

AXIS SIP Setup and Troubleshooting Guide

Using AXIS SIP devices with different PBX server



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Introduction

This guide aims to help users setup AXIS SIP products together with some commonly used PBXservers. It explains how to setup an AXIS product to use a PBX-server as well as how to configure the PBX-server itself. Please note that since this guide shows how to setup 3rd party PBX-servers, future versions might vary in exact functionality and visual appearance. However, the workflow should roughly be the same.

Most of the commercial PBX manufactures are using licencing models that favours their own SIP devices therefore it is important that you have enough third party licences available to be able to register AXIS SIP devices with the PBX.

1 Preparing an AXIS SIP device to register with a PBX

Make sure to factory default your AXIS product before making the changes below. Also make sure that port 5060 (UDP and TCP) are not blocked within your network since these ports are most commonly used by the SIP protocol.

1.1 Enabling SIP

The first thing we need to do is to enable SIP-functionality in our AXIS product as shown below. Some AXIS products have a SIP Setup assistant that guides you through an easy setup for the entire product (like button-initiated calls on Network Video Door Station). This guide only shows how to set up an account in the AXIS product not the specific product capabilities. If a Setup Assistant is available, it's recommended to be used. The same configuration specified below can be applied in the assistant separate pages.

Basic Setup	SIP Settings	•
	SIP Setup Assistant	
Video & Audio	Start the setup assistant for easy SIP configuration.	Start
* VoIP	SIP Settings	
Overview	C Enable SIP	
SIP Settings	Incoming SIP Calls	
DTMF Settings	Allow incoming SIP calls	
the Marco Co	Port Settings	
Live view Config	SIP port: 5060	
Detectors	SIP TLS port: 5061	
Applications	NAT Traversal	
Аррисацонз	Enable ICE	
Events	Enable STUN	
Recordings	Enable TURN	
Languages	Save	eset
System Options		
About		



1.2 Adding a SIP account

The next thing to do is to add the SIP account. Click on the **Add** button under **VoIP -> Account Settings** to bring up the screen below. The values used below are the same as in the CISCO Call manager.

Image: Association Modify Account Account Information Name: Association Image: Association Imad	
Modify Account Account Information Name: A8004 Default account (Note that only one account can b Account Credentials User ID: User ID as Authentication ID Authentication ID: Password: Caller ID: SIP Server Settings Domain name: Registrar address: 172.25.8.150	mod&id=
Account Information Name: A8004 The User ID Default account (Note that only one account can b Account Credentials User ID: User ID: User User ID as Authentication ID Authentication ID: AB004 Password: ID: SIP Server Settings Domain name: Registrar address: 172.25.8.150	0
Name: A8004 The User ID the extension number for user ID: Operault account (Note that only one account can be account Credentials The PBX-ser User ID: 100 User ID: 100 Authentication ID: a8004 Authentication ID: a8004 Password: Caller ID: SIP Server Settings Domain name: Registrar address: 172.25.8.150	
Caller ID:) is
Account Credentials the PBX-ser User ID: 100 Use Use User ID as Authentication ID Authentication ID: a8004 Password: Caller ID: 172.25.8.150 Domain name: Registrar address: 172.25.8.150 The Registrar address is the comparison of the term of the term of the term of the term of term	us in
User ID: 100 Use User ID as Authentication ID Authentication ID: a8004 Password: Caller ID: SIP Server Settings Domain name: Registrar address: 172.25.8.150 The Registrar address is the set of the	ver
Use User ID as Authentication ID Authentication ID: Password: Caller ID: SIP Server Settings Domain name: Registrar address: 172.25.8.150 The Regist address is the set of the s	
Authentication ID: a8004 The Authentiand Passwoologin/paswoologin/paswoologin/passwoologin/passwoologin/passwoolo	
Password: Caller ID: SIP Server Settings Domain name: Registrar address: 172.25.8.150 The Registration address is the set of th	ication ID
Caller ID: SIP Server Settings Domain name: Registrar address: 172.25.8.150 The Regist address is the set of the set	ord is the
SIP Server Settings Domain name: Registrar address: 172.25.8.150 The Regist address is to	-server
Domain name: Registrar address: 172.25.8.150 The Registra address is the second	
Registrar address: 172.25.8.150 The Regist address is the address is the registration of the registration	
address is the	trar
Transport Settings for the DBV-o	ne IP
Enable SIPS	
Transport mode: UDP 🔻	
Allow port update messages through MWI	
Proxy Settings	
Address Username	* (1) *
Add	
Account Status	
Reg. status: 🕒 OK (200)	
OK Cancel	

Under section 3.1 you can find an example how to setup a C3003-E



2 Configuring Cisco Unified Communications Manager

2.1 Introduction

These instructions assume that you have already installed CUCM as your PBX-server. In this example, we are using version 10.5 of CUCM. Later versions should be similar to setup although the visual appearance might change. The required steps are listed below.

- 1. Create a single phone security profile to be used for all AXIS devices.
- 2. Create a user for each AXIS device.
- 3. Add device information to the CUCM manager.

Please note that the steps described are for setting up basic functionality. There are many optional settings but they are not covered in this guide. Questions regarding the setup of Cisco PBX software should be directed to your Cisco integrator since it's not an AXIS product. You can give this guide to the Cisco Technician if he/she has questions on how the product should be configured.

2.2 Creating a Phone Security Profile

Select System -> Security -> Phone Security Profile as seen below.

cisco	Cisco U For Cisco Ur	nified CM	Administration ations Solutions
System 👻	Call Routing 👻	Media Resources	✓ Advanced Features ▼ Device ▼
Server Cisco U Cisco U Present Phone I	Inified CM Inified CM Group ce Redundancy G NTP Reference	roups	
Security	y	•	Certificate
Applicat	tion Server		Phone Security Profile
Licensir Geoloci	ng ation Configuration	•	SIP Trunk Security Profile CUMA Server Security Profile
Geoloca E911 M	ation Filter essages		

Next click on the Add New button and then choose Third-party SIP Device (Advanced) and click on the Next button as seen below.

Select the type of device	profile you would like to create-	
Phone Security Profile Type	* Third-party SIP Device (Advanced)	3 M



Enter the name for the profile and make sure to check the digest authentication checkbox as seen below.

Product Type: Device Protocol:	SIP-enhet från tredje part (avancerad) SIP
Name*	Axis SIP Device Security Profile
Description	
Nonce Validity Time*	600
Fransport Type *	TCP+UDP T
🗹 Enable Digest Aut	hentication
Parameters used in	Phone
SIP Phone Port* 506	0
	100 ⁸

Finally click on the **Save** button.

2.3 Creating a user

Select User Management -> End User as seen below.

Use	er Management 👻	Bulk Administr	ation 👻	Help -
	Application User			
	End User			
	User/Phone Add		•	
	SIP Realm			
	User Settings		+	
	Self-Provisioning			
	Assign Presence	Users		



Next click on the **Add New** Button and enter the **User ID**, **Last name** and **Digest Credentials** as seen below. We do not need to enter anything under Password since we are using digest authentication instead.

User Status	Enabled Local User
User ID *	a8004
Password	
Confirm Password	
Self-Service User ID	
PIN	
Confirm PIN	
Last name*	Last name
Middle name	
First name	
Title	
Directory URI	
Telephone Number	
Home Number	
Mobile Number	
Pager Number	
Mail ID	
Manager User ID	
Department	
User Locale	< None >
Associated PC	
Digest Credentials	
Confirm Digest Credential	s ••••
User Profile	Use System Default("Standard (Factory Default) View Details

Finally click on the **Save** button.

2.4 Creating the device information

Select **Device->Phone** as seen below

Device 👻	Application 👻 User	Management -	Bulk Ac
CTI Ro Gateke Gatew	oute Point eeper ay		
Phone			
Trunk Remot Device	e Destination	•	



Next click on the **Add New** button and then choose **Third-party SIP Device (Advanced)** and click on the **Next** button as seen below.

elect the t	pe of phone you would like to create	
Phone Type	* Third-party SIP Device (Advanced)	

Enter the **MAC Address** and set the **Device Pool** and **Phone Button Template** as seen below. You can also set the **Owner** to **Anonymous** as we are using digest authentication instead.

Device is not trusted		
1AC Address™	ACCC8E0208A1	
escription		
Device Pool*	Default	•
common Device Configuration	< None >	T
hone Button Template*	Third-party SIP Device (Advanced)	•
common Phone Profile*	Standard Common Phone Profile	•
Calling Search Space	< None >	T
AR Calling Search Space	< None >	T
ledia Resource Group List	< None >	T
ocation*	Hub_None	T
AR Group	< None >	۲
evice Mobility Mode*	Standard	۲
lwner	User Anonymous (Public/Shared Space)	
wner User ID		•
se Trusted Relay Point*	Standard	¥
lways Use Prime Line*	Standard	T
lways Use Prime Line for Voice Message st	Standard	¥
Geolocation	< None >	¥
Retry Video Call as Audio		
Ignore Presentation Indicators (internal	calls only)	
Logged Into Hunt Group		
Remote Device		



Then set the Device Security Profile, SIP Profile and Digest User as seen below.

BLF Presence Group*	Standard Presence group	۲
MTP Preferred Originating Codec*	711ulaw	Ţ
Device Security Profile*	Axis SIP Device Security Profile	•
Rerouting Calling Search Space	< None >	۲
SUBSCRIBE Calling Search Space	< None >	,
SIP Profile*	Standard SIP Profile	•
Digest User	a8004	•
Media Termination Point Requir	ed	
Unattended Port		
Require DTMF Reception		
Allow Presentation Sharing usir	ng BFCP	
Allow iX Applicable Media		

Click on the **Save** button and then click on the **Apply Configuration** button which displays the following window.

Note: Please save the configuration before continuing. When you click apply config, the dev go through a restart. When restart is initiated, calls in progress may be dropped but connected calls will be preserved unless the device pool includes SIP trunks.	ice may

Click on the **OK** button and then click on the **Line [1] – Add a new DN** link to set the extension number for the device.

	Modify Button Items
1	The Line [1] - Add a new DN
2	<u>אַזי Line [2] - Add a new DN</u>
3	Line [3] - Add a new DN
4	Line [4] - Add a new DN
5	Line [5] - Add a new DN
6	Line [6] - Add a new DN
7	Line [7] - Add a new DN
8	Line [8] - Add a new DN



Enter the **Directory Number** (extension number) as seen below.

Directory Number*	100		Urgent Priority
Route Partition	< None >	*	
Description			
Alerting Name			
ASCII Alerting Name			
External Call Control Profile	< None >	T	
Active			

Finally, click on the **Save** button and then you have completed the basic configuration of the AXIS SIP device.

2.5 Common causes of problems

If you are having problems getting the AXIS SIP device to register with the CUCM, go back to **Device → Phone** and click on the link to your SIP device and then click on the **Apply Config** button again. Some changes can take some time to apply.

If your PBX-server is not on your local network and you can't connect to it, looking into using STUN, ICE or TURN is advisable.

STUN and ICE are explained in chapter 6.1

.



3 Configuring a 3CX Phone System

3.1 Introduction

These instructions assume that you have already installed the 3CX Phone System. In this example, we are using version 14.x of the 3CX Phone System. Later versions should be similar to setup although the visual appearance might change. Please note that the steps described are for setting up basic functionality. There are many optional settings but they are not covered in this guide. Questions regarding the setup of 3CX software should be directed to your local 3CX representative since it's not an AXIS product.

3.2 Setting up extensions

Open up a web browser and go to the IP-address where you installed your 3CX Phone System. This will open up the 3CX management interface. Enter your credentials (the default credentials are admin/admin) and you should see the following menu to the left.

30X Server Manager • Ex	tensions	
Reports/Trunks Status	💄 Add Extension 🔮 Edit Extension 💄	Delete Extension
🕵 Extension Status	Filter:	
🕵 System Extensions Status	Extension Number First Name	Last Name
C 3CXPhone Clients	100	
금 Remote Connections		
The Phones		
😰 Server Activity Log		
🗟 Server Event Log		
😓 Services status		
> 🚨 Extensions		
📖 WebRTC Gateway		
📖 VoIP/PSTN Gateways		
IVOIP Providers		
Inbound Rules		
무급 Bridges		
↑ OutBound Rules		
> 👧 Digital Receptionist		
2. Ring Groups		
🔂 Call Queues		
> 🕒 Fax Machines		
> 🗘 Settings		
> 🛧 Updates		
> P Links		
> 👔 Help		

In the screenshot above we can see all the extensions currently listed in the PBX-server. There is only the default "100" extension to start with.

We can now add all the extensions that we want to use. Name them appropriately and setup a password for each one.





Click on Add Extension to start adding an extension. Enter the specific Extension Number and then enter a suitable First Name and Last Name, as this is what is used as Caller ID. Finally enter your ID and Password which are the login details setup in the AXIS SIP device.

▶ Add										
e E	tension Settings	🕝 Help								
Gener	al Voice Mail	Forwarding Rules	Phone Provisioning	BLF (Busy Lamp Fields)	3CXPhone	Other	Options	Office Hours Scheduling	Rights	
c	ser Information - onfigure user info	ormation below								
	Extension			1015	0					
	First Name			Karl-Theo	2					
	Last Name			Hofer	?					
	Email address			· · · · · · · · · · · · · · · · · · ·	0					
	Mobile Number	r			?					
A TI	uthentication he authentication	ID and Password ar	e used by the phone t	o authenticate with 3CX PI	hone System	. If the pl	none has a	user id field enter the exter	ision number.	
	ID			1015	2					
	Password			pass	?	***				

Finally click on the **OK** button to save and activate the changes.

After pressing the **OK** button a new view will give you the setup parameters you can use to configure the AXIS SIP device.

Extension	Created	
-----------	---------	--

Extension Created

Extension Number 1015 was created for Karl-Theo Hofer

You can find information on how to configure and provision your SIP phone here

The settings below are required to configure the SIP phone manually

Display name: Karl-Theo Hofer Proxy server / SIP server / registrar: 192.168.188.21:5060 Extension number / User ID: 1015 Authentication ID: 1015 Authentication Password: pass



3.3 Setting up paging

This can be useful to address and speak to several C3003-E at the same time. To do so please create for every C3003-E its own extension please repeat step 3.2 as often as needed.

Click on **Ring Groups** →Add Ring Group

🕵 Ring Groups Settings 🌈 Help								
General								
Enter the Ring Group details. The phones will ring until one of them is answered or until the timeout is reached.								
Virtual Extension Number (can not be in use as extension) 8000								
Name		Multi						
Ring Strategy		Paging 💌	?					
Ring Time (Seconds)		20	\odot					
Use Multicast for Paging								
Ring Group Members Select which extensions are a member of this Ring Group.								
1000 🔹	Add > < Remove	1009 1008	1					

Choose Paging as Ring Strategy

Move the extension that shall be member of the Ring Group into the member section. Press OK to safe the configuration.

If you now dial 8000 from one of the extensions the C3003-E will auto answer and you can transmit to all members at the same time.



3.4 Setting up the horn C3003-E

Go to Setup \rightarrow VoIP \rightarrow SIP Settings check the Enable SIP checkbox

AXIS	AXIS C3003-E Network Speaker Setup He
Basic Setup	SIP Settings
	SIP Settings
Audio	🖉 Enable SIP
• VoIP	Make a test call from the default SIP account to the specified SIP address. Default account: horn speaker (1015)
SIP Settings	Enter SIP address: sip(s):extension@domain Test call
Account Settings	Incoming Calls

Then go to Setup \rightarrow VoIP \rightarrow Account Settings click Add And a new window opens called Add Account see screenshot below.

Pacia Cotup	Account Settings					0			
basic setup	Name	SIP address	Transport	Default	Reg. status				
Audio	peer-to-peer	sip:192.168.188.26:506	UDP						
VoIP	1015 (1015)	1015 <sip:1015@192.168.188< td=""><td>23 UDP</td><td>0</td><td>۲</td><td></td><td></td></sip:1015@192.168.188<>	23 UDP	0	۲				
SIP Settings		AXIS C3003-	E Network Spea	ker - Goo	gle Chrome				
Account Settings		B 102 169		ator/ac		html2c	lo Action - mos		
Events		192.100.	.00.20/0pen	ator/aco	count_set.	siturii: c	IOACtion=mot		
Languages		Modify /	ccount						
System Ontions		Account Info	rmation						
system options	Add	lodify	1015						
About	Tect SID Call	Default acc	unt (Note that	only one	account can b	e the defa	ault account.)		
	Make a test call from	Account Cred	entials						
	Enter SIP address:	sip(s):extens	1015						
	-	🕑 Use User ID	as Authenticatio	n ID					
		Authentication I	0: 1015						
		Password:							
		Caller ID:	1015						
		SIP Server S	ttings						
		Domain name:							
		Registrar addres	s: 192.168.1	88.23					
		Transport Se	tings						
		Enable SIPS							
		Transport mode	UDP V						
		Allow port i	ndate messages	through M	IWT				
		Proxy Setting	s	chi oʻugir i.					
		Address		Us	ername				
							-		
		Add							
		Account Stat	15						
		Reg. status:	🔵 ОК (2	00)					
				OK	Cance				

Now the C3003 is registered with the3CX phone system and can call other extensions.



4 Peer to Peer call from GXV3240 IP phone to a A8004-VE

The easiest way to set up this IP phone is via web interface but you can also use phones touch screen to configure the device. All settings are done after factory reset which is available in the "Maintenance/Upgrade" section.

G X V 3 2 4	4 0	
	Enterprise Phone Administr	ation Interface
	Status Account Advanced Settings	Maintenance
🔞 Network Settings		0
(Wi-Fi Settings		
Time Settings	Download Device Configuration :	Download
Web/SSH Access	Configuration via LCD Menu :	Unrestricted
Dupgrade Upgrade	XML Config File Password :	
Syslog	HTTP/HTTPS User Name :	•
Logcat	HTTP/HTTPS Password :	•
Sebug	Upgrade Via :	нттр 🔤 🕕
Language	Firmware Server Path :	fm.grandstream.com/gs
TR-069	Config Server Path :	fm.grandstream.com/gs
Contacts	Firmware File Prefix :	•
LDAP Book	Firmware File Postfix :	•
😡 Device Manager	Config File Prefix :	•
	Config File Postfix :	
	mDNS Override Server :	Use Type A 🔤 🤑
	DHCP Option 66 Override Server :	Ves 🕕
	DHCP Option 120 Override SIP Server :	Yes 🕕
	3CX Auto Provision :	Ves
	Automatic Upgrade :	Check Every Day
	Automatic Upgrade Check Interval (m) :	10080
	Hour of the Day(0-23):	1
	Day of the Week(U-0) :	
	Auto Reboot to Upgrade Without Promot	Always Check at bootup
	:	I Yes
	Authenticate Conf File :	Tes .
	Factory Reset :	Clear the SD card Reset



Login to the Grandstream GXV3240 via the web interface by using the IP address.

Username = admin Password = admin

Enable "Account 1" and set all necessary information for this account in the "General Settings" section.

Account Active: check the box **Yes** Account Name: **AXIS A8004-VE** (this will appear on the screen of the phone) SIP Server: **enter the IP address of the A8004-VE** SIP User ID: **111** Click **Save** at the bottom Click **Apply** to apply the configuration changes

ATTENTION: the SIP user ID is the number you have to dial from your Grandstream in order to call the AXIS SIP device!!!

	Apply configuration changes.									
G X V 3 2 4	4 0	Dl	A]:		T	Ť	Theme 🕞 Re	boot Enc	Jish	xit
	Enterpr	ise Phone	e Admini	stration	1 Interface	9				
	Status	Account	Advanced	Settings	Maintenance					
Or General Settings		Account 1	Account 2	Account 3	Account 4	Account 5	Account 6		(D
🔞 Network Settings										
SIP Settings			Account Activ	e:	✓ Yes	٦				
Codec Settings			Account Nam	e:	AXIS A8004-VE					
Call Settings			SIP Serve	er :	192.168.0.110					
			SIP User I	D:	111					
		SIP A	uthentication I	D :						
		SIP Authentie	cation Passwor	d :						
		V	pice Mail Userl	D :						
			Nam	e:						
		Show Acc	ount Name On	ly :	• Yes					
			Tel UF	RI :	Disable		-			
					Save	Cancel				



Go to the "SIP Settings" section and disable/uncheck the "SIP Registration" field

Verify that the **Local SIP Port** is set to **5060** Click **Save** at the bottom Click **Apply** to apply the configuration changes

				Αμ	ply configura	tion changes.	Apply
G X V 3 2 4	40				1	heme 🕞 Reboo	ot 🔀 Exit
	Enterprise Pl	none Admir	nistratio	n Interface			English 👻
	Status Acco	ount Advance	ed Settings	Maintenance			
General Settings	Accou	nt 1 Account 2	Account	3 Account 4	Account 5	Account 6	1
🔞 Network Settings							
SIP Settings		SIP Registra	tion :	Yes			
Codec Settings	Unregister Be	fore New Registra	tion :	No			
🔞 Call Settings	F	Register Expiration	(m) :	60			
	Reregiste	r before Expiratior	n (s) :	0			
	Registratio	on Retry Wait Time	e (s) :	20			
		Local SIP	Port :	5060			
		SUBSCRIBE for N	IVVI :	Yes			
		Enable Session Ti	mer :				
		Session Expiratior	n (s) :	180			
		Min-SE	E (S) :	90			
	ι	JAC Specify Refree	sher :	Omit		l .	
	ι	JAS Specify Refree	sher :	UAC			
		Force INV	ITE :	• Yes			
		Caller Request Ti	mer :	• Yes			



Navigate to VoIP > Account Settings. The account settings for the demo account that was created earlier in the A8004-VE will need to match the account settings added in the GXV3240.

Select the account and click on Modify. It should the account that is checked as default.

Basic Setup	Account Set	tings			
	Name	SIP address	Transport	Default	Reg. status
Video & Audio	peer-to-peer	sip:192.168.0.110:5060	UDP		\bigcirc
VoIP Overview SIP Settings Account Settings DTMF Settings	Demo Account (111)	sip:192.168.0.110:5060	UDP	0	۲
Live View Config					
Detectors					
Applications	Add Modif	y Remove			
Events	Test SIP Call				
Recordings Languages	Make a test call from th Enter SIP address: sip(s	e selected SIP account to the s ;):extension@domain	pecified SIP ad	ldress.	
System Options					
About					



Name: no need to change it

User ID: **111** (this should match the user ID added in the GXV3240 account) Password: should remain blank (unless you added a password in the GXV3240 account) Click OK

Modify Acc	ount 🚱
Account Information	tion
Name:	Demo Account
Default account	(Note that only one account can be the default account.)
Account Credenti	als
User ID:	111
🕑 Use User ID as A	uthentication ID
Authentication ID:	111
Password:	
Caller ID:	
SIP Server Settin	gs
Domain name:	
Registrar address:	
Transport Setting	IS
Enable SIPS	
Transport mode:	UDP V
Allow port update	e messages through MWI
Proxy Settings	
Address	Username
Add	
Account Status	
Reg. status:	Does not register (0)
	OK Cancel

If you now dial 111 from your Grandstream the A8004-VE should take the call.



5 Configuring the Avaya Office Manager Vers. 9.1.4.0

To be able to use the system you have to create a user and an extension.

Solution Call Manag	ment System Settings Security Manager Applications
Solution	
SOLUTION OBJECTS 🗸	
View All (1)	Actions -
SERVER STATUS	IPOfficeDemo 172.26.30.13 Primary: Select
Online (1)	
Offline (0)	
SERVER TYPE	
Servers (1)	
Expansions (0)	
Application Servers (0)	
	IP Office Select Web Manager Version: 9.1.4.0 build 137
	Close



First you have to create a user

Go to Call Management \rightarrow Users



In the new view click Add Users choose the server instance you like to create the user on



Avsys IP Office Select Web Manager - Mozilla Firefox Arkiv Bedigers Viga Historik Bokmarken Verktyg Hjälp	-			-	Second State	-				
() Case 573743 ≍ / 🔼 Avaya IP Office Select ×	SIP Servers - QA Wiki	× 🛄 H	tps://sernformatt	d x 👔 #97267 :: 1	frouble 🛛 🗶 🖾 A	UOS A8004-VE Netwo	× 🧿 Case 573479	× 🔤 AXIS C3	3003-E Network × 🛛 🥅 https://	//ser_nformatted 🕱 🧿 Case 572843 🛛 🛪 🕂
(←)	html					8	🛛 🔻 📌 Fast C 🗌 🔍	, Sók		♣ 0 4 ⊜ 0 0 ♠ # - ≡
🗌 Axis_Video_Products 🍍 galaxis 🔕 My open cases 🕢 Aditro S	jälvservice 👔 MyTrouble :: Tr	rouble 🔸 F	W release 🗌 Axis	/APIX® Library 🎯 Ca	se 567868 🎬 SIP Servers -	QA Wiki 🐲 Backlog	Project Portfolio	GT SoftSwitch Russia 🗍 Mi	ediaBinaryLoader.axd 🔹 vapix fu	mktion 🗌 STIX - Home 🛐 AXIS P3364 = Trouble 🛛 😕 😭
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Second step: You have to create an extension

Go to Call Management \rightarrow Extensions

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In the new view click Add Extensions choose the server instance you like to create the user on



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Please choose "Common" and configure the extension and press update

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Now assign a USER and a PASSWORD to the extension



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Now you can register your AXIS SIP DEVICE with the Avaya PBX.

6 Creating an extension and SIP USER on Asterisk PBX version 1.8.x

Create a user and assign an extension to the user



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If your AXIS SIP device needs Video Support it's important to choose H264. In the displayed section you have to configure all account settings, create a password, choose SIP, if your SIP device is part of a NAT network choose NAT, and use the right DTMF mode, RFC2833 is recommended.



In the left hand menu choose Admin Settings \rightarrow Advanced Options \rightarrow click Show Advanced Options

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In the middle menu choose MISC and do your Video settings, check the Support for SIP Video box, and press Save and then press "Apply Changes"



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Now your Axis SIP device can register with the PBX

7 Common problem descriptions regarding sound quality

Problem	Intermittently	Periodically	Continuously
Conversational difficulty	High levels of jitter cause large numbers of packets to be discarded by the jitter buffer in the receiving IP phone or gateway. This may result in severe degradation in call quality or large increases in delay.		Echo becomes a problem when combined with a significant amount of delay. For example, if an IP phone was connected over wide area IP network to a VoIP Gateway then the delay would be large – echo that occurred on the trunk side of the Gateway would be audible in the IP Phone. If a user reports an acho problem then the
			source of this problem is



		likely to be on the other end of the connection. In the presence of high levels of delay the normal "protocol" of conversation breaks down. In addition, delay can make echo problems more obvious and annoying.
Gaps in speech	Voice Activity This may be due to a high rate of packet loss or packet discard due to jitter, or to a problem, with Detection associated with an echo canceller. Users report that words are being clipped - similar in effect to a lower quality speakerphone.	If users report that the start and end of words are being "clipped" then this is typically due to the Voice Activity Detector in the VoIP hardware. Voice Activity Detectors are used for silence suppression in packet voice systems, for echo suppression in echo cancellers and for echo suppression or directional control in speakerphones. Clipping can be the result of the sound level settings in the VoIP hardware being incorrectly configured.



Tick or Pop Sounds	Access link problems can be reduced by Using priority queuing for delay sensitive voice and video traffic Reducing the maximum MTU size on low speed links (512 kbits/s or less) Increasing the capacity of the access link If multiple links are used, then applying load sharing to maximize use of capacity Applying call admission control to limit the number of calls Using fragmentation and interleaving.	Low rates of timing drift may cause a periodic audible "tick". VoIP systems can sometimes hide this by doing necessary timing adjustments during silence periods. If an NTP timing server is used then VoIP systems may resynchronize or adjust their clock speed automatically. High rates of drift can be much more problematic, and may be symptomatic of hardware problems. These can be caused by high temperatures in end systems such as PCs or due to the use of cheap ceramic resonators instead of crystals in low cost IP phones.	Access link problems can be reduced by • Using priority queuing for delay sensitive voice and video traffic • Reducing the maximum MTU size on low speed links (512 kbits/s or less) • Increasing the capacity of the access link • If multiple links are used, then applying load sharing to maximize use of capacity • Applying call admission control to limit the number of calls • Using fragmentation and interleaving.
Audio quality poor or noisy, level too low or high			Generally if a connections sounds "dead" then the level of background noise is too low. This can be due to an echo canceller being mis-configured or to a poor loss plan. Other causes are mis-configuration of the background noise level in a Voice Activity Detection / Silence Elimination or Line Echo Canceller function. Voice sounds hollow, this is generally due to a high level of echo with a small amount of delay. For example, if an



		IP phone was connected over a LAN to a VoIP Gateway then the delay would be very small – echo that occurred on the trunk side of the Gateway may cause "hollowness" in the IP Phone. Distortion: Users may report that calls sound "noisy", "dead", "hollow", "cavelike" or "tunnel-like", have echo, sound slightly distorted, sound robotic, or be very choppy or garbled (see also Packet Loss)
speech broken up or distorted	be reduced by	be reduced by
	 Using priority queuing for delay sensitive voice and video traffic Reducing the maximum MTU size on low speed links (512 kbits/s or less) Increasing the capacity of the access link If multiple links are used, then applying load sharing to maximize use of capacity Applying call admission control to limit the number of calls Using fragmentation and interleaving. 	 Using priority queuing for delay sensitive voice and video traffic Reducing the maximum MTU size on low speed links (512 kbits/s or less) Increasing the capacity of the access link If multiple links are used, then applying load sharing to maximize use of capacity Applying call admission control to limit the number of calls Using fragmentation and interleaving.



8 How to troubleshoot SIP

Preliminary remarks:

Before you start trouble shooting please ask for the following information For SIP devices that are registered to a PBX/SIP registrar: If the SIP endpoint is registered to a pubic VOIP Network ask the customer if Axis can use the SIP account for test purposes.

SIP User SIP Authentication User SIP Password SIP Registrar IP address or Domain Name and its port number DTMF protocol (SIP-Info, RFC2833) Proxy Server and its port number (if applicable)

For Peer To Peer devices Make sure both devices are on the same network and both SIP clients do support Peer To Peer calls.

For both cases please always ask for a server report Ask always for a Wireshark trace http://IP of the camera/axis-cgi/debug/debug.tgz?cmd=pcapdump&duration=120



SIP trouble shooting first steps

We start with the basics of SIP before going through the important message types, all the important responses, call flows and media.

First step should be to determine if you will have a peer to peer connection or if you are going to register your SIP client with a SIP registrar.

To get a peer to peer connection to work you have to ensure that both SIP endpoint are in the same network.

Because when addressing the calling partner you send an invite that looks like <u>SIP:username@ipadress.to</u> be able to reach the target IP address you have to part of the same network.

If you have to register with a Registrar you need some register information from the operator of the SIP registrar/PBX.

The Display name (if you register with a public server this can be a public phone number)

Proxy server / SIP Server /registrar Ip address/FQDN (this can be a domain name or a IP address your client has to register to)

SIP Registrar Port number your client has to register to (default is 5060)

Extension number /User ID (that's your user name)

Authentication ID: this is your authentication Id needed for the challenge request more under "**The SIP registration mechanism**"

Authentication Password: is just your password.



Please install a SIP soft phone (I suggest EKIGA http://www.ekiga.org) to make calls and configure/install an A8004-VE or a C3003-E as peer2peer or as SIP client. In this example here I use a Grandstream phone .

To see those things from a user's standpoint is very important. Then we go to the protocol itself. You can make some calls and then go through the SIP messages to understand what happened.

For dissecting the call flow, **Wireshark(WS) is the preferred tool**. Wireshark is a free and can be found under <u>https://www.wireshark.org/</u>.

Configure WS to capture the packets on a particular interface (e.g. your LAN card). Press "go" and Wireshark will then record every IP packet that comes to or from your PC through this particular interface. Press "stop" when you think you've captured what you need.

The first step is to make a two-party call. Ext. 1010 (Ekiga) calls Ext. 1012 (Grandstream). Grandstream answers the call. Ekiga hangs up the call. This will cause the Ekiga phone to send INVITE, ACK, and BYE requests. The Grandstream will send 100 Trying, 180 Ringing, and 200 OK response messages.



After capturing the packets you want to look at them.

_																
4	Capturing from Wireless Network Connection [Wireshark 1.12.9 (v1.12.9-0-gfadb421 from master-1.12)]															
Eile	<u>E</u> dit	: <u>V</u> iew <u>G</u> o	<u>Capture</u> <u>Analyze</u>	Statistics	Telephony <u>T</u> ools	Internals H	elp									
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Filt	er:					Expression	on Clear	Apply Save								
No.		Time		Sourc	e	Destination		Protocol	Length	Info						
	749	2016-04-04	15:21:06.988	38160192.	168.188.20	192.168	188.23	SIP/SD	F 1455	Request:	INVITE	E sip:	1013@192.1	68.188	.23	
	750	2016-04-04	15:21:06.992	2030192.	168.188.21	192.168	188.20	TCP	1514	80→55037	[PSH,	ACK]	Seq=392355	Ack=1	Win=946	Len=1460
	751	2016-04-04	15:21:06.992	2570192.	168.188.21	192.168	188.20	TCP	322	80→55037	[PSH,	ACK]	Seq=393815	Ack=1	Win=946	Len=268
	752	2016-04-04	15:21:06.992	2860192.	168.188.20	192.168	188.21	TCP	54	55037→80	[ACK]	Seq=1	Ack=39408	3 Win=	6978 Ler	=0
	753	2016-04-04	15:21:06.998	36780192.	168.188.21	192.168	188.20	TCP	324	80→55037	[PSH,	ACK]	Seq=394083	Ack=1	Win=946	Len=270
	754	2016-04-04	15:21:07.065	59500192.	168.188.21	192.168	188.20	TCP	371	80→55037	[PSH,	ACK]	Seq=394353	Ack=1	Win=946	Len=317
	755	2016-04-04	15:21:07.066	50150192.	168.188.20	192.168	188.21	TCP	54	55037→80	[ACK]	Seq=1	Ack=39467	0 Win=	6975 Ler	=0
	756	2016-04-04	15:21:07.078	36150192.	168.188.21	192.168	188.20	TCP	1514	80→55037	[PSH,	ACK]	Seq=394670	Ack=1	Win=946	Len=1460
	757	2016-04-04	15.21.07.078	86500102	168 188 21	102 168	188 20	TCP	833	80-55037	Гран	ACK1	Sen-306130	Ack-1	win-046	Lon-770

The easiest way to tell Wireshark to only show you SIP messages and disregard everything else is with the "VoIP Calls" command. This can be invoked as follows.

Wireless Network Connection [Wireshark 1.12.9 (v1.12.9-0-gfadb421 from	master-1127)	
Elle Edit View Go Capture Analyze Statistics Telephony Tools (r	stemals Help	
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Filter IA32	ression Clear Apply Save	
No. Time Sourc ISUP Messages	ation Protocol Length Info	
3533 2016-04-04 15:30:05.6137510192 LTE	168.188.20 H264 773 PT-H264, SSRC=0x1D25A10A, Seq=19167, Time=971212164, Mark NAL unit - Coded slice of a non-IDR picture	
3534 2016-04-04 15:30:05.6196830192 MTP3	168.188.24 RTP 214 PT=ITU-T G.711 PCMU, SSRC=0x486BC3C3, Seq=17384, Time=47520	
3535 2016-04-04 15:30:05.6212430192 gtp	• 168.188.20 TCP 1514 80-55037 [P5H, ACK] Seq=1118780 AcK=1 Win=946 Len=1460	
3537 2016-04-04 15:30:05.6214180192	168.108.20 TCP 974 80-5503 FPSH, ACK 3 500-1120240 ACK = 14/10-246 Len=920	
3538 2016-04-04 15:30:05.6225530192	168.188.24 SIP 474 Request: BYE sip:10120192.168.188.24:5060	
3539 2016-04-04 15:30:05.6267550192 SMPPOperations	168.188.20 RTP 214 PT=ITU-T G.711 PCMU, SSRC=0x260DA57, Seq=28128, Time=534233111	
3540 2016-04-04 15:30:05.62/8130192 10- Messages	Lo8.188.24 H264 146 PT-H264, SSRC-UX881DSCF, Seq-24162, Time-S31810 NAL unit - Coded Silce of a non-IDR picture IS8 24 H264 120 PT-H264 SSRC-UX881DSCF, Seq-24162, Time-S31810 Nack Native Unit - Coded Silce of a non-IDR picture	
3542 2016-04-04 15:30:05.6295070192	168.188.20 SIP SIP STATUS 200 K	· · · · · · · · · · · · · · · · · · ·
3543 2016-04-04 15:30:05.6400860192 gp.	168.188.24 RTP 214 PT=ITU-T G.711 PCMU, SSRC=0x486BC3C3, Seq=17385, Time=47680	
WAP-WSP.	150 100 30 Trn 310 00 55032 Foru 1002 ros 113150 161 1 010 015 1 00 305	1.
Frame 3538: 474 bytes on wire (3792 bits), 474 bytes	s captured (3792 bits) on interface 0	
Ethernet II, Src: IntelCor_1b:b6:51 (fc:f8:ae:1b:b6	:51), Dst: Grandstr_66:f2:8e (00:0b:82:66:f2:8e)	
Internet Protocol version 4, Src: 192.168.188.20 (1)	J2.168.188.20), Dst: 192.168.188.24 (192.168.188.24)	
User Datagram Protocol, Src Port: 5060 (5060), DST P Session Initiation Protocol (EVE)	vort: 3060 (3060)	
Request-Line: BYE sip:10120192.168.188.24:5060 5IF	P/2.0	
⊞ Message Header		
0000 00 0b 82 66 f2 8e fc f8 ae 1b b6 51 08 00 45 00		
0010 01 cc 25 a4 00 00 80 11 19 ff c0 a8 bc 14 c0 a8	3 - Marrier and a second s	
0030 70 3a 31 30 31 32 40 31 39 32 2e 31 36 38 2e 31	i piloi201 92.168.1	
0040 38 38 2e 32 34 3a 35 30 36 30 20 53 49 50 2f 32 0050 2e 30 0d 0a 43 53 65 71 3a 20 33 20 42 50 45 0d	2 88.24:50 60 STP/2	
0060 0a 56 69 61 3a 20 53 49 50 2f 32 2e 30 2f 55 44	4 .via: 51 P/2.0/UD	
0070 50 20 31 39 32 2e 31 36 38 2e 31 38 38 2e 32 30 0080 20 35 20 36 20 36 67 77 61 60 62 69 2d 70 20 66	/ P 192.16 8.188.20	•

On my PC, one SIP call was made while Wireshark was running in capture mode. Each SIP call has its own entry.



Wireless	Netwo	rk Conn	ection - V	OIP Calls	110	DT 11264	cene 0.0004	afer car	- 24462		<u> </u>		
							Detected	1 VoIP Call. Se	elected 1 Cal	l.			
Start Time	-	Stop T	ime	 Initial 9 	Speaker	 From 		▲ To		 Protocol 	 Packets 	▲ State	 Comment
3.4	457843		15.26840	2 192.16	8.188.20	<sip:k< td=""><td>arlth@192.168.1</td><td>88.2) < sip:1012</td><td>2@192.168.18</td><td>8.24 SIP</td><td></td><td>19 COMP</td><td>LETED</td></sip:k<>	arlth@192.168.1	88.2) < sip:1012	2@192.168.18	8.24 SIP		19 COMP	LETED
•													
						Total: C	Ills: 1 Start pack	kets: 0 Compl	leted calls: 1	Rejected calls:	0		
		Prepar	e Filter			Flow		Player		Sel	ect <u>A</u> ll		<u>C</u> lose

Select the session and click on the "Flow" button to see the following.



Time	192.168.188.20 192.168.188.2	Comment
2016-04-04 15:29:53.818948000	INVITE SDP (Una	SIP From: <sip:karith@192.168.188.20 td="" to:<sip:1012@192.168.188<=""></sip:karith@192.168.188.20>
2016-04-04 15:29:53.822605000	(5060) 100 Trying	SIP Status
2016-04-04 15:29:53.844613000	(5050) 180 Ringing	SIP Status
2016-04-04 15:29:59.071894000	(5060) 200 OK SDP (g71	SIP Status
2016-04-04 15:29:59.078517000	(5060) ACK	SIP Request
2016-04-04 15:29:59.680479000	(5066) RTP (g711U) (5004)	RTP Num packets:17 Duration:0.319s SSRC:0x486BC3C3
2016-04-04 15:29:59.754657000	(5068) RTP (H264)	RTP Num packets:19 Duration:0.229s SSRC:0x8B1D5CF
2016-04-04 15:29:59.755472000	[INVITE SDP (g711] (5060)	SIP From: <sip:karlth@192.168.188.20 td="" to:<sip:1012@192.168.188<=""></sip:karlth@192.168.188.20>
2016-04-04 15:30:00.002833000	(5060) 100 Trying	SIP Status
2016-04-04 15:30:00.004257000	(5060) 200 OK SDP (g71	SIP Status
2016-04-04 15:30:00.005991000	(5060) ACK	SIP Request
2016-04-04 15:30:00.017238000		SIP Request
2016-04-04 15:30:00.017550000	(5068) RTP (H264)	RTP Num packets:351 Duration:5.643s SSRC:0x8B1D5CF
2016-04-04 15:30:00.020842000	(5056). RTP (g711U)	RTP Num packets:283 Duration:5.639s SSRC:0x486BC3C3
2016-04-04 15:30:00.032177000	(5060) 200 OK (5060)	SIP Status
2016-04-04 15:30:00.616534000	(5060) INFO	SIP Request
2016-04-04 15:30:00.616547000	(5066) RTP (g711U) (5004)	RTP Num packets:271 Duration:5.821s SSRC:0x260DA57
2016-04-04 15:30:00.617277000	(5060) 200 OK	SIP Status
2016-04-04 15:30:00.629369000	(5060) INFO (5060)	SIP Request
2016-04-04 15:30:00.631221000	(5060) 200 OK	SIP Status
2016-04-04 15:30:00.731215000	(5068) RTP (H264)	RTP Num packets:778 Duration:5.385s SSRC:0x1D25A10A
2016-04-04 15:30:04.314107000	(5060) INFO	SIP Request
2016-04-04 15:30:04.314889000	(5060) 200 OK	SIP Status
2016-04-04 15:30:05.622553000	(5060) BYE (5060)	SIP Request
2016-04-04 15:30:05.629507000	(5060) 200 OK	SIP Status
	: :	
	Save <u>A</u> s	<u>C</u> lose

Wireshark displays the call flow in an easy-to-understand ladder graph. To get inside one of these message simply click on it. .



For example, if I click on the INVITE, I see the following:

1089 2016-04-04 15:29:59.7549570192.168.188.20	192.168.188.24	H264	105 PT=H264,	SSRC=0x8B1D5CF	, Seq	=23800, Time=0, Mark NAL	unit - Coded slice of an
1090 2016-04-04 15:29:59.7554720192.168.188.20	192.168.188.24	SIP/SDF	1010 Request:	INVITE sip:101	2@192	.168.188.24:5060, in-dial	og
1091 2016-04-04 15:29:59.7602060192.168.188.20	SRC=0x	=0x486Bc3c3, Seq=17091, Time=640					
1092 2016-04-04 15:29:59.7829840192.168.188.20	192.168.188.24	RTP	214 PT=ITU-T	G.711 PCMU, SS	SRC=0x	486BC3C3, Seq=17092, Time	=800
1093 2016-04-04 15:29:59.7868110192.168.188.20	192.168.188.24	H264	80 PT=H264,	SSRC=0x8B1D5CF	, seq	=23801, Time=5940 NAL uni	t - Coded slice of a non-
1094 2016-04-04 15:29:59.7869840192.168.188.20	192.168.188.24	H264	61 PT=H264,	SSRC=0x8B1D5CF	;, Seq	=23802, Time=5940, Mark N	AL unit - Coded slice of
1095 2016-04-04 15:29:59.8002280192.168.188.20	192.168.188.24	RTP	214 PT=ITU-T	G.711 PCMU, SS	SRC=0x	486BC3C3, Seq=17093, Time	=960
1006 3016 04 04 15:30:50 0100030103 160 100 30	100 160 100 04	11264	00 pt 1064	cone Augustator	- Coc		 Coded clice of a non
					0		
Frame 1090: 1010 bytes on wire (8080 bits), 1010 I	bytes captured (8080	bits) on	interface 0		4	Wireless Network Connection - Gr	aph Analysis
Ethernet II, Src: IntelCor_1b:b6:51 (fc:f8:ae:1b:b	o6:51), Dst: Grandst	66:†2:8	e (00:0b:82:66:	†2:8e)	10		
Internet Protocol Version 4, Src: 192.168.188.20	(192.168.188.20), Ds	t: 192.16	8.188.24 (192.1	68.188.24)			102150100.20
User Datagram Protocol Src Port: 5060 (5060), Dsi	t Port: 5060 (5060)					Time	192.108.188.20
Session Initiation Protocol (INVITE)							192.100.100.2
Request-time. INVITE STP. 1012@192.108.188.24:500	50 SIP/2.0					2016-04-04 15:29:53.818948000	INVITE SDP (Una.
Method: INVITE						2016-04-04 15:29:53.822605000	100 Tryings
Request-URI: s1p:1012@192.168.188.24:5060						2016-04-04 15:20:52 844612000	(\$060) - 180 Ringing - (\$060)
[Resent Packet: Faise]						2010-04-04 15:25:55:544015000	(5060) (5060) (5060)
■ Message Header						2016-04-04 15:29:59.071894000	(\$060) (\$060)
	abcabus-464103 3413	010 0000	-01-1495-00596			2016-04-04 15:29:59.078517000	(5060) ACK (5060) S
	9NG4DK5C464103-3412	1910-8966	-a01048b9058T;r	port		2016-04-04 15:29:59.680479000	RTP (g711U)
User-Agent: EKiga/4.0.2		4000000				2016-04-04 15:29:59 754657000	RTP (H264)
From: <s1p:karitn@192.168.188.20>;tag=D023380;</s1p:karitn@192.168.188.20>	3-3412-1910-8965-a01	14809058T					(5068) (5006) (5006) (5006)
Call-1D: 14243803-3412-1910-8965-a0104809058T0	PLAPKARLIHI					2010-04-04 15:29:59.755472000	(5060)
Supported: 100rel, replaces						2016-04-04 15:30:00.002833000	(5060) 100 Trying (5060) 5
						2016-04-04 15:30:00.004257000	200 OK SDP (g71
Allow, TAVITE ACK OPTIONS BYE CANCEL SUBSCRIPT	NOTTEN DEFED MESSA					2016-04-04 15:30:00.005991000	ACK
Contont Longth: 284	E, NOTIFT, REFER, MESSA	aE, INFO, P	ING, PRACK			2016 04 04 15:20:00 017228000	(5060) (S060) (S060)
Content Type: application/sdp						2010-04-04 13:50:00:017258000	(5060)
Max_Eorwards: 70						2016-04-04 15:30:00.01/550000	(\$068) (\$006)
Max Forwards. 70						2016-04-04 15:30:00.020842000	(5066) RTP (g/11U) (5004) F
- Session Description Protocol					_	2016-04-04 15:30:00.032177000	200 OK
Version (v): 0						2016-04-04 15:30:00 616534000	INFO I
Owner/Creator Session Id (0): - 1459776593	2 TN TP4 192 168 18	8 20					(5060) RTD (a71111)
Session Name (s): Ekiga/4.0.2	2 10 114 1521100.10	5.20				2016-04-04 15:50:00.616547000	(5066) (5004)
Connection Information (c): IN TP4 192.168.1	188.20					2016-04-04 15:30:00.617277000	(5060) 200 OK (5060) 5
Time Description, active time (t): 0 0						2016-04-04 15:30:00.629369000	INFO Shern S
Media Description, name and address (m): au	dio 5066 RTP/AVP 0 1	01				2016-04-04 15:30:00.631221000	200 OK
Media Attribute (a): sendrecv						2016 04 04 15:20:00 721 21 5000	(5060) (15060) (15060) (15060)
Media Attribute (a): rtpmap:0 PCMU/8000/1						2010-04-04 13:50:00.751215000	(5068)
Media Attribute (a): rtpmap:101 telephone-ev	/ent/8000					2016-04-04 15:30:04.314107000	(5060)
Media Attribute (a): fmtp:101 0-16,32,36						2016-04-04 15:30:04.314889000	(SOED) 200 OK (SOED) 5
Media Attribute (a): maxptime:240						2016-04-04 15:30:05.622553000	BYE
Media Description, name and address (m): vio	deo 5068 RTP/AVP 113					2016-04-04 15:30:05 629507000	200 OK
Bandwidth Information (b): A5:4096						2010 04 04 19:50:05:025507000	(\$060)1 1(\$060)
Bandwidth Information (b): TIA5:4096000							4
Media Attribute (a): sendrecv							· · · · · · · · · · · · · · · · · · ·
Media Attribute (a): rtpmap:113 H264/90000						_	
	ax-mbps=190080;profi	le-level-	id=42801e				Save <u>A</u> s

Do you see that little plus sign ("+") next to the words "Session Initiation Protocol"? Click on it and the actual INVITE message will open up.

Do you see that little plus sign ("+") next to the words "Session Description Protocol"? Click on it and the actual INVITE message will open up.

At this point you can examine the SIP URI, headers, and even the message body. Since SIP is fairly easy to read and understand, it doesn't take much to figure out what is happening with each message in a call flow. Later in the document we will have a look at the registration process and the SDP part.

You can save these traces, too. Wireshark saves traces in pcap (packet capture) files. On our AXIS devices you have the possibility to capture such files directly on the device via VAPIX

http://root:pass@IP/AXIS-cgi/debug/debug.tgz?cmd=pcapdump&duration=30



(AXIS device will take 30s network trace)

One last WS feature I want to show you t is how to get/replay to the actual media. As you may know, media is sent in <u>RTP</u> packets and since RTP is just another kind of IP packet, Wireshark captures those, too. Go back to the list of SIP calls, select the one in question, and press "Play" to see the following.



This allows you to "Decode" and "Play" the audio stream between the Softphone and the Grandstream phone.

This is a good way to check the audio quality.

9 UNDERSTANDING SESSION DESCRIPTION PROTOCOL (SDP)

It's not possible to trouble shoot SIP without having a fairly good understanding of the Session Description Protocol (SDP). SIP only deals with establishing, modifying, and ending sessions but SDP takes care about the media within the session.

SDP is a protocol that describes the media of a session but it doesn't negotiate the media. Instead, one party tells the other party, "here are all the media types I can support — pick one and use it."

SDP contains a series of <character>=<value> lines, where <character> is a single case-sensitive alphabetic character and <value> is structured text.

SDP has three main sections – session, timing, and media descriptions. Each message may contain multiple timing and media descriptions, but only one session description.

The definition of those sections and their possible contents are as follows. It's important to know that not every character/value may be present in an SDP message.

Session description

- v= (protocol version number, currently only o)
- o= (originator and session identifier : username, id, version number, network address)

s= (session name : mandatory with at least one UTF-8-encoded character)

i=* (session title or short information)

u=* (URI of description)

e=* (zero or more email address with optional name of contacts)

 $p{=}^{\ast}$ (zero or more phone number with optional name of contacts)

c=* (connection information—not required if included in all media)

b=* (zero or more bandwidth information lines)

One or more Time descriptions ("t=" and "r=" lines; see below)

- z=* (time zone adjustments)
- k=* (encryption key)
- a=* (zero or more session attribute lines)

Zero or more Media descriptions (each one starting by an "m=" line; see below)

Time description (mandatory)

- t= (time the session is active)
- r=* (zero or more repeat times)

Media description (if present)

- m= (media name and transport address)
- i=* (media title or information field)
- c=* (connection information optional if included at session level)
- b=* (zero or more bandwidth information lines)
- k=* (encryption key)
- a=* (zero or more media attribute lines overriding the Session attribute lines)

For Example

The following is an example of an actual SDP message.

v=0

0=1011 2890844526 2890844526 IN IP4 192.168.188.21

pg. 42

s= SDP Blog

c=IN IP4 192.168.188.23

t=o o

m=audio 49170 RTP/AVP 0 8 97

a=rtpmap:o PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:97 iLBC/8000

m=video 9078 RTP/AVP 96

a=rtpmap:31 H264/90000

The following lines are worth to pay attention to in an SDP message

c= This will tell me the IP address where the media will come from and where it should be sent to.

m= There will be an entry for each media type. If your client supports real-time audio there will be an m= audio line. If your client supports real-time video there will be m=video line. Each media line indicates the port number and the type of codecs that will be defined in attribute lines.

a= There will be an attribute line for each codec advertised in the media line.

Looking at the example above we can see the following:

The client will use IP version 4 with an address of 192.168.188.21. It can support three audio codecs and one video codec. The audio codecs are G.711 uLaw (PCMU), G.711 aLaw (PCMA), and iLBC. The audio codecs will use port 49170 and all have a sample rate of 8000 Hz. The video codec is H.264 on port 51327.

After receiving the SIP message (Invite with SDP) with the above SDP content, the recipient will respond with a SDP packet identifying its IP address, ports, and codec values. The recipient will also pick from the list of the sender's codecs which ones it will use and potentially start real-time media flows.

If you like take some Wireshark traces and try to figure out how media is being described and used.

10 Traversal of UDP through NAT

STUN is an industry standard approach for traversal of NAT and the technical details are published as RFC 3489. It requires that your IP phone has access to a STUN server somewhere on the Internet. Your VoIP service provider should be able to give you the address details of their STUN server, but don't despair if they cannot. See the section below that explains how to make your phone use STUN. **A simple explanation of how STUN works**, it sends a number of queries to the specified STUN server. The STUN server carries out a few simple tests to determine things like: Is the IP phone behind a NAT device? What is the external IP address of the NAT device? How tightly does the NAT device enforce rules for blocking inbound UDP connections? Does it make a difference to inbound connection has already been established to that remote address? It then reports the results back to the AXIS device. The AXIS device is now able to use this information to modify the SIP messages it sends when it registers and, if you are lucky, everything will now work perfectly.

STUN does not include an authentication dialogue so generally any phone can use any STUN server. Here are some addresses that might work if you in need of a STUN Server: stun.xten.com, stun.SIPgate.net, stunserver.org

11 Introduction to ICE (from http://www.pjSIP.org)

Interactive Connectivity Establishment (ICE) is the ultimate weapon a client can have in its NAT traversal solution arsenals, as it promises that if there is indeed one path for two clients to communicate, then ICE will find this path. And if there are more than one paths which the clients can communicate, ICE will use the best/most efficient one.

ICE works by combining several protocols (such as STUN and TURN) altogether and offering several candidate paths for the communication, thereby maximising the chance of success, but at the same time also has the capability to prioritize the candidates, so that the more expensive alternative (namely relay) will only be used as the last resort when else fails. ICE negotiation process involves several stages:

- candidate gathering, where the client finds out all the possible addresses that it can use for the communication. It may find three types of candidates: host candidate to represent its physical NICs, server reflexive candidate for the address that has been resolved from STUN, and relay candidate for the address that the client has allocated from a TURN relay.
- prioritizing these candidates. Typically the relay candidate will have the lowest priority to use since it's the most expensive.
- encoding these candidates, sending it to remote peer, and negotiating it with offer-answer.
- pairing the candidates, where it pairs every local candidates with every remote candidates that it receives from the remote peer.
- checking the connectivity for each candidate pairs.

 concluding the result. Since every possible path combinations are checked, if there is a path to communicate ICE will find it.

12 The SIP registration and how it works

It is normal for SIP registrations to be authenticated by a challenge process. The SIP device sends a REGISTER request with minimal credentials and the Registrar server sends back a "401 Unauthorised" response. The device then re-sends the registration request, but this time it adds an "Authorization" header containing a digest of user information and an encrypted version of the password.

If the information matches with a known user, the registration will be accepted and a record is written to the location table.

In case you have or your customer are having trouble to register a AXIS SIP device please have a look at the illustrated steps:

13 Appendix

SIP Error Messages (this is an extract of

<u>https://en.wikipedia.org/wiki/List_of_SIP_response_codes</u>, be aware of that most of the SIP servers do have the possibility to change the codes and that they can have totally different meaning.)

Informational(1xx)

Informational responses are used to indicate call progress. Normally the responses are end to end (except 100 Trying). The main objective of informational responses is to stop retransmission of INVITE requests.

Informational responses include the following responses:

100 Trying

-This special case response is only a hop-by-hop request.

-It is never forwarded and may not contain a message body.

-It is used to avoid the retransmission of INVITE requests.

180 Ringing

-This response is used to indicate that an INVITE has been received by the user agent and alerting is taking place.

181 Call is Being Forwarded

-This response is used to indicate that the call has been forwarded to another endpoint.

-It is sent when the information may be of use to the caller.

-It gives the status of the caller, as a forwarding operation may result in the call taking longer to be answered.

182 Call Queued

-This response is used to indicate that the INVITE has been received and will be processed in a queue.

183 Session Progress

-It indicates that information about the progress of a session may be present in a message body or media stream.

-Unlike a 100 Trying response, a 183 is an end-to-end response and establishes a dialog. -A typical use of this response is to allow a UAC to hear a ringtone, busy tone, or recorded announcement in calls through a gateway into the PSTN.

Success(2xx)

This class of responses is meant for indicating that a request has been accepted. It includes the following responses:

200 OK

-200 OK is used to accept a session invitation. -It indicates a successful completion or receipt of a request.

202 Accepted

-202 Accepted indicates that the UAS has received and understood the request, but that the request may not have been authorized or processed by the server.

-It is commonly used in responses to SUBSCRIBE, REFER methods.

Redirection(3xx)

Generally these class responses are sent by redirect servers in response to INVITE. They are also known as redirect class responses. It includes the following responses:

300 Multiple Choices

-It contains multiple Contact header fields to indicate that the location service has returned multiple possible locations for the SIP URI in the Request-URI.

301 Moved Permanently

-This redirection response contains a Contact header field with the new permanent URI of the called party.

-The address can be saved and used in future INVITE requests.

302 Moved Temporarily

-This redirection response contains a URI that is currently valid but is not permanent. -That is, the location is valid for the duration of the time specified.

305 Use Proxy

-This response contains a URI that points to a proxy server having authoritative information about the calling party.

-This response could be sent by a UAS issuing a proxy for incoming call screening.

380 Alternative Service

-This response returns a URI that indicates the type of service the called party would like.

-For example, a call could be redirected to a voicemail server.

Client Error(4xx)

Client error responses indicate that the request cannot be fulfilled as some errors are identified from the UAC side. The response codes are generally sent by UAS. Upon receiving an error message, the client should resend the request by modifying it based on the response. Discussed below are some of the important client error responses.

400 Bad Request

-It indicates that the request was not understood by the server.

-Request might be missing required header fields such as To, From, Call-ID, or CSeq.

401 Unauthorized

-It indicates that the request requires the user to perform authentication.

-401 Unauthorized is normally sent by a registrar server for REGISTER request.

-The response contains WWW-Authenticate header field which requests for correct credentials from the calling user agent.

-A subsequent REGISTER will trigger from the User Agent with correct credentials.

403 Forbidden

-403 Forbidden is sent when the server has understood the request, found the request to be correctly formulated, but will not service the request.

-This response is not used when authorization is required.

404 Not Found

-404 Not Found indicates that the user identified by the SIP URI in the Request-URI cannot be located by the server or that the user is not currently signed on with the user agent.

405 Method Not Allowed

-It indicates that the server or user agent has received and understood a request but is not willing to fulfil the request.

-Example: A REGISTER request might be sent to a user agent.

-An Allow field must be present to inform the UAC as to what methods are acceptable.

406 Not Acceptable

-This response indicates that the request cannot be processed due to a requirement in the request message.

-The Accept header field in the request did not contain any options supported by the UAS.

407 Proxy Authentication Required

-This request sent by a proxy indicates that the UAC must first authenticate itself with the proxy before the request can be processed.

-The response should contain information about the type of credentials required by the proxy in a Proxy-Authenticate header field.

-The request can be resubmitted with the proper credentials in a Proxy-Authorization header field.

408 Request Timeout

-This response is sent when an Expires header field is present in an INVITE request and the specified time period has passed.

-It could be sent by a forking proxy or a user agent.

-The request can be retried at any time by the UAC.

422 Session Timer Interval Too Small

-The response is used to reject a request containing a Session-Expires header field.

-The minimum allowed interval is indicated in the required Min-SE header field.

-The calling party may retry the request without the Session-Expires header field or with a value less than or equal to the specified minimum.

423 Interval Too Brief

-The response is returned by a registrar that is rejecting a registration request because the requested expiration time on one or more Contacts is too brief.

-The response must contain a Min-Expires header field listing the minimum expiration interval that the registrar will accept.

480 Temporarily Unavailable

-This response indicates that the request has reached the correct destination, but the called party is not available for some reason.

-The response should contain a Retry-After header indicating when the request may be able to be fulfilled.

481 Dialog/Transaction Does Not Exist

-This response indicates that a response referencing an existing call or transaction has been received for which the server has no records or state information.

483 Too Many Hops

-This response indicates that the request has been forwarded the maximum number of times as set by the Max-Forwards header in the request.

-This is indicated by the receipt of a Max-Forward: 0 header in a request.

486 Busy Here

-This indicates the user agent is busy and cannot accept the call.

487 Request Terminated

-This response can be sent by a UA that has received a CANCEL request for a pending INVITE request.

-A 200 OK is sent to acknowledge the CANCEL, and a 487 is sent to cancel the INVITE transaction.

Server Failure (5xx)

This class response is used to indicate that the request cannot be processed because of an error with the server. The server failed to fulfil an apparently valid request. The response may contain a Retry-After header field. The request can be tried at other locations because there are no errors indicated in the request. Some of the important server failure responses are discussed below.

500 Server Internal Error

-500 indicates that the server has experienced some kind of error that is preventing it from processing the request.

-It is one kind of server failure that indicates the client to retry the request again at this server after several seconds.

501 Not Implemented

-It indicates that the server is unable to process the request because it is not supported.

-This response can be used to decline a request containing an unknown method.

502 Bad Gateway

-This response is sent by a proxy that is acting as a gateway to another network.

-It indicates some problem in the other network is preventing the request from being processed.

503 Service Unavailable

-This response indicates that the requested service is temporarily unavailable at that time.

-The request can be retried after a few seconds, or after the expiration of the Retry-After header field.

504 Gateway Timeout

-This response comes when the request failed due to a timeout occurred in the other network to which the gateway connects.

-It is a server error class response because the call is failing due to a failure of the server in accessing resources outside the SIP network.

505 Version Not Supported

-The server denies a request when it comes with a different SIP version number. The denial is indicated in this message.

-Currently SIP version 2.0 is the only version implemented.

513 Message Too Large

-This response is used by a UAS to indicate that the request size was too large for it to process. 580 Preconditions Failure

-This response is used to reject an SDP offer in which required preconditions cannot be met.

Global Error (6xx)

This response class indicates that the server knows that the request will fail wherever it is tried. As a result, the request should not be sent to other locations.

Only a server having definitive knowledge of the user identified by the Request-URI in every possible instance should send a global error class response. Otherwise, a client error class response should be sent.

A Retry-After header field can be used to indicate when the request might be successful. Some of the important responses are discussed below:

600 Busy Everywhere

-This response indicates that the call to the specified Request-URI could be answered in other locations.

603 Decline

-This response could indicate the called party is busy, or simply does not want to accept the call.

604 Does Not Exist Anywhere

-This response is similar to the 404 Not Found response but indicates that the user in the Request-URI cannot be found anywhere.

-This response should only be sent by a server having access to all the information about the user.

606 Not Acceptable

-This response indicates that some aspect of the desired session is not acceptable to the UAS, and as a result, the session cannot be established.

-The response may contain a Warning header field with a numerical code describing exactly what was not acceptable.

-The request can be retried with different media session information.