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Chapter 1. IP Office SIP Extensions

1. IP Office SIP Extensions

IP Office 5.0 and higher supports the use of SIP extension devices with the IP Office system. These can be SIP phones, SIP software clients or traditional analog devices attached to the SIP Analog Telephony Adapter (ATA).

Within the IP Office configuration, SIP extensions are licensed using the **3rd Party IP End-points** license which is also used for non-Avaya H323 IP extensions. The number of SIP extensions supported is subject to available licenses and to the normal extension limits of the IP Office control unit being used.

This document provides notes on registering SIP devices with an IP Office Release 9.0 or higher system. It assumes that you are familiar with IP Office configuration using IP Office Manager, System Status Application and System Monitor.

This document only covers basic registration with the IP Office. Full configuration of the SIP extension device or client software will be covered by the manufacturer's own documentation.



• No NAT

Connection of SIP extension devices from locations where Network Address Translation (NAT) is applied to the connection is not supported. The IP Office does not provide NAT traversal services (for example STUN or TURN) for SIP extension devices.

• Multiple Line SIP Devices

Some SIP devices can support multiple lines or user accounts, each configured separately. If used with an IP Office each SIP line requires a separate IP Office SIP extension, user and license. Note this refers to a SIP device that can handle multiple simultaneous calls itself and not one that is handling multiple calls by holding them on the IP Office/receiving call waiting indication for waiting calls on the IP Office.

• The IP Office is the SIP Registrar and SIP Proxy

In most cases, a SIP extension device is configured with settings for a SIP registrar and a SIP proxy. For SIP devices connecting to an IP Office, the LAN1 or LAN2 IP address on which the SIP registrar is enabled is used for both roles.

Codec Seelction

Unlike H323 IP devices which always support at least one G711 codec, SIP devices do not support a single common audio codec. Therefore it is important to ensure that the IP Office SIP extension codecs configured match a codec for which the SIP device is configured.

• Phone Features

Beyond basic call handling via the IP Office (see the features listed below), the features available will vary between SIP devices and Avaya cannot make any commitments as to which features will or will not work or how features are configured.

- Answer calls.
 Hold.
 - Unsupervised Transfer.
- Voicemail Collect.
- Set Forwarding/DND.

• Hang Up.

Make calls.

-
 - Supervised Transfer.
- Park/Unpark.

1.1 Licensing

SIP Extensions are within the IP Office configuration use **3rd Party IP End-points** licenses. Successful registration consumes one license count.

This license is also used for non-Avaya H323 IP extensions. There must be sufficient licenses for the number of extensions required.

Licence			3rd Party IP Endpoints	→ × × × × × × × ×
Licence Type		Licences		
🐜 Avaya IP endpoints		Licence Key	ZU5W4NLogD0kZQ1X6KL@woqYGyrOvW2c	
L	_	Licence Type	3rd Party IP Endpoints	
		Licence Status	Valid	
		Instances	255	
		Expiry Date	Never	

1.2 Enabling SIP Extension Support

Once the IP Office system has valid 3rd Party IP End-points licenses 10, it can support SIP extensions on its LAN1 and/or LAN2 interfaces.

Note that changing the SIP registrar settings of an IP Office system requires the IP Office system to be rebooted.

- 1. Using IP Office Manager, receive the IP Office system configuration.
- 2. Select System.

LAN1

- 3. Select either the LAN1 or LAN2 tab as required.
- 4. Select the VoIP sub-tab.

LAN Settings VoIP Network	Topology			
H323 Gatekeeper Enable				•
V Auto-create Extn	Auto-create User		H323 Remote E	xtn Enable
SIP Trunks Enable				
SIP Registrar Enable				=
Auto-create Extn/User			SIP Remote Extr	n Enable
Domain Name				
	UDP UDP	IDP Port 5060	Remote UDP Port 50	i60
Layer 4 Protocol	📝 ТСР	CP Port 5060	Remote TCP Port 50	60
	TLS T	LS Port 5061	Remote TLS Port 50	61
Challenge Expiry Time (secs)	10			
RTP				
Port Number Range				
Minimum	49152 🚔 Maximu	m 53246 🚔		
Port Number Range (NAT)				
Minimum	49152 🚔 Maximu	m 53246		
•		11		-

- SIP Registrar Enable Check that SIP Registrar Enable is selected.
- **Domain Name:** *Default = Blank*

This is the local SIP registrar domain name that will be needed by SIP devices in order to register with the IP Office. If this field is left blank, registration is against the LAN IP address. The examples in this documentation all use registration against the LAN IP address.

- Layer 4 Protocol: Default = Both TCP & UDP The transport protocol for SIP traffic between the IP Office and SIP extension devices. Both TCP and/or UDP can be used.
- **TCP Port:** *Default* = 5060 The SIP port if using TCP. The default is 5060.
- **UDP Port:** *Default* = 5060 The SIP port if using UDP. The default is 5060.
- **Challenge Expiry Time (sec):** *Default = 10* The challenge expiry time is used during SIP extension registration. When a device registers, the IP Office SIP Registrar will send a challenge back to the device and waits for an appropriate response. If the response is not received within this timeout the registration is failed.
- Auto-create Extn/User: Default = On If this option is selected, the IP Office will automatically create user and SIP extension entries in its configuration based on SIP extension registration. If this method is being used for installation, it is important to check that the settings created match the SIP device. It is also important to deselect this option after installation of the SIP extension devices.
- 5. If you have made any changes, send the configuration back to the IP Office.

1.3 SIP Extension Settings

SIP extensions can be created manually using $\stackrel{\text{def}}{=}$ | **SIP Extension** or <u>automatically created</u> $\stackrel{\text{def}}{=}$ during SIP device registration. Even if auto-created, the extension settings created in the IP Office configuration should be checked after installation.

This section looks just at the key configuration settings that affect SIP extension devices. For full details of all the fields shown refer to the IP Office Manager Manual.

1. Sele	ect 🥙	Exte	ensio	ons and	d locate the SIP extension. Select the Extn tab.	
Extn	Vol	P T38	3 Fax			
Exte	ension Id				8000	
Bas	e Extensi	on				
Cal	ler Displa	ау Туре			On v	
Res	et Volum	ne After (Calls			
Dev	vice Type	2		a	Unknown SIP device	
Loc	ation				Automatic 🔹	
Mo	dule				0	
Por	rt				0	
For	ce Autho	orisation				

Base Extension

- This should match the **Extension** setting of the SIP user added to the IP Office configuration.
- Force Authorization: Default = On
 - If enabled, SIP devices are required to register with the IP Office system using the **Name** and **Login** Code configured for the user within the IP Office configuration.

2. Select the VoIP tab.

Extn VoIP T38 Fa	IX	
IP Address	0 . 0 . 0 . 0	 VoIP Silence Suppression Local Hold Music
Codec Selection	System Default	 Allow Direct Media Path
	Unused >>> G.711 ALAW 64K G.711 ULAW 64K G.729(a) 8K CS-ACELP G.723.1 6K3 MP-MLQ >>>	 Re-invite Supported Codec Lockdown
Reserve License	None	•
Fax Transport Support	None	•
TDM->IP Gain	Default	•
IP->TDM Gain	Default	•
DTMF Support	RFC2833	•

Codec Selection

If the **Codec Selection** is left set to **System Default**, the extension will use the system codec preferences. In most cases this is preferred and any changes required should be made at the system level to ensure consistency for all IP trunks and extensions. However, if required, the **Codec Selection** of each individual trunk and extension can be adjusted to differ from the system defaults. See below.

Codec Lockdown

In response to a SIP offer with a list of codecs supported, some SIP user agents supply a SDP answer that also lists multiple codecs. This means that the user agent may switch to any of those codecs during the session without further negotiation. The system does not support multiple concurrent codecs for a session, so loss of speech path will occur if the codec is changed during the session. If **Codec Lockdown** is enabled, when the system receives an SDP answer with more than one codec from the list of offered codecs, it sends an extra re-INVITE using just a single codec from the list and resubmits a new SDP offer with just the single chosen codec.

• DTMF Support

This can be set to one of the two common methods used by SIP devices; *RFC2833* or *Inband*. The selection should be set to match the method used by the SIP device. However, if the method is not known or can vary on a per call basis, deselecting **Allow Direct Media Path** allows a VCM channel to be used for DTMF support when necessary.

Local Hold Music

Select this option if the SIP device supports its own hold music source.

• Re-invite Supported

If the SIP device is able to receive REINVITE messages select this option.

Reserve License:

Each non-Avaya IP phones requires an **3rd Party IP Endpoint** license. Normally the available licenses are issued in the order that devices register. This option allows an extension to be pre-licensed before the device has registered by selecting the option **Reserve 3rd party IP endpoint licence**.

Custom Codec Selection

1. Using IP Office Manager, receive the system's configuration.

- 2. To display the extension's settings, click **Extension** in the left-hand panel.
- 3. Select the **VoIP** tab.
- 4. Change the **Codec Selection** to **Custom**.
- 5. The **Unsed** and **Selected** lists can be used to select which codecs the device uses and the order of preference.
- 6. Save the configuration changes back to the system.

1.4 SIP User Settings

2

1

SIP users can be created manually using $\stackrel{\text{disc}}{=}$ | User or <u>automatically created</u> 15 during SIP device registration. Even if auto-created, the user settings created in the IP Office configuration should be checked during installation.

This section looks just at the key configuration settings that affect SIP extension devices. For full details of all the fields shown refer to the IP Office Manager Manual.

Select	📕 Use	r and I	ocate the	e SIP ex	tensi	on user.	Select the	e User	tab.		
Menu P	rogramming	Mobility	Group Me	mbership	Annou	ncements	Personal Direc	tory			
User	Voicemail	DND	ShortCodes	Source No	umbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming	
Name			Extn400	00							^
Passwo	ord										
Confirr	n Password										
Accour	nt Status		Enable	d					•	•	
Full Na	me										
Extensi	on		4000								
Email A	Address										=
Locale										•	
Priority	,		5							•	
System	Phone Righ	ts	None							•	
Profile			Basic U	Jser						•	
			Reco	eptionist							
			Ena	ble Softpho	one						

• Name

If the SIP extension is set to **Force Authorization** (the default), this field is used as the **Authorization Name** that must be set in the SIP device's configuration.

• Extension

This should match the SIP ID of the SIP device and the Base Extension setting of the SIP extension in the IP Office configuration.

2. Select the **Telephony | Call Settings** tab.

• Call Waiting On

Most SIP devices require this setting to be enabled in order to allow features such as transferring calls. This requirement no longer applies for IP Office Release 9.0 and higher.

3. Select the Telephony | Supervisor Settings tab.

Menu	Menu Programming Mobility Group Membership Announcements Personal Directory									
User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming	
Call S	ettings Sup	ervisor Se	ttings Multi	line Options Call	Log TUI					
Logi	n Code						Force l	Login		
Logi	n Idle Period	(secs)					Force A	Account Code		
Mon	itor Group		<none></none>			•				
Cove	erage Group		<none></none>			-	Incom	ing Call Barring		
Statu	ıs on No-Ans	wer	Logged Or	ı (No change)		-	Outgoing Call Barring			
Res	et Longest Id	e Time –					Inhibit	Off-Switch Forwar	d/Transfer	
	All Calls						Can In	trude		
	Stornal Incon	aina				l.	Z Canno	t be Intruded		
	External Incor	ining					Can Tr	ace Calls		
							CCRA	gent		

• Login Code

If the SIP extension is set to **Force Authorization** (the default), this field is used as the **Authorization Password** that must be set in the SIP device's configuration.

1.5 Allowing SIP Extn/User Auto Creation

The IP Office system can be set to automatically create extension and user entries in its own configuration as each SIP device registers with the system. It can speed up installation to enable this setting when installing several devices and then disable the setting once the installation has been completed.

Note that changing the SIP registrar settings of an IP Office system requires the IP Office system to be rebooted.

- 1. Using IP Office Manager, receive the IP Office system configuration.
- 2. Select System.
- 3. Select either the LAN1 or LAN2 tab as required.
- 4. Select the **VoIP** sub-tab.

H323 Gatekeeper Enable					- LU222 D		
🖊 Auto-create Extn	Auto-create	User			H323 Remot	e Extn Enable	
SIP Trunks Enable							
I SIP Registrar Enable							
Auto-create Extn/User					SIP Remote	Extn Enable	
Domain Name							
	UDP	UDP Port	5060	* *	Remote UDP Port	5060	A V
ayer 4 Protocol	👿 ТСР	TCP Port	5060	*	Remote TCP Port	5060	*
	TLS	TLS Port	5061	* *	Remote TLS Port	5061	*
Challenge Expiry Time (secs)	10						
{TP							
Port Number Range							
Minimum 4	9152 🚔	Maximum	53246	*			
Port Number Range (NAT)							
Minimum 4	9152 🌲	Maximum	53246	* *			

5. Change the **Auto-create Extn/User** settings to the state required.

6. Send the configuration back to the IP Office.

1.6 System Monitor

The status of the SIP extensions in the IP Office configuration can be viewed using the IP Office System Monitor application. Select **Status | SIP Phone Status** to display the SIP extension list.

🗐 SIPPhone	Status				
Total Configu	ıred: 1		Waiting 1 :	secs for up	date
Total Registe	ered: 1		Registered Status		
Extn Num	IP Address	Transport	User Agent	SIP 0	Status
334	192.168.42.203	UDP	X-Lite release 1103d stamp 53117	RM	SIP: Registered
L					
-					
Display Options Show All C Registered C UnRegistered Print Cancel					

Chapter 2. SIP Device Configuration

2. SIP Device Configuration

This section gives examples of the installation settings used with a variety of SIP devices tested with IP Office. These are only the basic details for registration with an IP Office system. Full installation and configuration, for example assigning device IP addresses, is covered in the device or software manufacturer's own documentation.

The devices covered are:

- <u>Astra 9133i</u> 19
 - <u>Avaya A10 ATA</u> 21
 - <u>CounterPath Eyebeam/X-Lite Softphones</u>
 - <u>Grandstream GXP 2000, GXP 2020</u> 28
 - Innovaphone IP22, IP24, IP28 30
 - Nokia S60 v3 SIP Client 33
 - Patton Micro ATA 34
 - Polycom Soundpoint 35

The general process for connection to the IP Office can be done in two ways. Either allowing the IP Office to auto-create extension and user entries when a SIP device registers or manually creating those entries and then registering the SIP device. The steps are summarized below.

Using Auto Create	Using Manual Configuration			
1. Add and check 3rd Party IP End-points licenses.	1. Add and check 3rd Party IP End-points licenses.			
2. Check the SIP Registrar settings.	2. Check the SIP Registrar settings.			
3. Enable Auto-Create Extn/User.	3. Add SIP Extension settings to the IP Office			
4. Attach and configure the SIP device.	configuration.			
5. Modify the IP Office user and extension settings.	4. Add SIP User settings to the IP Office configuration.			
6. Disable Auto-Create Extn/User.	5. Attach and configure the SIP device.			

2.1 AAstra 9133i SIP

AAstra produce a range of SIP phone devices. The example used here is a 9133i phone.

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Browse to the IP address of the phone.
 - 2. Login. The default user name and password are *Admin* and *22222*.
 - 3. Click on **Line 1** or the line that will be used for IP Office calls.

Status	Configuration Line 1		
System Information	Basic SIP Authentication Settings		
Operation	Screen Name	SIP334	
User Password Programmable Keys	Phone Number	334	User User Extension Extn Base Extension
Directory	Caller ID		
Reset	Authentication Name	Extn334	User User Name
Basic Settings Preferences	Password	••••	User Telephony Call Settings Login Co
Call Forward	BLA Number		
Advanced Settings Network	Line Mode	Generic 👻	
Global SIP	Basic SIP Network Settings		
Line 1	Proxy Server	192.168.42.1 —	
Line 3	Proxy Port	5060	
Line 4	Backup Proxy Server	0.0.0.0	1
Line 5 Line 6	Backup Proxy Port	0	
Line 7	Outbound Proxy Server	0.0.0.0	_
Line 8 Line 9	Outbound Proxy Port	0]
Action URI	Registrar Server	192.168.42.1 —	System LAN LAN Settings IP Address
Configuration Server	Registrar Port	5060]
Troubleshooting	Backup Registrar Server	0.0.0.0]
	Backup Registrar Port	0	
	Registration Period	0]

4. Enter the values to match the IP Office configuration settings as indicated above.

B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.

 $\ensuremath{\mathsf{C.Make}}$ test calls from and to the SIP device.

D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

Key Programming

The AAstra phones have a range of programmable keys that can be used to activate phone functions or to speed dial numbers. To use these to activate IP Office functions, the key must be configured to speed dial an IP Office short code.

1. Login to the phone and select **Programmable Keys**.

Status	Programmable Keys Configuration				
System Information	Key	Туре	Value	Line	
Operation User Password	Hard Key 1:	speeddial 😽	*17	1 🗸	
Programmable Keys	Hard Key 2:	do not disturb 🔽		1 ~	
Directory Reset	Hard Key 3:	speeddial 🗸	*37*1#	1 🗸	
Basic Settings	Hard Key 4:	speeddial 💉	*38*1#	1 🗸	
Preferences Call Forward	Hard Key 5:	speeddial 🗸	*30	1 🗸	
Advanced Settings	Hard Key 6:	none 🗸		1 ~	
Network Global SIP	Hard Key 7:	none 💌		1 ~	
Line 1 Line 2 Line 3	BLF List URI:	JS			

- 2. In the example above:
 - The first button on the side of the phone has been made a voicemail button by setting it to speed dial the default IP Office short code for voicemail access.
 - The second button is the phones own **Do Not Disturb** function. Therefore when used it will not be reflected in the users DND status on the IP Office system.
 - The third and fourth buttons are set to use the IP Office default short codes for parking and unparking a call from park slot 1 on the IP Office system. The phones own **Park** function does not work with IP Office systems.
 - The fifth button is set to the IP Office default short code for Call Pickup Any.

Notes

- Appearances L1-L3 do function and auto hold will work when switching lines
- Speaker button works but release is done thru the hang up button, you cannot end a speaker call by hitting speaker button again.
- To transfer use the **xfer** key. Press **xfer**, enter the transfer destination and press **xfer** again.
- To park a call, use the **xfer** key to transfer the call to the short code for parking a call. For example in this case transfer to *38*1#. To unpark, dial the short code created for that function.
- The **Conf** button can be used to conference calls. With the first party connected, press **Conf**, dial the second party and when answered press **Conf** again.

2.2 Avaya A10 ATA

The Avaya A10 Analog Telephone Adapter provides 4 Phone/FXS ports on its rear plus a LAN port. It can be used to connect analog phone devices to the IP Office via the LAN, with the extensions appearing in the IP Office configuration as SIP extensions.

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Browse to the IP address of the A10.
 - 2. Enter the administrator name and password. The defaults are *nimdbg* and *54321*.
 - 3. Select **Telephony** and then **SIP**.

Hama	132,100,1,17 Telephony / Sir								
Import/Export	Gateways	Interfaces	Profiles						
	Name		Domain	Default-Server	Registration	Authentication	Binding	State	
Network	sip			1	To /	(none)	eth0	Enabled	X
IP/DNS NAT/NAPT									d,

4. Select the **Gateways** tab and click on **sip**. 192.168.1.1 / Telephony / SIP / Gateway sip

Home		
Import/Export	Configuration Status	
	IP Interface	💌 eth0 💌 🔿
etwork	SIP Gateway	Enabled 💌
NAT/NAPT	Local Call Signaling Port	5060
ACL QoS	Call Signaling Traffic Class	local-default 💌
DynDNS DHCP Server	INVITE Transaction Timeout	32 seconds
WAN	Non-INVITE Transaction Timeout	32 seconds
elephony Call-Router H.323	Transport Protocols	▼ TCP
SIP VolP Profiles	Penalty Box	600 seconds Time for which a non-responsive destination should stay in the penalty box, i.e. should not be contacted anymore
Tone Profiles PSTN Profiles		Apply
orts	Services	
Ethernet FXS	default	×
arious		

5. Click on default in the **Services** section. Select the **Configuration** tab.

University	192.168.1.1 / Telephony / SIP / Gateway sip / Service default				
Import/Export	Configuration Registration a	Configuration Registration and Authentication			
	Domain				
Network					
IP/DNS NAT/NAPT ACI	Default-Server (Outbound Proxy)	Set always the actual Registrar as Default Server			
QoS	Force Keep-Alives	I▼ 3600 seconds			
DHCP Server	Call Transfer	Version: 5			
WAN	Session Timer	Version: 8			
Telephony Call-Router	Create new session after redirect				
H.323 SIP VoIP Profiles	Alternate Contact Address	C Detect NAT Address C User Defined IP Address			
Tone Profiles PSTN Profiles	SIP Profile	default 🔽 O			
Ports	VoIP Profile	default 🔽 오			
Ethernet		Apply			

- Ensure that the **Domain** field is empty and the check box not selected.
- Enable the check box for **Default-Server (Outbound Proxy)** and select **Set always the actual Registrar as Default Server**.
- Click Apply√.

6. Select the Reg	istration and	Authe	ntication ta	b.					
Home Import/Export	Configuration	/ SIP / Gat	teway sip / Servic and Authentication	e default	—System LAN	I LAN Settings IP Add	iress		
Network IP/DNS	Registrar		Ignore redirection Register to redire	of Registrar 192.1	68.42.1	Host 5060 Host	Port 🦳 Register v Port	ria Default-Ser	rver
NAT/NAPT ACL QoS DynDNS DHCP Server	Registration Lifetime	300	seconds					Арріу	~
WAN	Users To Register								
Telephony	User Name	Register	Display Name	Phone Context	Authenticate	Authentication Name	Password	Default	
Call-Router	338	register	SIP 338	SIP	authenticate	Extn338	******	default	×
SIP		◄					•••••		Č,
VoIP Profiles Tone Profiles	User User I Extn Base E	Extension dension		User User	User Name - Telephony C] all Settings Login Cod	ie		

- Enable the Registrar checkbox. Select **Ignore redirection of Registrar** and enter the IP address and SIP port of the IP Office LAN on which the SIP registrar is enabled. Click **Apply**.
- 7. In the **Users To Register** section, create a user matching the IP Office SIP extension and user. Enter the settings and click on $\overrightarrow{\Box}$.
- 8. Select Call-Router. Select Interfaces and then FXS. 192.168.1.1 / Telephony / Call-Router

U.s. es a	10211001111110	repriority / call-field					
Import/Export	Interfaces	Routing Tables	Functions	Services	Configuration	Active Calls	Status
	FXS H.323	3 SIP					
Network	Name			Bound Port	Routing Des	tination	
IP/DNS NAT/NAPT	fxs-0			fxs00	to-sip (Tabl	e)	×
ACL	fxs-1			fxs 0 1	to-sip (Table	e)	×
QoS	fxs-2			fxs02	to-sip (Tabl	e)	×
DynDNS	fxs-3			fxs03	to-sip (Table	e)	×
WAN							Ť

9. Click on fxs-0.

	192.168.1.1 / Telephony /	Call-Router / FXS Interface fxs-0
Import/Export	Configuration Statu	IS
Network IP/DNS NAT/NAPT	Call-Routing Destination	C Interface (none) C Table to-sip C Service (none)
ACL QoS	Precall Service	(none)
DynDNS DHCP Server	CID Presentation	(none)
WAN	Subscriber Number	338
Telephony	Call Hold	
Call-Router H.323	Call Waiting	
SIP	Call Transfer	
VolP Profiles Tone Profiles PSTN Profiles	Additional Call Offering	
	PSTN Profile	default 💌
Ports	Tone Profile	US 🔻
Ethernet FXS		Apply

- Enable the **Call-Routing Destination** checkbox. Select **Table** and in the adjacent drop down list select *to-sip*.
- Enable the **Subscriber Number** checkbox and enter the IP Office extension number for the SIP extension and user.
- Click **Apply**√.

10.Click on the 🗘 a	rrow icon after to-sip . 192.168.1.1 / Telephony / Call-Route	er / Routing Table to-sip				
Home Import/Export	Configuration	Configuration				
Network	Looks Up For called-e164 Of	Destination	Execute Function (Optional)			
IP/DNS	т	sip (SIP Interface)		\mathbf{X}		
NAT/NAPT ACL QoS	called-e164 value or default	O Interface (none) ▼ O Table (none) ▼	Optional function to execute	~*		
DynDNS DHCP Server WAN	(To change an entry, enter the value of an existing entry)	C Service (none)	(none) 💌	0		

• Ensure that the table contains T with the destination sip (SIP Interface).

11.Select Call-Router again and then select the Routing Tables tab.

11 - march	102.	122100.1117 Telephony / call-Roater						
Import/Export	In	nterfaces	Routing Tables	Functions	Services	Configuration	Active Calls	Status
	R	Routing Tables						
Network	Na	ame			Looks up f	or		
IP/DNS	fr	rom-sip			called-e164			×
ACL	to	to-sip		called-e164			×	
QoS	LΓ				called-e1	64 💌]	Ť
DVDDNS					·		-	

12.Select *from-sip*. 192.168.1.1 / Telephony / Call-Router / Routing Table *from-sip*

Import/Export	Configuration			
Network	Looks Up For called-e164 Of	Destination	Execute Function (Optional)	
IP/DNS NAT/NAPT ACL QoS DynDNS DHCP Server	called-e164 value or default 338 (To change an entry, enter the value of an existing entry)	Interface Table Service (none)	Optional function to execute	Ť

- Enter the IP Office SIP extension number.
- For the **Destination** select **Interface** and select the matching fxs port for that extension number.
- Click 💣.
- 13.Repeat for any other SIP extensions on the unit. 192.168.1.1 / Telephony / Call-Router / Routing Table from-sip

Home Import/Export	Configuration			
Network	Looks Up For called-e164 Of	Destination	Execute Function (Optional)	
IP/DNS	338	fxs-0 (FXS Interface)		\mathbf{X}
NAT/NAPT ACL QoS DynDNS DHCP Server WAN	Called-e164 value or default (To change an entry, enter the value of an existing entry)	○ Interface (none) ▼ ○ Table (none) ▼ ○ Service (none) ▼ ○ none	Optional function to execute (none)	Ť

14. Click Save to save the settings so that they will still apply after the unit is restarted. 192.168.1.1 / Save

Home						
Import/Export	Save Configuration					
	You are going to save the modified configuration persistently.					
Network	This is needed to retain the current configuration beyond the next reload. Are you sure you want to write the current running-config to the startup-config?					
IP/DNS						
NAT/NAPT ACL	Save	Cancel				

- B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.
- C.Make test calls from and to the SIP device.
- D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

Notes

- When calling from an phone attached to an FXS port, there is a delay of approximately 5 seconds while the unit wait for dialing to be completed before it routes the dialed digits to the IP Office. To avoid this delay dial # after dialing the digits.
- The G723 Codec should not be used with the Avaya A10 ATA. However that codec is not enabled by default.

Home	192.168.1.1	relephony / vo	ip profiles / profil	e derault				× × *
Import/Export	Voice	Fax Modem	Dejitter Buffer	Status				
	Voice Co	decs						
Home Import/Export Voice Fax Modem Dejitter Buffer Status Network Position Codec Rx Length [ms] Tx Length [ms] Silence Suppression IP/DNS NAT/NAPT 3 0 1 9711ulaw64k 20 20 © default © yes n ACL 3 5 2 9711alaw64k 20 20 © default © yes n QoS DynDNS 3 g729 20 20 © default © yes<								
IP/DNS NAT/NAPT		g711ulaw64k	20		20	⊙ default C yes C no	~	\boldsymbol{X}
ACL	2	g711alaw64k	20		20	⊙ default O yes O no	✓	×
DynDNS	□ □ 3	g729	20		20	⊙ default O yes O no	✓	×
DHCP Server WAN		transparent				⊙ default C yes C no		ď
Telephony Call-Router	Additiona	I Voice Paramet	ers					
H.323 SID	Default Sil	ence Suppression			📕 If not sp	ecified by the codec		
VoIP Profiles	Highpass	Filter			Voice in	put filter for A/D conversion		
Home Import/Export Voice Fax Modem Dejitter Buffer Status Network Position Codec Rx Length [ms] Tx Length [ms] Silence S IP/DNS IAT/INAPT I g11ulaw64k 20 20 Import/Export ACL Import/Export Import/Export Import/Export Import/Export Import/Export JynDNS Import/Export Import/Export Import/Export Import/Export Import/Export ACL Import/Export Import/Export Import/Export Import/Emport Import/Emport ACL Import/Export Import/Emport Import/Emport Import/Emport Import/Emport DynDins Import Import Import Import Import Import Call-Router Highpass Filter </td <td>uput filter for D/A conversion</td> <td></td> <td></td>	uput filter for D/A conversion							
NAT/NAPT ACL 1 g/11ulaw64k 20 20 C default C yes C no QoS DynDNS DHCP Server 3 g729 20 20 C default C yes C no DHCP Server WAN T transparent I transparent C default C yes C no Call-Router H.323 Default Silence Suppression I f not specified by the codec Highpass Filter Voice input filter for A/D conversion Ports DTMF Relay Voice ouput filter for D/A conversion DTMF Relay RTP Payload Type For Tone Events (NTE) 101 KTP Payload Type For Signaling Events (NSE) 100								
Ethernet	RTP Paylo	ad Type For Tone	Events (NTE)		101			
FXS	RTP Paylo	ad Type For Signa	ling Events (NSE)		100		uppression It ○ yes ○ no ✓ × It ○ yes ○ no ✓ × It ○ yes ○ no ✓ × It ○ yes ○ no ✓ × Ht ○ yes ○ ∧ no ✓	
Various					Inclusion			
System	RIP Iram	c class			plocal-voice			
Time							Appl	y 🖌

2.3 CounterPath eyeBeam/X-Lite

CounterPath produce a range of VoIP products. X-Lite is a simple SIP client application that can be used as a PC softphone test SIP operation. X-Lite can be downloaded from http://www.counterpath.com/.

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Either enable the IP Office to allow <u>automatic creation</u> 15 based on SIP phone registration or manually add the SIP extension and user details to the IP Office configuration.
 - 2. Start the X-Lite SIP client application.

4

3. Click on the down arrow icon and select SIP Account Settings....

Click on Add			
Properties of Account 1		×	
Account Voicemail Topolog	y Presence Advanced		
User Details			
Display Name	SIPMe		
User name	334		User User Extension Extn Base Extension
Password	••••		User Telephony Call Settings Login Code
Authorization user name	Extn334		User User Name
Domain	192.168.42.1		System LAN LAN Settings IP Address
Domain Proxy			
🔽 Register with domain and	receive incoming calls		
Send outbound via:			
💿 domain			
C proxy Address			
Dialing plan			
	OK Cancel	Apply	

- 5. Set the fields to match the IP Office configuration settings are indicated above.
- 6. In the **Domain Proxy** section enable **Register with domain and receive incoming calls** and select **domain**.
- 7. When completed click on **OK**.

Enabled	Acct #	Domain	Username	Display Name	<u>A</u> dd
	1	192.168.42.1 (default)	334	SIPMe	
					<u>R</u> emove
					Properties
					Make <u>D</u> efault

8. Ensure the the account is **Enabled**.

- 9. Click **Close**. The X-Lite client will now attempt to register with the IP Office. The success or failure of that process will be displayed by the client.
- 10.If left with its default configuration, then on calls from an IP Office DS extension to the X-Lite client, the speech from the client will not be heard. The solution is to either configure the client with a single audio codec 27 or to perform the following process.

a. Dial *****7469** and select call. The **Advanced Options** menu is displayed.

Advanced Options

Advanced Options	×
Filter: Apply Filter	Clear Filter
Option Name	Value 🔺
audio:aec:manual_offset	0
audio:agc:desired_level	1500
audio:concealment:enabled	1
audio:headset:_section_desc	0
audio:headset:aec_enabled	1
audio:headset:audio_in_agc_enabled	1
audio:headset;audio_in_device	(default wave in)
audio:panic:increase_amount_if_below_in_milliseconds	10 🗸
•	

b. Enter *honor* in the filter field and click **Apply Filter**.

Advanced Options			×
Filter: honor	Apply Filter	Clear Filter	
Option Name		Value	
system:network:honor_first_codec		1	

- c. Set the value for **system:network:honor_first_codec** to **1**.
- d. Click on the \mathbf{X} icon to close the menu.

Advanced Option	าร	2	4
Save changes?			
Yes	No	Cancel	

- e. Click on Yes to save the change.
- B. If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.
- C.Make test calls from and to the SIP device.
- D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

Codec Selection

If the X-Lite client is left configured to support multiple audio codecs, then on calls to the extension there will be no return speech from the client. This can be resolved by configuring the client to only support a single audio codec, matching one of the codecs configured for the IP Office SIP extension.

- 1. Click on the down arrow icon and select **Options**.
- 2. Click on **Advanced** and then on **Audio Codecs**.

Options		×
General	Disabled codecs:	Enabled codecs:
Advanced	BroadVoice-32 BroadVoice-32 FEC DVI4 DVI4 Wideband G711 aLaw GSM	-> G711 uLaw
Video Codecs	LIGC L16 PCM Wideband Speex Speex FEC Speex Wideband Speex Wideband FEC	
Network	Codec Properties	G711 aLaw
	Bitrate range (bps):	80000 - 80000
Quality of Service	Fidelity:	Narrowband (8000)
	Best Quality (PESQ):	0.0 4.5
Diagnostics		Apply Revert OK Cancel

3. Ensure that the **Enabled codecs** column contains just a single codec. That codec must be one supported by the IP Office extension configuration for the SIP extension.

4. Click OK.

2.4 Grandstream

Grandstream devices can support multiple user accounts for the same or different SIP provider accounts. The configured accounts are displayed on the phone display and the user can select which account is used when making a call. For IP Office operation, each account can represent a different IP Office SIP extension and user.

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Browse to the IP address of the phone. By default the phone uses DHCP and displays its IP address on the display. Enter the password (the default is **admin**).
 - 2. Click **Login**. Select **Account 1** or the account that you want to use for IP Office connection.

Gra	ndstream Device Configuration
STATUS BASIC SETTINGS ADVANCED SE	TTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 ACCOUNT 5 ACCOUNT 6
Account Active:	O No O Yes
Account Name:	Brad 4142
SIP Server:	192.168.42.1 System LAN LAN Settings IP Address
Outbound Proxy:	192.168.42.1
SIP User ID:	4142 User User Extension
Authenticate ID:	Extn Base Extension
Authenticate Password:	User Telephony Call Settings ogin Code
Name:	Brad SiPhone
local SIP port:	5060 (default 5060)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP T1 Timeout:	1 sec 💌
SIP T2 Interval:	4 sec 💌
SIP Transport:	• UDP • TCP
Use RFC3581 Symmetric Routing:	• No C Yes
NAT Traversal (STUN): SUBSCRIPE for MUL	No O No, but send keep-alive O Yes
PUBLISH for Presence	O No O Ves
Proxy-Require:	
Voice Mail UserID:	*17 (UserID for voice mail system)
	staire 1. G 7290/B staire 5. G 726-32
Preferred Vocoder:	choice 2: PCMA choice 6: iLBC
(in listed order)	choice 3: G.723.1 Choice 7: G.722 (wide band)
	choice 4: PCMU 🔽 choice 8: GSM 🔽
SRTP Mode:	Disabled O Enabled but not forced O Enabled and forced
eventlist BLF URI:	
Special Feature:	Standard 💌
	Undata Cancol Bohaat
All R	lights Reserved Grandstream Networks, Inc. 2004-2008

3. Set the fields indicated above to match those required for the IP Office system.

4.	Click on Update.
	Grandstream Device Configuration
	STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 ACCOUNT 5 ACCOUNT 6
	Your configuration changes have been saved.
	They will take effect on next reboot.
	Reboot
	All Rights Reserved Grandstream Networks, Inc. 2004-2008
5.	Click on Reboot . The phone may take up to 1 minute to reboot.
	Grandstream Device Configuration
	The device is rebooting now
	You may relogin by clicking on the link below in 30 seconds.
	Click to relogin
	All Rights Reserved Grandstream Networks, Inc. 2004-2008

B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.

 $\ensuremath{\mathsf{C.Make}}$ test calls from and to the SIP device.

D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

2.5 Innovaphone IP22, IP24, IP28

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Browse to the IP address of the unit.

Configuration	Info	Admin	License	Update	NTP	Sync	HTTP-Server	HTTP-Client	Logging	SNMP	Telnet	Certificates
General		-										
IP	uration Info Admin License U ral Version 7.00 hotfix3 IP28 SerialNo 00-90-33-21-01-7 DRAM 16 MB FLASH 8 MB Coder 8 Channels of G: Sync - SNTP Server 135.64.181.220 Time 05.06.2009 07:13 Uptime 17d 11h 37m 2	28[09-7030 1.7479a)	U.11], B	ootcode	09-7030011], Har	dware[402]						
ETH0	DRAM	1	6 MB	-7 u (38)								
LDAP	FLAS	H 8	MB									
TEL1	Coder	8	Channels of	G.711,G.72	26,G.729	9						
TEL2	Sync	-										
TEL3	SNTP	Server 1	35.64.181.22	0								
TEL4	lime Uptim	е 1	5.06.2009.07. 7d 11h 37m	:13 29s								
TELE												

- 2. In the left hand column select **GATEWAY**.
- 3. You will be prompted to login. The default user name is **admin**. The default password is **ip22**, **ip24** or **ip28** depending on the unit type.

onfiguration	General I	nterfaces	SIP	GK	Routes	CDR0	CDR1	Calls	ê	admin	Н
eneral		_									
	Call Logging										
HO	Route Logging	g 📃									
AP	Billing CDRs (only 📃									
11	Logging Filter	(GW:Nr)		:							
12	Licenses -										
12	Name Cou	nt Usage									
.1.3		0.1									
:L4		Cancel									

4. Select Interfaces.

Configuration	General	Interfaces	SIP GK	Routes	CDR0	CDR1	Calls	i	admin	Help
General	1.1.1	CONUS					- De -l-ttl-			~
IP	птепасе	CGPN-IN	LUPN-IN CG	PN-Out CD	PN-Out S	tate Alla	s Registratio	n		
ETH0	TEL1	+			U	p				
LDAP	TEL2	+			0	р				
TEL1	TELA	+				p				
TEL2	TEL4	- +				p p				
TEL3	TEL 6	+			U	n D				
TEL4	TEL7	+			U	p				
TEL5	TEL8	+			U	р				
TEL6	TEST	+								
TEL7	TONE	+								
TEL8	HTTP	+								
Administration	ECHO	+								
Gateway										

5. Select **TEL1** in the **Interfaces** page.

Name
Disable 🗌
Tones EUROPE-PBX 💌
Interface Maps Manual 🛛
Internal Registration
Protocol None 💌
Feature Codes Support 🔲 (with Feature Codes)
Dynamic Group
Direct Dial
Locked White List
OK Cancel Apply Delete Help

6.	In the Protocol extension and us	drop dov ser.	wn list select S	IP . Enter th	e details as in	dicated below to match y	your IP Office SIP
	Name	SIP4420)				
	Disable						
	Tones	UK	*				
	Interface Maps	Manual		*			
	Internal Regist	ration -					
	Protocol		SIP 🔽				ttings LIP Address
	Server Address		135.64.181.22	20	(primary)	0,0000011210121000	
	Server Address				(secondary)	User User Extension Extn Base Extension	n
	ID@Domain		4420		@ 135.64.1	81.220	
	Username		SIP4420]		
	Password		••••		Retype ••••	••••	
	Feature Codes	Support	(with Feat	ure Codes)		User Telephony Call	Settings Login Code
	Dynamic Group)]		
	Direct Dial]		
	Locked White L	list]		
	Media Propert	ies					
	General Coder	Preferen	ce G729A 💉	 Framesiz 	e [ms] 30	Silence Compressi	on 📃 Exclusive 📃
	Local Network	Coder	G711A	 Framesiz 	e [ms] 30	Silence Compressi	on 📃
	Enable T.38 🔽] Enabl	e SRTP 📃	No DTMF De	etection 📃	MOH Mode 📃	
	ОКС	ancel	Apply	Delete	Help		

7. <u>Click</u> **OK**.

Configuration	General Interfaces SIP GK Routes CDR0 CDR1 Calls	admin Help
General		
IP	Interface CGPN-In CDPN-In CGPN-Out CDPN-Out State Alias Registration	
ETH0	TEL1 SIP4420 + Up :4420 → 135.64.181.220	
LDAP	TEL2 SIP4421 + Up	
TEL 1	TEL3 SIP4422 + Up	
TEL 2	TEL4 SIP4423 + Up	
TEL 3	TEL5 SIP4424 + Up	
TELA	TEL6 SIP4425 + Up	
TEL4	TEL7 SIP4426 + Up	
TEL5	TEL8 SIP4427 + Up	
TEL6	TEST +	
TEL7	TONE +	
TEL8	HTTP +	
Administration	ECHO +	
Gateway		

8. Select Routes.

Configuration	General Interf	aces SIP GK	Routes CDR0 CDR1 Calls	admin He	lp
General		-	0.000		
IP		lo	Counter CGPN Maps		
ETH0					

9. Two new routes are needed, one for dialing from the phone attached to the TEL port and one for incoming calls to the SIP account registered with the TEL port.

10.Click on the top-left → icon. For th destination use the drop down list to This applies a 4 second timeout for c	ne source select the o select the matching lialing before the nur	checkbox for the TEL p RAB entry. Ensure tha mber dialed is sent to t	ort just configured. For the at Force enblock is selected. he destination.
This applies a 4 second timeout for c Description	Add UUI Final Route Final Route Final Map No Reroute on wrong No Verify CGPN Interworking(QSIG,SIP) Rerouting as Deflection Routing on Diverting No Force enblock Add # Disable Echo Canceler Call Counter	nber dialed is sent to t	max
 TEST TONE HTTP ECHO SIP1 SIP2 SIP3 SIP4 OK Cancel Apply He 	əlp		

11.Click **OK**. Click on the $\xrightarrow{\frown}$ next to the newly added route. This time selecting the check box for the same RAB entry and in the drop-down list selecting the TEL entry. Click **OK**.

12. The **Routes** form should show the routes just added. The b indicates the Force enblock setting of the outgoing dialing from the phone attached to the TEL1 port.

nfiguration	General Interfaces SIF	P GK Routes	CDR0 CDR1 Calls
General		-	C (CCD) II
IP			Counter CGPN Maps
ETH0	· □, [TEL1:SIP4420] · □,	→ RAB1:SIP4420) b →
LDAP	RAB1:SIP4420	\rightarrow TEL1:SIP4420	\rightarrow

13.To edit an existing route click on the \rightarrow arrow just before the To column.

B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.

C.Make test calls from and to the SIP device.

D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

2.6 Nokia S60 v3 SIP Client

The Nokia S60 SIP Client is a SIP client application that can be installed and used on a range of Nokia phones. The process below was performed on a Nokia e64 but

For Nokia S60 SIP Clients, the IP Office SIP Extension setting Force Authorization should be disabled.

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Select Menu | Tools | Settings | Connection | Sip settings | New SIP profile.
 - 2. Enter the following settings:
 - **Profile name:** Give the profile a name that indicates its function.
 - Service profile: Select IETP.
 - Default access point: Enter your access point.
 - Public user name: Enter an address of the form <IP Office extension number>@<IP Office SIP Enabled LAN IP address>, for example 338@192.168.42.1.
 - Use compression: Select no.
 - Registration: Select always on.
 - Use security: Select no.
 - **Proxy server:** Leave blank.
 - Registrar server:
 - Registrar server adress: Enter the IP Office SIP Enabled LAN IP address.
 - Realm: Enter an address of the form <IP Office user name>@<IP Office SIP Enabled LAN IP address>, for example Extn338@192.168.42.1.
 - User name: Enter the IP Office extension number.
 - Password: Enter the IP Office user's login code.
 - Transport type: Select auto.
 - Port: Match the port set on the IP Office LAN SIP Registrar tab, by default this is 5060.
 - 3. Select Menu | Tools | Settings | Connection | Internet telephone | New profile.
 - Select the SIP profile just created above.
 - 4. Select Menu | Communication | Internet tel. | Options | Settings.
 - Change the **Default call type** to *Internet call*.
- B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.
- C.Make test calls from and to the SIP device.
- D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

2.7 Patton Micro ATA

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Browse to the IP address of the Micro ATA.
 - 2. Login and select **SIP**.

🚸 Home	SIP Configuration	·	
Network - LAN			
Status	SID Server Settings (Ourrest Server 192,169,42,1:5060; Demain	Page PTP Part: 9002)	
Settings	SIP Server Settings (Current Server, 192,108,42,1,5000, Domain	.,Base RTP Foil: 6002)	_
🚸 ToS	* SIP Registration Server Address:	192.168.42.1	System LAN LAN Settings IP Address
Telephony	SIP Port:	5060	
VoIP Status	SIP Domain:		
♦ SIP	Voice Port:	8002	
CODECS	* Leaving a setting blank will force the unit to use the information obtained	via DHCP and/or DNS	
Phone 1	Sand Registration Request with Evoire Time: 3600		
Speed Dial	Send Registration Request with Expire Time.		
System	Send Unregistration at boot		
Documentation	Send SUBSCRIBE.		
Logout	SUBSCRIBE Server IP or FQDN(defaults to registration serve	r):	

- 3. Enter the values to match the settings of the IP Office LAN on which the SIP Registrar is enabled. Click **Save**.
- 4. Select **CODECS**.

A Llomo						
Home AN	Audio/CODEC Configuration					
Statue	CODECS					
 Settings 	Selected	Silence Suppression	Preferred-Codec			
ToS	🗹 G711U	on 😽	0			
Telephony	🗹 G711A	on 🗸	\circ			
VoIP Status	✓ G723	on 🗸	\circ			
SIP	🗹 G726	on 😽	0			
CODECS	✓ G729	on 🗸	۲			
Phone 1						

- 5. Set the codecs to match those set for the IP Office SIP extension. Click Save CODEC Configuration.
- 6. Select Phone 1.

 ♦ Home ♦ Network - LAN ♦ Status 	User Information Phone Number	User User Extn Base E 343	Extension dension CallerID Name	SIP343		
 Settings ToS 	User Name	Extn343	Password	•••••	—User Telephony —User User Nan	Call Settings Login Code ne
Telephony	Port	5060	SIP Registration statu:	s Registered		
 VoIP Status SIP 	Voice Mail Settin	I g *17				
 CODECS Phone 1 						

- 7. Enter the values to match the IP Office SIP extension and user settings. Click Save.
- B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.
- C.Make test calls from and to the SIP device.
- D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

2.8 Polycom SoundPoint Phones

- A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.
 - 1. Browse to the IP address of the phone. By default the phone uses DHCP and displays its IP address on the display.
 - 2. Select **SIP**. You will be requested to enter the administrator name and password. The default values are *Polycom* and *456*.
 - 3. in the **Outbound Proxy** and **Server 1** sections, set the **Address**, **Port** and **Transport** details to match the IP Office LAN on which the SIP registrar is enabled.

					SoundPoin	t IP Configuration
POLICOM		Home	General	Network	SIP	Lines
		SIP C	onfiguration	Parameters:		
Serve	rs			Loca	l Settings	
	Serve	rs				
L			Outbound I	Proxy		
			Address 192	2.168.42.1		
Ī			Port 506	50		
Ī			Transport UD	Ponly 🔽		
			Server	1		
[Address 192	2.168.42.1		
Ī			Port 506	50		
Ī			Transport UD	Ponly 💌		

- 4. Click **Submit**. The phone will reset and load the new settings. That can take up to 2 minutes.
- 5. When you can return to the administration menu, select **Lines**. In the Line 1 section, enter the details to match the IP Office SIP extension and user.

					SoundPoin	t IP Configuration
W POLICOM		Home	General	Network	SIP	Lines
			Line Para	ameters:		
Line 1				Line 2		
	Line 1					
			Identifi	ication		
		Dis	play Name	SIP4637		
			Address	4637	_	User User Extens
		A	uth User ID	SIP4637	_	
		Auth	Password	••••		
			Label	SIP4637		
			Туре	Private Shared		
		Third F	Party Name			
		Num	n Line Keys			
		Calls P	er Line Key			
			Serv	ver 1		
			Address	192.168.42.1		
			Port	5060		
			Transport	UDPonly		

- 6. Click **Submit**. The phone will reset and load the new settings. That will take up to 2 minutes.
- 7. Select **Network** and then **Audio Processing**. Check that the codecs match those configured for the SIP extension on the IP Office. If you make any changes click **Submit** and wait for the phone to reset.
- B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.
- C.Make test calls from and to the SIP device.
- D.If not installing any further SIP devices, **Disable Auto-Create Extn/User** if it is enabled.

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