type • Enable / Disable the source (same as the ON /OFF available in the Source page) • Assign a name to the layer, displayed in the table when the columns are closed • Offer the source's parameters, which change depending on the source selected. (all described in details in previous chapters) • Configure the volume of the layer in percentage
Audio files	This column simply visualizes the number of audio files uploaded and stored. Visit the Audio Files section (or Local File source chapter) for more information on how to handle audio files playback
System	This is an expandable column that opens up the details of the IPAC devices: MAC Addresses, HW  Type, FW Version installed, System Time, Up Tlme (since how long the device is powered), License (if

Function**Description**

the device contain any valid license, i.e. Informacast).

System ↑	HW Type	FW Version	System Time	Up Time	Close <
00:08:e1:07:6c:1a	M400	2.7.0	Sep 21 2021 - 01:41:15 PM	0h 18m	Licenses: InformaCast
00:08:e1:08:06:21	TPA400	2.7.0	Sep 21 2021 - 01:41:15 PM	0h 32m	Licenses: InformaCast
00:08:e1:08:54:47	TPA400	2.7.0	Sep 21 2021 - 01:41:15 PM	0h 33m	Licenses: InformaCast
00:08:e1:08:46:a9	M400	2.7.0	Sep 21 2021 - 01:41:15 PM	5h 0m	Licenses: InformaCast
00:08:e1:08:11:2d	TPA400	2.7.0	Sep 21 2021 - 01:41:18 PM	0h 19m	Licenses: InformaCast
00:08:e1:08:4d:31	MPA400	2.7.0	Sep 21 2021 - 01:41:15 PM	0h 24m	Licenses: InformaCast

Remote Config Tool - System

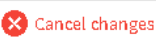
Control

From left to right:

- **SAVE:** saves any modification applied to the single device. This affects the individual device only - When any modification is performed this button blinks, indicating that changes for that device must be SAVED to be applied
- **TEST:** plays immediately a 4 seconds 1KHz sine wave. This allows a device to be quickly tested and identified in an installation
- **OPEN WEB UI:** opens in a new page the Web Interface of the device
- **REBOOT:** reboots the single device



Function	Description
Copy/insert	Copy / Push Cancel Push Insert Push Insert Push Insert Push Insert Push COPY / INSERT: copies the whole configuration of a device to other devices. When clicking on copy on one device, all copy buttons of the other devices will transform in an INSERT button. Click on as many INSERT as needed to apply the same settings of the COPIED device on the desired devices. The Copy function will allow you to create custom installations where devices part of the same zone might have different configurations. <u>Copy / Insert will only affect Sources configuration and not the settings.</u> The Copy / Insert function requires to click on SAVE ALL in the top right corner of the page to apply the new configuration to the affected device(s)
Push	Attention! Do you want to PUSH the configurations from device Car Park (00:08:e1:07:6c:1a) to ALL devices or only for devices in ZONE Car Park? Warning: This operation will automatically save the new configuration into the devices. Cancel All Devices Zone Devices Push Configuration to Zone or All This function pushes the configuration of one device to all other OR to devices part of the same zone. A dialogue shows up after clicking the button allowing the user to decide. The “push” function is what actually speeds up large system configurations. After setting up one device it is possible to copy its configuration to all others in 2 clicks. ⓘ Unlike the Copy / Insert function the configuration that will be replicated includes also the settings: Master Volume, Sonic IP, HTTPS enable/disable plus all priority layer ≡

Function ↕	Description ↕
	configurations. On top of this the new configuration is automatically saved and applied to the devices
Bulk Actions	   
Reboot all	Reboots all the devices discovered at once. It opens a dialogue asking for confirmation before proceeding.
Read all	The Read All button is used to parse the configuration of all IPAC devices in the network and refresh the table. If there are some pending changes not yet saved the user is warned as they must be either canceled or applied (saved) before refresh the table. ■ When a new source configuration is applied, IPAC might take some seconds while restarting the services required to advertise itself and be identified by the remote config tool, in those occasions if the number of devices discovered is not the one expected, click on Read All so that the number is updated.
Save all	Saves any modification made in ALL the devices discovered. If no modifications are done, there is no operation performed and the user is informed.
Cancel changes	Cancels any modification performed in any device in the list and restores the initial condition

Speaker Status

The Speaker Status page is the main dashboard of the IP Audio Client application. The window is quite essential , it displays the current playback status with the source arranged from top to bottom reflecting the priority layers. A green marker colors the currently playing source.

Settings
Sources
Audio Files
Speaker Status
System Status
Logs
Defaults
Update
Reboot
Remote Config

IP Audio Client | M400 #2

Speaker Status

Informacast | InformaCast
Registered
n/a
Playing
nothing
+ Advanced info
Play count
0

Synapps | Syn-Apps
Registered
No
Playing
nothing
Play count
0

IC Paging | IC-Paging
Play count
0

RTP BGM | rtp
Play count
1

Barix Radio | HTTP
Play count
1

IP Audio Client - Speaker Status

Each source contains the information related to the source itself and they change according to the status of the single source. A thorough description on how the speaker status changes for each source is given in the source chapter of this guide.

The Play count is a counter that shows the number of times the corresponding source played. The counters are reset every time a change in the source configuration is applied or after a reboot.

Dante® Status

 Only present on Dante® Licensed devices

The Dante® status tab is where IPAC shows some important details on Dante®:

Settings
Sources
Streaming
Audio Files
Scheduler
Speaker Status
Dante Status

TPA400_Rack | IP Audio Client

Dante Status

Dante Device

TPA400

Servo State

SYNC

GMID

0x001dc1518ae2

Drift

23881

Dante® Status



Dante® Device	The name of the device assigned in the Dante® controller
Servo State	Indicates the clock status: • NONE - this device is not following any PTP leader (normally it appears when this device is the leader). • SYNC - the clock is in sync with the leader.
GMID	The ID of the device providing the clock in the system, the PTP leader.
Drift	The drift between the leader and this device's clock (if this device is the leader the drift is 0) measured in microseconds

The Dante® Status tab appears only on Dante® licensed devices

System Status



– Device information Hardware type IPFormer_TPA400 (ID 2019_8126) IPAM type IPAM400 (ID 8) MAC address 00:08:E1:08:06:21 Linux kernel version Linux version 5.5.8-yocto-standard (oe-user@oe-host) (gcc version 9.3.0 (GCC)) #1 SMP PREEMPT Tue Jul 27 16:03:07 UTC 2021 Bootloader version U-Boot 2017.03-g8537ddd (Sep 13 2018 - 19:47:54 +0200) Allwinner Technology	
Hardware type	The actual IPAC device hardware (in parenthesis is the Part Number)
IPAM type	The Barix IPAM codec module installed in the device (in parenthesis the ID of this component)
MAC Address	Media Access Control address - This is the hardware address of the Ethernet interface of the device
Linux kernel version	The version of the Linux operating system in use
Bootloader version	The bootloader version currently in use



– Network

Wired Connection

Status	Connected
IP address	192.168.3.31
Netmask	255.255.255.0
Default gateway	192.168.3.254
DNS servers	Auto

Status	Shows if the device is connected via the Ethernet interface to the network
IP Address	The current IP address of the Ethernet interface
Netmask	The subnet mask on which the device is operating
Default Gateway	The gateway address currently set
DNS Servers	The DNS server addresses (primary and secondary) currently in use. Provided either by the DHCP or entered manually in the settings page

– System

System time (refresh)	Tue Sep 21 13:47:09 UTC 2021
Uptime	0 days, 7 hours, 6 minutes, 50 seconds

System time	The device's current local time
Uptime	How long the device is powered on

– Licenses

Name	core-image-barix-ipspeaker
Status	active
Issue date	2021-09-14 00:00:00 +02:00
Expire date	2051-09-14 00:00:00 +02:00
ID	5466
Features	InformaCast
Signature	Valid

Name	The name of the base license for the IPAC device
Status	Either if the license is active or expired
Issue date	The date when the license was issued
Expire date	The date when the license will expire



ID	The license unique identifier
Features	If this license has any additional features (e.g. InformaCast)
Signature	Either if the license that is running on the image is valid or invalid (in such case the application wont run)

– Mass storage devices						
Type	Usage	Size	Used	Avail.	F. Sys	Status
USB		7.5G	547M	7.0G	vfat	read-write

Mass Storage	Shows up only if there is any USB external memory connected to the IPAC device providing the details of the same
--------------	--

– Amplifier	
Temperature	47.56 °C

**Only available on IPAC devices with temperature sensor implemented in the amplifier (e.g. IP Former TPA400)*

Amplifier temperature	Shows the current temperature of the amplifier. It updates at every page reload.
-----------------------	--

– Power Source & Speaker	
Power Source	PoE+
Speaker Impedance	8.86 Ω
Amp Gain	21V peak output voltage

**Only available on ceratain IPAC devices with automatic gain amplifier adjustment depending on the power available and impedance measurement feature (e.g. Exstreamer MPA400)*

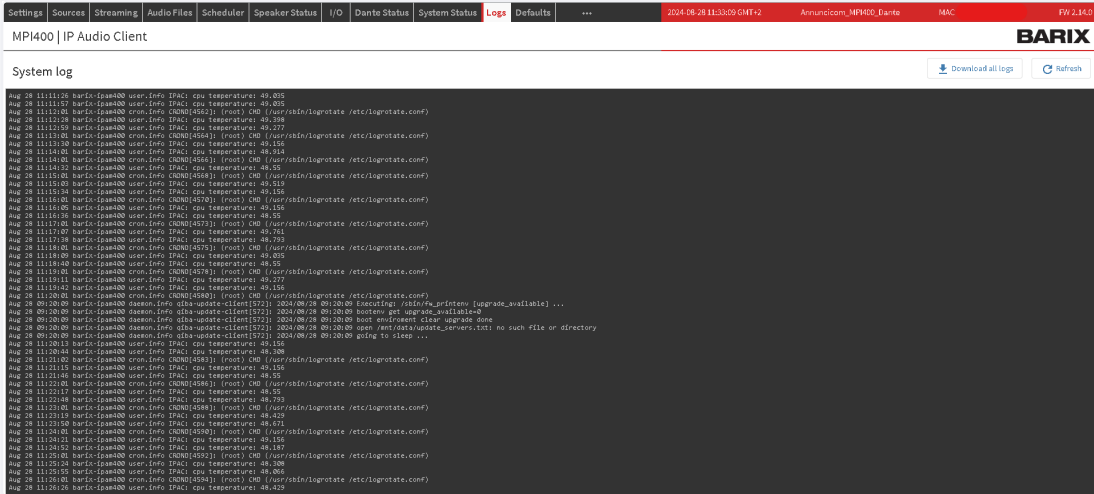
Power Source	The power source detected to power the IPAC device (PoE, PoE+, PoE++, 4PPoE or External Power Supply)
Speaker Impedance	The speaker load detected connected to the IPAC device’s terminals
Amp Gain	The amplifier output voltage (Vpp) capability

Logs

The Logs page displays the events occurring operation of the IPAC device. Logs are useful for debugging issues, they give a detailed insight into every process within the OS. On the top-



right corner, it is possible to click the button “[Download all messages](#)” to download all log files.



- ✓ When contacting the Barix support it is useful to attach along with the communication the Log files downloaded and the status page screenshot. Do not forget to give a description as much detailed as possible about the steps made that led to the issue.

Packet Capture Procedure

Packet capturing is a powerful diagnostic tool that allows you to monitor and analyze network traffic. This feature is useful for understanding the data flow between hosts on your network. Follow these steps to initiate and retrieve a packet capture:

1. Initiate Capture

- In the address bar of your web browser, append the following to the current WebUI URL:

CODE

```
/web/rest/v1/tools/start-interface-capture/eth0
```

- Press Enter to execute the command.

2. Confirm Capture Initiation

- Upon successful initiation, you will see a confirmation message:
"Capture started for interface [interface number] for 5 minutes or 5MB. Output will be available in [specified time]."

3. Retrieve Capture Data



- After the specified time has elapsed, return to the main WebUI.
- Navigate to the LOGS tab.
- Download the complete log package clicking on the corresponding button
- Locate the "captures" folder within the downloaded package to access your packet capture data (.pcap file, which can be opened with a software like Wireshark for detailed analysis)

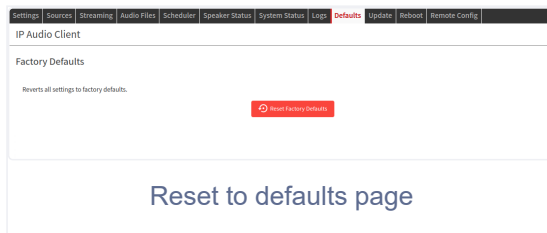
 The packet capture will automatically terminate after 5 minutes or when it reaches 5MB of data, whichever occurs first.

Defaults Page

Reset to factory defaults - Soft reset

The defaults page allows performing a soft-reset (it doesn't change the device's application mode). Such operation reverts all settings except the network settings back to factory defaults (it doesn't perform a firmware downgrade, just a reset of the settings for the running application). The reset to factory defaults can be enabled or disabled from the settings page, under "Security Settings".

 When resetting to defaults the application mode is not changed. I.e. if the unit is running in AES67 Mode it will keep running in that mode after a reset.



Reset to factory defaults – Hard reset

It is also possible to perform a hard reset. A hard reset will revert all settings to factory defaults including the network settings.

To perform a hard reset locate the reset button on the front panel of the IPAC device and keep it pressed for approximately 10 seconds using a sharp-pointed tool while the device is powered. The status led will blink green 3 times, informing that the reset operation is started.

After a few seconds, the device will reboot with factory defaults. Reload the page after the operation is completed.

- When resetting to defaults the application mode is not changed. I.e. if the unit is running in AES67 Mode it will keep running in that mode after a reset.

Rescue Image Process

In some occasions it is possible that the application faces some issues that cannot be restored with a hard reset to default. In this situation a rescue image process might solve:

- If your IPAC device is powered, shut it off
- Make sure the device is connected to internet. For this process to work the device will attempt a connection to Barix Update Server (release.barixupdate.com), via HTTP (port 80), if successful it will download the rescue image from this server
- While the device is off, press the reset button
- Keep the button pressed while powering up the device
- Keep the button pressed in for 8 seconds after the device is powered, then release
- When the rescue process starts the led blinks orange
- The device will restart indicating the FW install was successful

After this process the device will run the firmware image fetched from the download server.

Update

IP Audio Client offers the possibility to update the firmware of the device from this page and to program the automatic firmware update.

- ✓ Run always the latest firmware available from Barix as it might include bug fixes, improvements or new useful features.

By default the Automatic FW update is *disabled*.

In the same page it is possible to visualize the current firmware version running and to manually update it.

Manual Firmware Update



1	Download the latest firmware from here - The firmware is a <i>.tar</i> archive.
2	In the <i>Update Tab</i> Click on “ Choose File ”, locate and select the <i>.tar</i> archive just downloaded
3	Click on “ Upload ”, a progress bar will show up. Wait until the process is finished - while the update is in progress DO NOT leave the Update Tab or power off the device
4	Once the upload is complete the device will reboot . It will take approx. 1 min. Wait until the web page is reloaded.

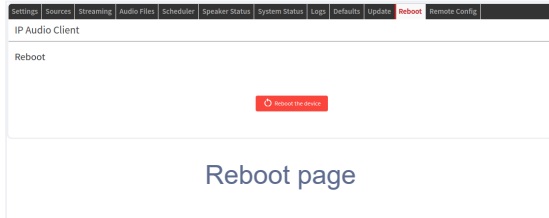
Reboot

The reboot is the process of power cycling the device (ON → OFF → ON). It's a complete power cycle. The reboot can be performed from the web user interface or from the reset button located on the front panel of the device.

Reboot - web interface

Click "Reboot the device" to restart the device.





Reboot - physical button

The reboot of the device from the button is performed by pressing the Reset button located on the front panel of the IPAC device for ~3 seconds.

 A long press of this button (10 seconds) will reset the device to factory defaults.

AES67 Application Mode

This section describes the relevant functions present when the device is set to run in AES67 Mode.

When the device is set to run in this mode it doesn't use anything about the multi-source arrangement with priority handling, the unit becomes solely capable of decoding AES67 received streams.

 The end to end latency in this mode measured using an AES67 encoder set to encode with a latency of 4ms, is 10ms (from the encoder analog input to the IPAC decoder analog output)

AES67 Quick Start

To get your AES67 stream played in seconds follow these instructions:



1	Make all the necessary connections: audio, network and power up the unit
2	During boot the device will announce its IP Address (Sonic IP function), make sure to hear the audio on the output and note the IP address on a paper. An IPAC device takes approximately 40s to finish the boot process
3	Connect to the IP Address of the device via web browser by typing in the URL the IP address just heard
4	Enter username and password (<i>username: admin - password: printed on the backside of the device</i>)



5	By default configuration, IPAC boots into Standard Mode. Change the Application mode selector from the settings tab
6	Click SUBMIT - The device reboots. The whole cycle takes ~50s. When the device is ready again will automatically reload the Login page (if the automatic reload fails enter the IP Address of the device manually in the browser URL) enter the credentials entered before.
7	Locate the AES67 tab. If a source is discovered (either via mDNS-RTSP or SAP) the stream(s) found will be available in the Sources list. Click on PLAY to play the corresponding stream. The music starts immediately.

AES67 Configuration

The AES67 page is where the user has access to the AES67 related settings: Discovery and Sources.

Settings
AES67
PTP - RTP
System Status
Logs
Defaults
Update
Reboot

MPI400 | AES67

Discovery Configuration

mDNS discovery
RTSP Port
SAP multicast address

Sources

Protocol	Name	Domain	Address	RTP Address	RTP Port	Last Activity	Period	SDP
SAP	DanteAVIO : 2	N/A	192.168.11.229	239.69.191.50	5004	Just now	30	Show more

For more information on AES67 follow this [practical guide](#), well done by Ravenna and published on [The Broadcast Bridge](#).

AES67 Discovery

[AES67](#) standard doesn't specify any discovery mechanism. Discovery is the process of identification of AES67 streams in a network which also determines how the SDP ([Session Description Protocol](#)) file is exchanged in a system. The SDP file contains all relevant information about the RTP stream and can be used by the receiving device to setup its internal processes and play the stream accordingly.

Name	The name of the stream identified - For SAP / mDNS sources this name is the one found in the SDP file																		
Domain	This indicates which PTP domain is used to synch with this stream. AES67 PTP domain typically is always 0 when not indicated otherwise. This information is contained in the SDP file																		
Address	The IP Address of the TX device originating the stream. This information is contained in the SDP file																		
RTP address	The destination RTP Address used by the TX and from where the stream is retrieved. This information is contained in the SDP file																		
RTP port	The destination RTP port number used by the TX and from where the stream is retrieved. This information is contained in the SDP file																		
Last activity	Shows the last activity of this stream. When a stream is available this is “Just Now”. When a stream is inactive it shows the time since the stream became inactive NOTE: an inactive stream cannot be played and it can be removed from the list																		
Period	It is related to the SAP interval, how often the TX sends the SDP updates to the appointed multicast address. This information is automatically retrieved by SAP itself and advertised by the TX device																		
SDP	Allows to show the full SDP content provided by the TX device or if a source is manually added it shows the content of the SDP file provided <table><tr><th>Protocol</th><th>Name</th><th>Domain</th><th>Address</th><th>RTP Address</th><th>RTP Port</th><th>Last Activity</th><th>Period</th><th>SDP</th></tr><tr><td>SAP</td><td>DanteAVIO : 2</td><td>N/A</td><td>192.168.11.229</td><td>239.69.191.50</td><td>5004</td><td>Just now</td><td>30</td><td>Show less</td></tr></table> SDP Content <pre>v=0 o=- 808922 808922 IN IP4 192.168.11.229 s=DanteAVIO : 2 i=2 channels: CH1 CH2 c=IN IP4 239.69.191.50/32 t=0 0 a=keywds:Dante a=recvonly m=audio 5004 RTP/AVP 97 a=rtptime:1 a=ts-refclk:ptp=IEEE1588-2008:00-1D-C1-FF-FE-51-8A-E2:0 a=mediaclk:direct=0</pre> SDP file content For all information about SDP Content click here	Protocol	Name	Domain	Address	RTP Address	RTP Port	Last Activity	Period	SDP	SAP	DanteAVIO : 2	N/A	192.168.11.229	239.69.191.50	5004	Just now	30	Show less
Protocol	Name	Domain	Address	RTP Address	RTP Port	Last Activity	Period	SDP											
SAP	DanteAVIO : 2	N/A	192.168.11.229	239.69.191.50	5004	Just now	30	Show less											



Play/stop/remove 	PLAY: When a stream is found or manually added it can be played by clicking on the corresponding icon. The stream will play immediately. In case multiple streams are available only one at a time can be played. It plays always the last stream for which the PLAY button was pressed. STOP: When a stream is playing, it is possible to stop its playback at any time REMOVE: This button is available only when a stream is inactive. It allows to remove a stream from the list, but be aware that if the stream is advertised again by the TX via SAP or mDNS it will show up again
Add a stream manually	Click on the “+” icon on the top right corner of the list to add a stream manually providing the SDP content of the same - Some AES67 compatible devices don't implement any discovery protocol but they provide an SDP file to be copied and pasted in the receiving device New AES67 Source Name Source name SDP Cancel Submit Add a new source manually

AES67 Supported Audio Format

The IPAC Implementation of AES67 implements the following audio format: L24/48000/2
(Linear PCM 24 bits / 48kHz sampling frequency / Stereo)

PTP - RTP

This page is an informative page with very important information on the PTP status and RTP statistics.

PTP or [Precision Time Protocol](#) is fundamental in an AES67 system as it is the backbone for the clock synchronization among different devices in the system. IP Audio Client can only act as a “follower” (or “slave”, term now deprecated in the industry), meaning that a Grandmaster clock device has to be present in the network to provide clock to IPAC nodes.



PTP Configuration

PTP Domain

– PTP Status

Mode *Slave*
Status *locked*
GMID *00-1D-C1-FF-FE-51-8A-E2*

PTP Domain

Set the PTP domain to be used. Domains allow multiple clock distribution systems to share the same communications medium. The Default domain for AES67 is 0, which enables devices to communicate via multicast address 224.0.1.129, where SYNC, ANNOUNCE, FOLLOW UP, DELAY_REQ, DELAY_RESP messages are sent by the devices in the system. Changing the domain changes the multicast address used.

Default: 0

Mode

This field can only be “slave” or “follower” as the current PTP implementation in IPAC devices cannot act as “master” or “leader”

Status

“locked” when the device is locked to the clock of the master device
“unlocked” when the device is not locked to any master clock
“locking” when the device found a master clock and it is synchronizing

GMID

Grand Master ID. The unique identifier of the device that has been elected as master clock provider

– RTP Sink Status and Errors

Source *AVIOAI2-518ae2:2*
Muted *No*
Min Packet Arrival time *0*
Receiving RTP packet *Yes*
RTP sequence errors *4*
RTP stream source errors *0*
RTP payload type errors *0*
RTP packet timestamp errors *4*

Source

The name of the source sending RTP packets

Muted

Yes: when audio is not flowing to the output



	No: when audio is flowing to the output
Min Packet Arrival Time	Not implemented
Receiving RTP packets	Yes: when RTP packets are received by the tx No: when RTP packets are not received by the tx
RTP Sequence errors	Number of RTP sequence errors found at stream reception
RTP Stream source errors	Number of errors generated by the stream source
RTP Payload type errors	Number of errors due to a bad RTP payload type
RTP Packet timestamp errors	Number of packets with timestamp errors (normally increases along with the RTP sequence errors)

Dante Application mode

 Only available in Dante Licensed IPAC devices

This section describes the relevant functions present when the device is set to run in Dante Exclusive Mode.

In this mode the IPAC device doesn't offer the multi-source arrangement with priority handling, the unit becomes solely capable of decoding 2 channels Dante streams.

Why Dante Exclusive Mode?

The end-to-end latency of a Dante source in IPAC Standard Mode takes into account the additional delay introduced by the multi-layered architecture and is approximately 190 ms. For customers deploying devices in Dante-only systems where no other sources are needed, it is possible to use this mode that achieves an end-to-end latency of 8 ms (measured with a Dante AVIO 2x0 set to encode at 1 ms delay).

There is no Dante related control available in Dante Exclusive Mode. All the controls and configuration is performed in the Dante Controller.

From the IPAC web configuration interface it is still possible to access the SETTINGS, DANTE STATUS, LOGS, UPDATE, REBOOT and UPDATE tabs, well described throughout the previous chapters of this manual.



Dante Exclusive Mode - Quick Start

When using the IPAC device in Dante Exclusive Mode follow these steps to get your unit up and playing a Dante stream in seconds.



1	Connect the IPAC device and power up. The unit takes between 40 to 50s to boot.
2	Open Dante Controller from a PC in the same network of the IPAC device.
3	In the routing table of Dante Controller make the necessary connections to play audio from a Dante transmitter
4	In case a multicast stream is required (when routing audio to 2 or more IPAC devices), enable multicast flow in Dante controller for the transmitter device used.

Compliance and further Information

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the device into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Safety and precaution recommendations apply. Find them in the download section at www.barix.com



[Find your distributor on this list](#) for more hardware.

For questions that are extending the documentation, feel free to contact us on:



International: +41 434 33 22 22

USA: +1 866 815 0866

Barix.com / Barixmail@barix.com

Copyright © 2025 Barix. All information and the use of this product including all services are covered under the [Barix](#)

[Terms & Conditions](#) and our [Privacy Policy](#). Please follow the [Safety and Precaution](#)

[Recommendations](#). Barix is a ISO 9001:2015 certified company. All rights reserved. All

information is subject to change without notice. All mentioned trademarks belong to their

respective owners and are used for reference only.

