How to use 2N SIP Mic with Axis network speakers Use cases and step-by-step guides

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1. Introduction

The first 2N product to be distributed by Axis is 2N SIP Mic, a two-way network microphone console with a built-in audio management server. It can be easily integrated with Axis network speakers to form a complete public-address system, possible to purchase from a single distributor. Based on open standards, 2N SIP Mic can also be connected to other IT systems.

This document describes suggested use cases for 2N SIP Mic with Axis network speaker system, and provides step-by-step guides for how to configure the hardware. Note that only use cases that support Axis equipment are described, and for details about other possibilities using 2N SIP Mic, an extended user manual can be found on 2N website:

https://wiki.2n.cz/sipmic/latest/en/3-funkce-a-uziti/3-2-priklady-nastaveni/3-2-4-http-pozadavek

2. Hardware



The default login credentials for 2N SIP Mic are "Admin" (user name) and "2n" (password).

2N SIP Mic comes with DHCP enabled as default setting. To use the default IP address, 192.168.1.100, press the call button 15 times after restart. For discovery, use the 2N[®] Helios IP network scanner found at www.2n.cz/products/2n-network-scanner

This document is intended for setups where 2N SIP Mic is connected to Axis audio products such as network speakers AXIS C1004-E, AXIS C2005, and AXIS C3003-E, or network audio bridge AXIS C8033. The audio system is assumed to be already in place.



3. Trigger audio from 2N SIP Mic

3.1 Use transmit.cgi and send HTTP request

2N SIP Mic has support for HTTP trigger with password authentication and can send general VAPIX commands to all Axis products. However, it cannot receive any VAPIX commands.

When using 2N SIP Mic in combination with Axis network audio systems, it is possible to address zones for public announcements by using transmit.cgi. A maximum of 12 IP addresses can be assigned per zone/button at the same time. Zones can easily be changed, removed, or added in an instant, over the network.



3.2 Use SIP calls and DTMF triggers

2N SIP Mic has support for SIP and DTMF (dual-tone multiple-frequency) signals. SIP calls can be initiated by 2N SIP Mic either via a VoIP PBX, or, if the addressed device is in the same network, as a SIP peer-to-peer connection. To address many speaker zones, the buttons of 2N SIP Mic may be insufficient. It is then possible to instead use paging groups defined in the SIP PBX, in order to address the zones.



4. Suggested use cases

A brief description of the suggested use cases is provided in the following subsections, and detailed step-by-step guides are provided in section 5.

In use cases A, B, and C, audio is triggered by using transmit.cgi or sending a http request. In use cases D, E, and F, audio is triggered using SIP calls and DTMF.

4.1 A – Use transmit.cgi for public announcements to one speaker group

A basic installation for this use case could contain 2N SIP Mic in combination with an Axis network speaker group consisting of products such as AXIS C3003-E, AXIS C1004-E, AXIS C2005 or AXIS C8033.

An operator would address this whole group for a general announcement.

4.2 B – Use transmit.cgi for public announcements to several speaker groups

An installation for this use case could contain 2N SIP Mic in combination with an Axis network speaker group consisting of C1004-E and/or AXIS C2005-E for background music. Two other speaker groups, consisting of multiple AXIS C3003-E speakers, could be installed in a parking area and in a playground.

An operator could address the first speaker group (playing background music) for a general announcement, but later on, also the other speaker groups for another announcement to all three zones.

4.3 C – Use HTTP request to trigger audio clip stored on leader speaker of a group

A basic installation for this use case could contain 2N SIP Mic in combination with an Axis network speaker group consisting of products such as AXIS C3003-E, AXIS C1004-E, AXIS C2005, or AXIS C8033.

An operator would trigger an audio clip stored on a speaker, by pressing a button on 2N SIP Mic. Clip example:

http://ip address of the speaker/axis-cgi/playclip.cgi?location=ding_dong.mp3&trepeat=-1&tvolume=100

4.4 D – Use SIP for public announcements to one speaker

A basic installation for this use case could contain 2N SIP Mic in combination with an Axis network speaker group consisting of products such as AXIS C3003-E, AXIS C1004-E, AXIS C2005, or AXIS C8033.

An operator would address an Axis SIP-capable device, in this case AXIS C1004-E, for a general announcement, by initiating a peer-to-peer call from 2N SIP Mic.

4.5 E – Use SIP for public announcements to a speaker group

A basic installation for this use case could contain 2N SIP Mic in combination with an Axis network speaker group consisting of products such as AXIS C3003-E, AXIS C1004-E, AXIS C2005, or AXIS C8033.

An operator would address the whole speaker group for a general announcement, by initiating a SIP call from 2N SIP Mic.

4.6 F – Use DTMF for public announcement to a speaker group

A basic installation for this use case could contain 2N SIP Mic in combination with an Axis network speaker group consisting of products such as AXIS C3003-E, AXIS C1004-E, AXIS C2005, or AXIS C8033.

An operator would address a speaker group for playing a pre-recorded message triggered by a DTMF tone.

5. Step-by-step guides for the use cases

5.1 A – Use transmit.cgi for public announcements to one speaker group Step 1: Configure a zone with a VAPIX destination.



Step 2: Assign the zone to a 2N SIP Mic button.



5.2 B – Use transmit.cgi for public announcements to several speaker groups

Step 1: Configure a zone with multiple VAPIX destinations, to address several IP addresses in parallel.



Step 2: Assign the zone to a 2N SIP Mic button.



5.3 C – Use HTTP request to trigger audio clip stored on leader speaker of a group

Step 1: Prepare an audio clip on the leader speaker of the speaker group.

Basic Setup	Audio Clip List	0
* Audio	Audio Clip List	
Overview System Settings Device Settings Clips	Name Location Abreadabrs (enclaudioclos/kbrscadabrs.mp3) Alarm (enclaudioclos/kbrscadabrs.mp3) Birds chirsing (enclaudioclos/kbrscadabrs.mp3) Birds chirsing (enclaudioclos/kbrscadabrs.mp3) Birds chirsing (enclaudioclos/brscadabrs.mp3) Birds chirsing (enclaudioclos/brscadabrs.mp3)	4
+ VoIP	Camera cidea /etc/audiocipa/camera_cicles.mp3 Drog-dwg /etc/audiocips/dong.mp3	1. Choose a clip to play
Detectors	Internal message example //etclaudiocipalinternal_message_axiample.mp3 Logo //etclaudiocipaliogo.mp3	
Applications	Ring tone 1 /etc/audiodipal/ingtone_1.mp3 Ring tone 2 /etc/audiodipal/ingtone_2.mp3	*
+ Events	Add Play Stop Download Modify Remove	
Languages		
System Options		
About		
	Configure Audio Clip Link Audio clip dingdong.mp3 2. Ad Valume 100 100 100 100 100 100 100 100 100 10	djust the clip
	Configured link: 3. Use this http string in the 2N SI	P Mic config

Step 2: Configure a 2N SIP Mic button to trigger the audio clip on the speaker.



5.4 D – Use SIP for public announcements to one speaker

Step 1: Enable SIP on the speaker so that it can receive SIP calls.

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Advanced SIP Settings	_		15	
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Registration Interval (seconds):	300			
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Step 2: Configure and enable the right VoIP settings for peer-to-peer calls on 2N SIP Mic.

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Step 3: Configure 2N SIP Mic to initiate a peer-to-peer call.

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1 Activate the zone	
2. Name the zone	10.00
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1 M M	
3. Destination type must b	e SIP
4. Direct call for peer2peer	2
E Destination address*	
must be a valid SIP URI	
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and the second sample	
1 × 1	· · · · ·
Art see	
4711a	
	6 Save cha

Note that the SIP destination address must be a valid SIP URL that includes a host part before the @ and the network IP address after the @. In this example the host part can be anything between 0 and 9.

Step 4: Assign the zone to a 2N SIP Mic button.



When you press the configured button, 2N SIP Mic will initiate a peer-to-peer SIP call to the defined SIP destination (the speaker).

5.5 E – Use SIP for public announcements to a speaker group

Step 1: Enable SIP on the leader speaker so that it can receive SIP calls.

Basic Setup	SIP Settings			0
Audio	SIP Setup Assistant			
Overview	Start the satup assistant for a	easy SIP configuration.		Startum
SIP Settings	SIP Settings			
Account Settings	🕅 Enable SIP 🛑 1	I. Enable SIP)	
a trit settings	Calling timeout (seconds):	60		
Detectors	Incoming SIP Calls			
Applications	Allow incoming 51F calls	🕳 2. All	ow incoming	calls
Execute	Port Settings			
LYCINS	S3P porti	5060		
Languages	BDP TLS ports	3061		
System Ontions	RTP start ports	4000		
system options	NAT Traversal			
About	Enable IPv6 here			
	Enable ICE			
	Enable STUN			
	Enable TURN			
	Audio Codec Settings			
	Available codecs Xx4/1950 (1950 Hz) speet/18000 (1800 Hz) speet/18000 (1800 Hz) 0.726-32 (8000 Hz)	-	Selected codecs PCMJ (3000 Hz) PCMA (8000 Hz) spus (48600 Hz) L16/16000 (16000)	
	Advanced 5IP Settings		1	
	Disable Automatic UDP to	TCP Switch		
	Max call duration (seconds):	3600		
	Registration Interval (seconds	i): 300		
	Audio direction:	Send and receiv		

Step 2: Configure and enable the right VoIP settings to register 2N SIP Mic to a VoIP PBX.

	to your	ter Programmert. X	
	Settings	1. SIP user ID	
B Spinnes 19 Jacob G State	0	Therapport protocols	
	2300000	2. SIP proxy IP	
e		Transmitter Trans	
	1000	4. SIP 5. SIP password authentication user	
	AP send - MRT - ART JP - BALLA	Approximately (1) 500 Approximately (2) Approximately (2) Approxima	
ET		1	6. Save char

Note: Before you start your configuration, make sure that you have the needed information and resources to see the configuration through. You can use the device provisioning form to collect the configuration parameters. Axis provides no support for PBX configuration, but some configuration examples are available if needed.

Device provisioning form				
Brand/type of SIP registrar / PBX				
SIP user ID				
SIP authentication user				
SIP password				
SIP port number (if not 5060)				
SIP register address (FQDN)				
SIP proxy server address (FQDN)				
SIP via UDP or TCP				
DTMF RFC2833				
DTMF SIP info				
Video H.264 support enabled in PBX				
Sufficient PBX licenses available				
Supported voice codec				
RTP port range				
Contact details for SIP service provider customer support				
Contact details for PBX support				

Step 3: Configure 2N SIP Mic to initiate a VoIP call to a zone, via a PBX.

N ^a Blows	Set ione	*
	1. Ac	tivate the zone
a demonstra	2. Na	me the zone
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	3. Ch	oose SIP destination
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	5. Destin	ation's extension number d number)
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	671r	
		6. Save cha



Step 4: Assign the preconfigured zone to a 2N SIP Mic button.

When you press the button, 2N SIP Mic will initiate a call to the destination number (the zone/leader speaker).

5.6 F – Use DTMF for public announcement to a speaker group

Step 1: Enable DTMF on 2N SIP Mic.



Step 2: Configure a 2N SIP Mic button to send DTMF.



Step 3: Configure the leader speaker to receive DTMF tone.





Step 4: Configure the speaker to trigger the action "Play audio clip" upon receiving a DTMF tone.

About Axis Communications

Axis offers intelligent security solutions that enable a smarter, safer world. As the global market leader in network video, Axis is driving the industry by continually launching innovative network products based on an open platform – delivering high value to its customers and carried through a global partner network. Axis has long-term relationships with partners and provides them with knowledge and ground-breaking network products in existing and new markets.

Axis has more than 2,700 dedicated employees in more than 40 countries around the world, supported by a network of over 90,000 partners across 179 countries. Founded in 1984, Axis is a Sweden-based company listed on NASDAQ OMX Stockholm under the ticker AXIS.

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