



IP Office

SIP Extension Support

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Documentation information

For the most current versions of documentation, go to the Avaya Support web site (<http://www.avaya.com/support>) or the IP Office Knowledge Base (<http://marketingtools.avaya.com/knowledgebase/>).

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Chapter 1.

IP Office SIP Extensions

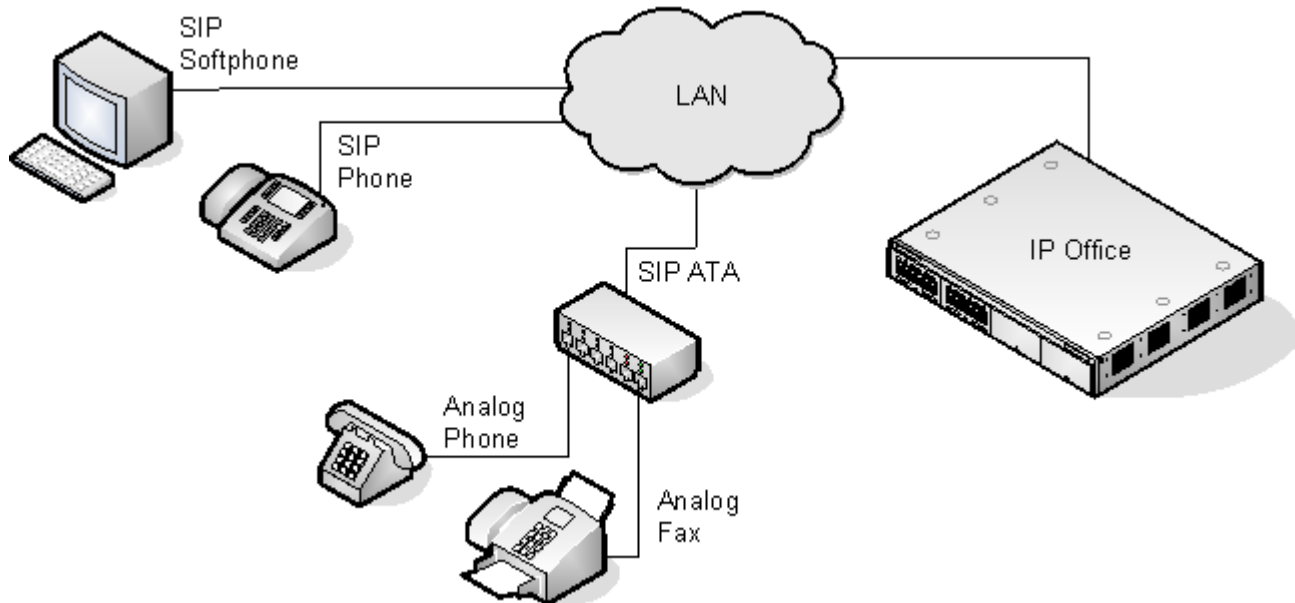
1. IP Office SIP Extensions

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 **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 the IP Office 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.

- **IP Office Voice Compression = SIP Audio Codec**

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 match a codec for which the SIP device is configured.

- **IP Office Call Waiting = SIP 'REFER'**

For the IP Office user associated with a SIP extension, Call Waiting should be enabled if the SIP device supports REFER. This is required for functions such as transferring calls.

- **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. | • Voicemail Collect. |
| • Make calls. | • Unsupervised Transfer. | • Set Forwarding/DND. |
| • Hang Up. | • Supervised Transfer. | • Park/Unpark. |

1.1 Licensing

SIP Extensions are within the IP Office configuration use **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.

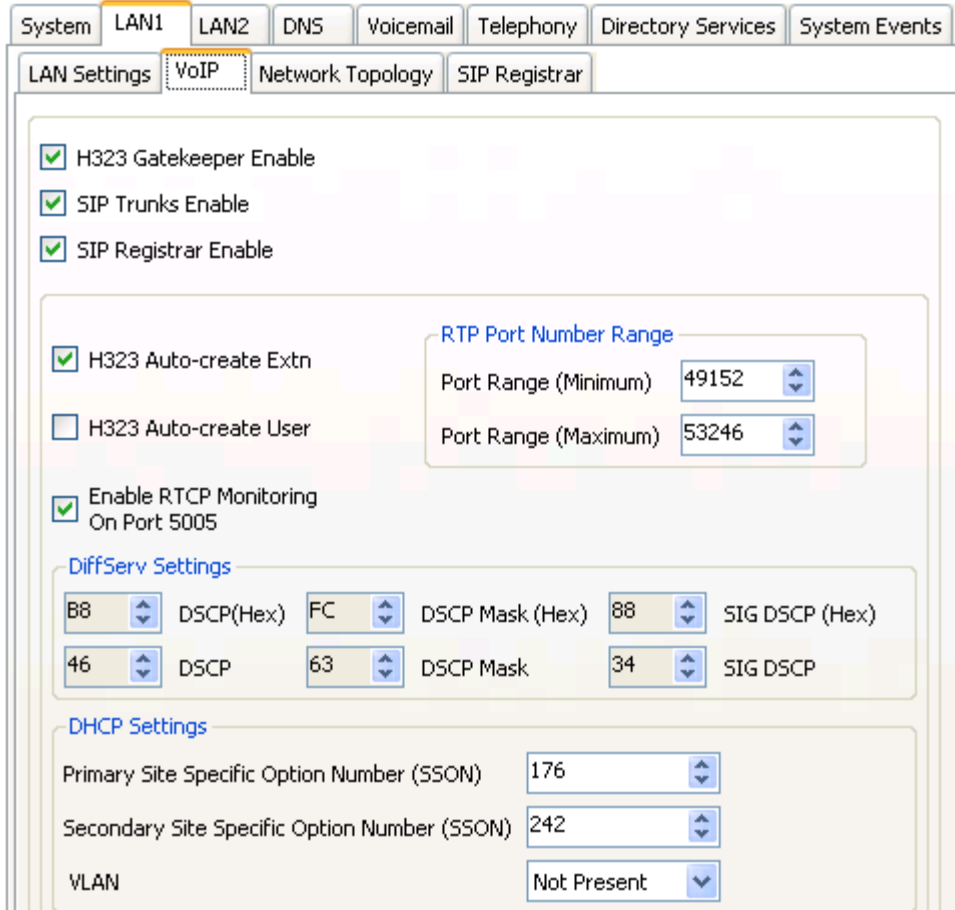
The screenshot displays the Avaya IP Office configuration interface. On the left, a 'Licence' pane lists various license types, with 'IP End-points' highlighted and circled in red. A red line connects this selection to the 'IP End-points' configuration pane on the right. The configuration pane shows the following details:

IP End-points	
Licence Key	FXBh0y@TSjPH4SwnNy5E0Bad5gjMmrwbc
Licence Type	IP End-points
Licence Status	Valid
Instances	20
Expiry Date	Never

1.2 Enabling SIP Extension Support

Once the IP Office system has [valid IP End-points licenses](#)^[8], 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.
- Using IP Office Manager, receive the IP Office system configuration.
- Select  **System**.
- Select either the **LAN1** or **LAN2** tab as required.
- Select the **VoIP** sub-tab.



System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events

LAN Settings VoIP Network Topology SIP Registrar

H323 Gatekeeper Enable

SIP Trunks Enable

SIP Registrar Enable

H323 Auto-create Extn

H323 Auto-create User

Enable RTCP Monitoring On Port 5005

RTP Port Number Range

Port Range (Minimum) 49152

Port Range (Maximum) 53246

DiffServ Settings

88 DSCP(Hex) FC DSCP Mask (Hex) 88 SIG DSCP (Hex)

46 DSCP 63 DSCP Mask 34 SIG DSCP

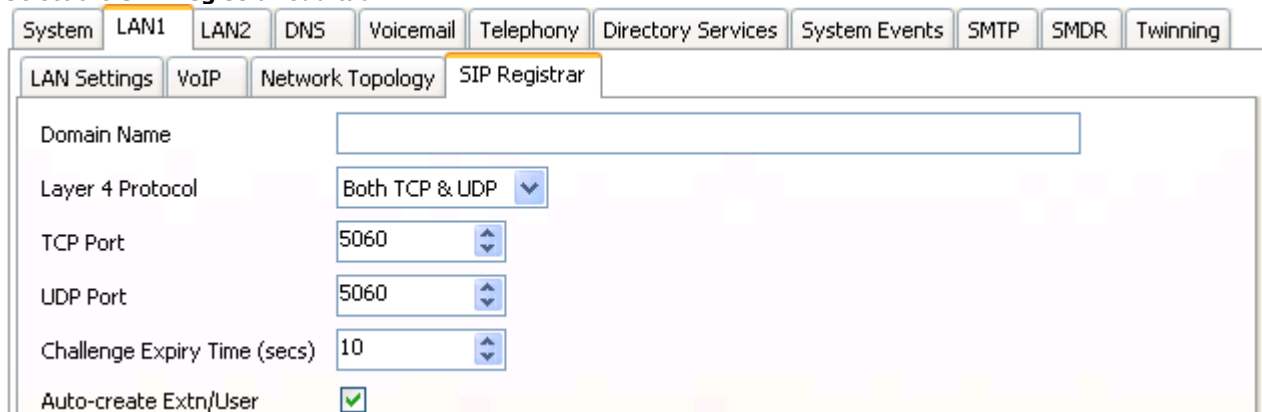
DHCP Settings

Primary Site Specific Option Number (SSON) 176

Secondary Site Specific Option Number (SSON) 242

VLAN Not Present

- Check that **SIP Registrar Enable** is selected.
- Select the **SIP Registrar** sub-tab.



System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR Twinning

LAN Settings VoIP Network Topology SIP Registrar

Domain Name

Layer 4 Protocol Both TCP & UDP

TCP Port 5060

UDP Port 5060

Challenge Expiry Time (secs) 10

Auto-create Extn/User

- 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.

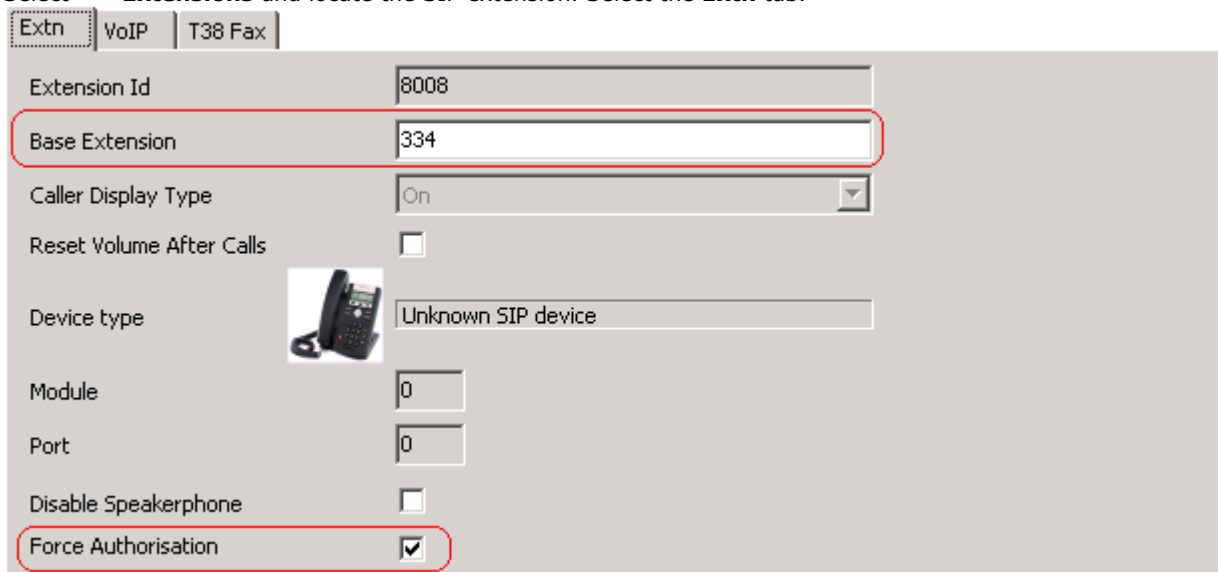
8. Send the configuration back to the IP Office.

1.3 SIP Extension Settings

SIP extensions can be created manually using  | **SIP Extension** or [automatically created](#)¹⁴⁾ during SIP device registration. Even if auto-created, the extension 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.

1. Select  **Extensions** and locate the SIP extension. Select the **Extn** tab.




Extn VoIP T38 Fax

Extension Id 8008

Base Extension 334

Caller Display Type On

Reset Volume After Calls

Device type  Unknown SIP device

Module 0

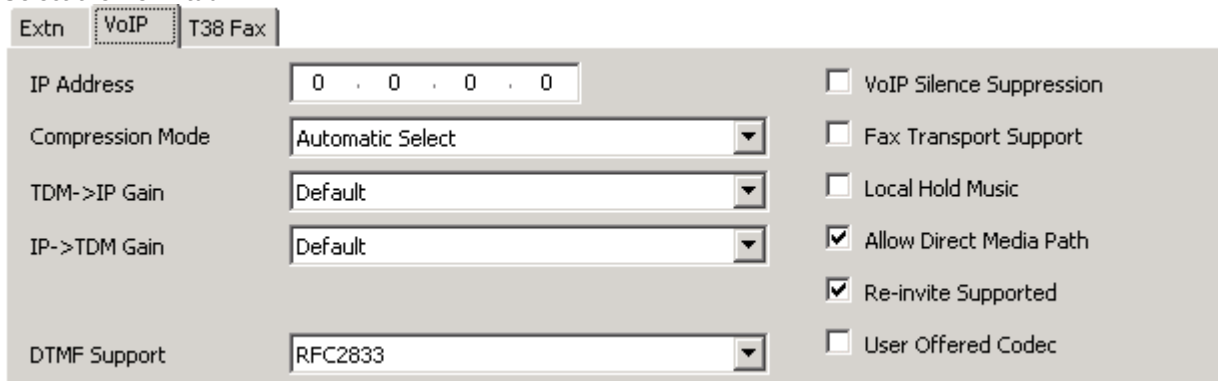
Port 0

Disable Speakerphone

Force Authorisation

- **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 Fax

IP Address 0 . 0 . 0 . 0

Compression Mode Automatic Select

TDM->IP Gain Default

IP->TDM Gain Default

DTMF Support RFC2833

VoIP Silence Suppression

Fax Transport Support

Local Hold Music


Allow Direct Media Path

Re-invite Supported

User Offered Codec

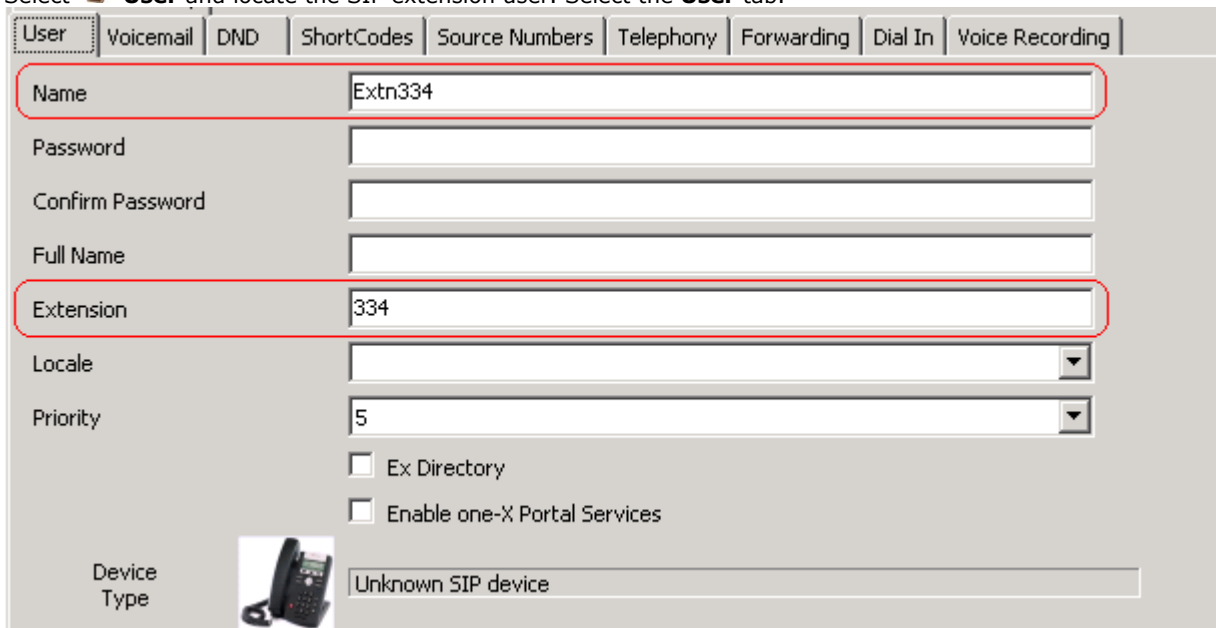
- **Compression Mode**
The selected mode must match an audio codec supported by the SIP device. If set to **Automatic Select**, then the codecs supported by the IP Office are set through the configuration option **System | Telephony | Telephony | Automatic Codec Preferences**.
 - **User Offered Codec**
If the SIP device is configured with a preferred first codec, enabling this option ensures that codec is used on calls to the SIP device.
- **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.

1.4 SIP User Settings

SIP users can be created manually using  **User** or [automatically created](#) ¹⁴ 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.

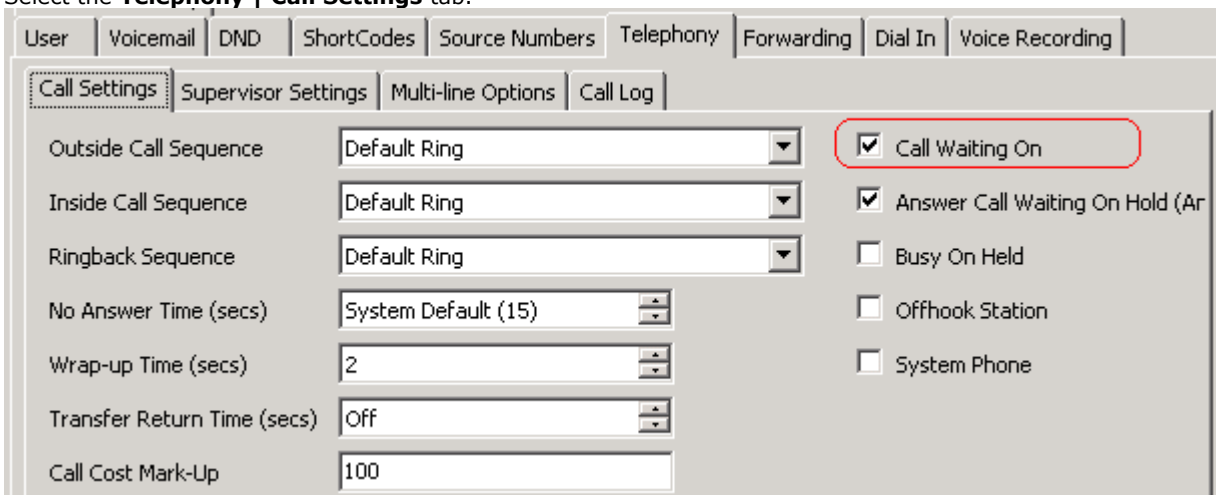
1. Select  **User** and locate the SIP extension user. Select the **User** tab.



User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording
Name	Extn334							
Password								
Confirm Password								
Full Name								
Extension	334							
Locale								
Priority	5							
	<input type="checkbox"/> Ex Directory							
	<input type="checkbox"/> Enable one-X Portal Services							
Device Type	Unknown SIP device							

- **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.



User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording
Call Settings	Supervisor Settings	Multi-line Options	Call Log					
Outside Call Sequence	Default Ring	<input checked="" type="checkbox"/> Call Waiting On						
Inside Call Sequence	Default Ring	<input checked="" type="checkbox"/> Answer Call Waiting On Hold (Ar						
Ringback Sequence	Default Ring	<input type="checkbox"/> Busy On Held						
No Answer Time (secs)	System Default (15)	<input type="checkbox"/> Offhook Station						
Wrap-up Time (secs)	2	<input type="checkbox"/> System Phone						
Transfer Return Time (secs)	Off							
Call Cost Mark-Up	100							

- **Call Waiting On**
Most SIP devices require this setting to be enabled in order to allow features such as transferring calls.


3. Select the **Telephony | Supervisor Settings** tab.

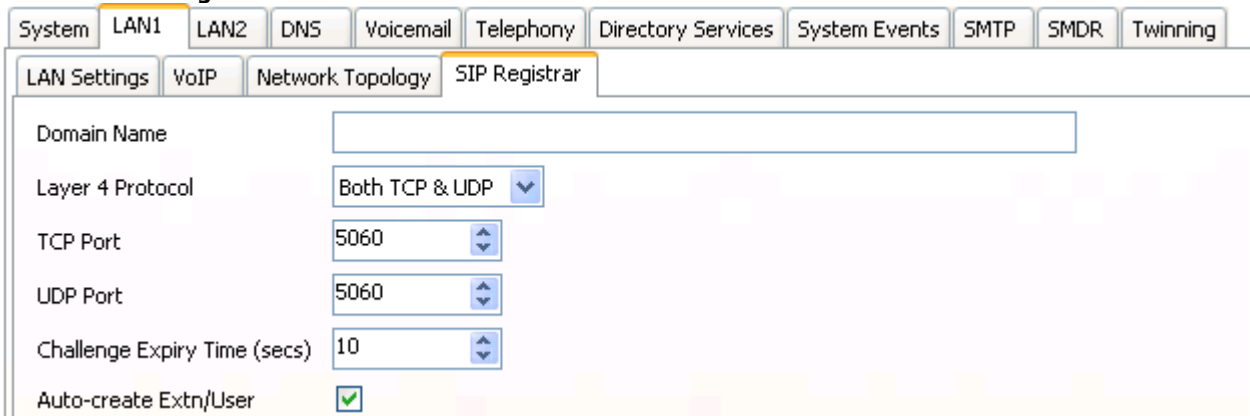
The screenshot shows the 'Supervisor Settings' tab in the SIP User Settings interface. The 'Login Code' field is highlighted with a red box and contains the text '****'. Below it are fields for 'Login Idle Period (secs)', 'Monitor Group' (set to '<None>'), 'Coverage Group' (set to '<None>'), and 'Status on No-Answer' (set to 'Logged On (No change)'). There is a 'Reset Longest Idle Time' section with two radio buttons: 'All Calls' (selected) and 'External Incoming'. At the bottom, there is an 'After Call Work Time (secs)' field set to 'System Default (10)'. On the right side, there is a list of checkboxes: 'Force Login', 'Force Account Code', 'Outgoing Call Bar', 'Inhibit Off-Switch Forward/Transfe', 'Can Intrude', 'Cannot be Intruded' (checked), 'Can Trace Calls', 'CCR Agent', and 'Automatic After Call Work'.

- **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

1. Note that changing the SIP registrar settings of an IP Office system requires the IP Office system to be rebooted.
2. Using IP Office Manager, receive the IP Office system configuration.
3. Select  **System**.
4. Select either the **LAN1** or **LAN2** tab on which the SIP registrar is enabled.
5. Select the **SIP Registrar** sub-tab.



System | LAN1 | LAN2 | DNS | Voicemail | Telephony | Directory Services | System Events | SMTP | SMDR | Twinning

LAN Settings | VoIP | Network Topology | SIP Registrar

Domain Name:

Layer 4 Protocol: Both TCP & UDP

TCP Port: 5060

UDP Port: 5060

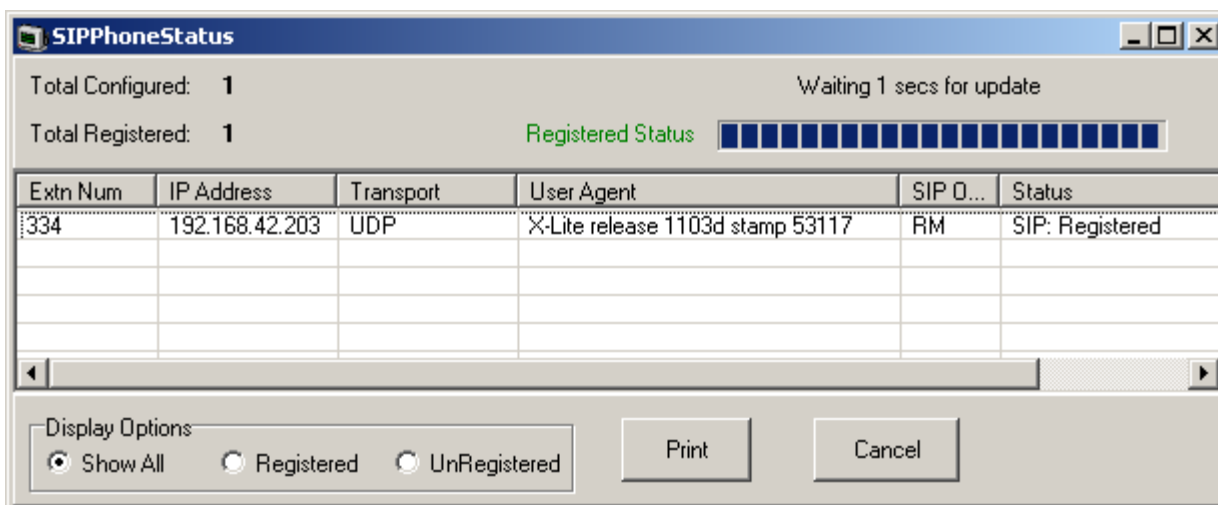
Challenge Expiry Time (secs): 10

Auto-create Extn/User:

6. Change the Auto-create Extn/User settings to the state required.
7. 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 in System Monitor. Select **Status | SIP Phone Status**, to display the SIP extension list.



SIPPhoneStatus

Total Configured: 1 Waiting 1 secs for update

Total Registered: 1 Registered Status 

Extn Num	IP Address	Transport	User Agent	SIP O...	Status
334	192.168.42.203	UDP	X-Lite release 1103d stamp 53117	RM	SIP: Registered

Display Options: Show All Registered UnRegistered

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:

- [CounterPath Eyebeam/X-Lite Softphones](#) ^[17]
- [Polycom Soundpoint](#) ^[20]
- [Grandstream GXP 2000, GXP 2020](#) ^[21]
- [Avaya A10 ATA](#) ^[23]
- [Patton Micro ATA](#) ^[27]
- [Nokia S60 v3 SIP Client](#) ^[28]
- [Innovaphone IP22, IP24, IP28](#) ^[29]

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
<ol style="list-style-type: none">1. Add and check IP End-points licenses.2. Check the SIP Registrar settings.3. Enable Auto-Create Extn/User.4. Attach and configure the SIP device.5. Modify the IP Office user and extension settings.6. Disable Auto-Create Extn/User.	<ol style="list-style-type: none">1. Add and check IP End-points licenses.2. Check the SIP Registrar settings.3. Add SIP Extension settings to the IP Office configuration.4. Add SIP User settings to the IP Office configuration.5. Attach and configure the SIP device.

2.1 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.

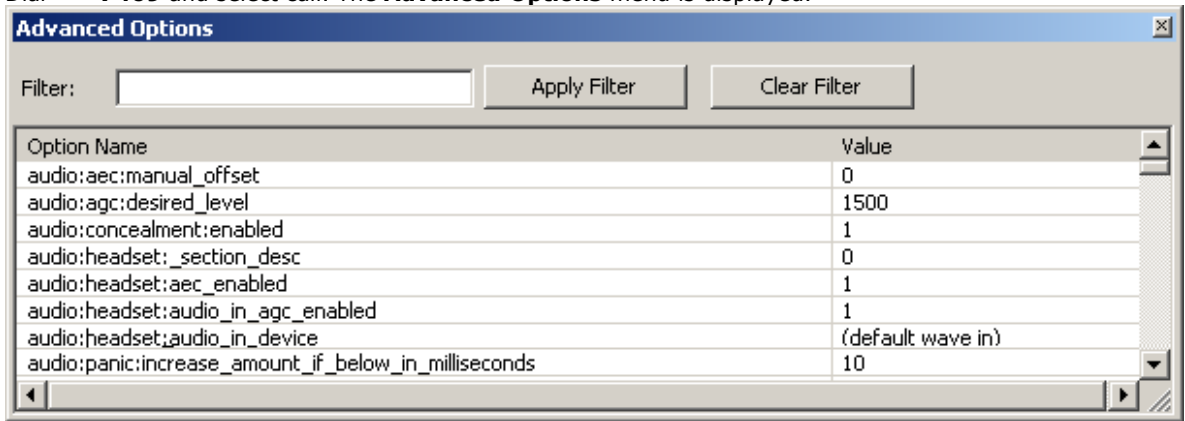
1. Either enable the IP Office to allow [automatic creation](#) ¹⁴ based on SIP phone registration or manually add the SIP extension and user details to the IP Office configuration.
2. Start the X-Lite SIP client application.
3. Click on the down arrow icon and select **SIP Account Settings....**
4. Click on **Add....**

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**.

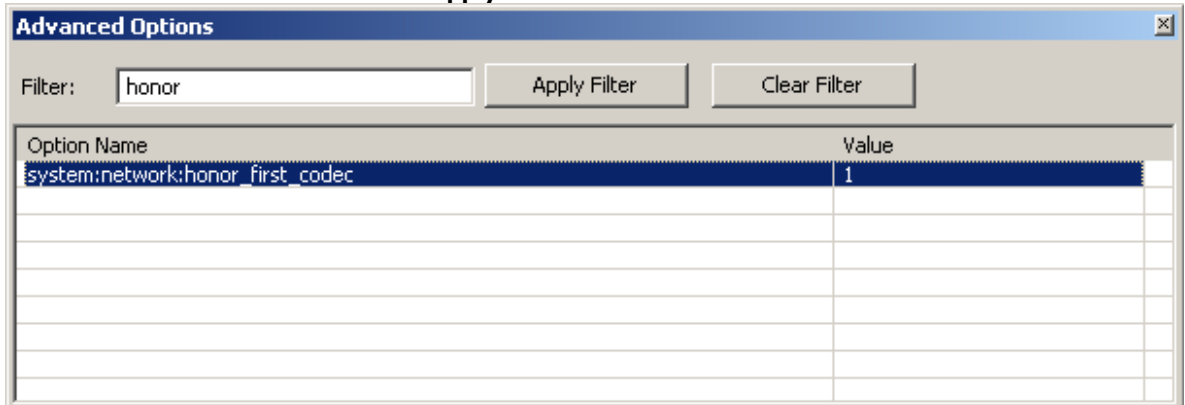
Enabled	Acct #	Domain	Username	Display Name
<input checked="" type="checkbox"/>	1	192.168.42.1 (default)	334	SIPMe
<input type="checkbox"/>				
<input type="checkbox"/>				
<input type="checkbox"/>				

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](#) ¹⁹ or to perform the following process.

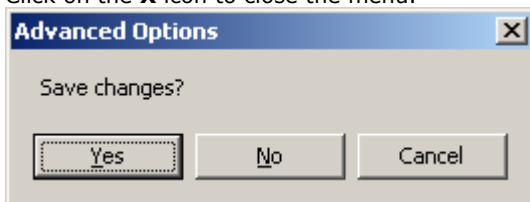
- a. Dial *****7469** and select call. The **Advanced Options** menu is displayed.



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



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



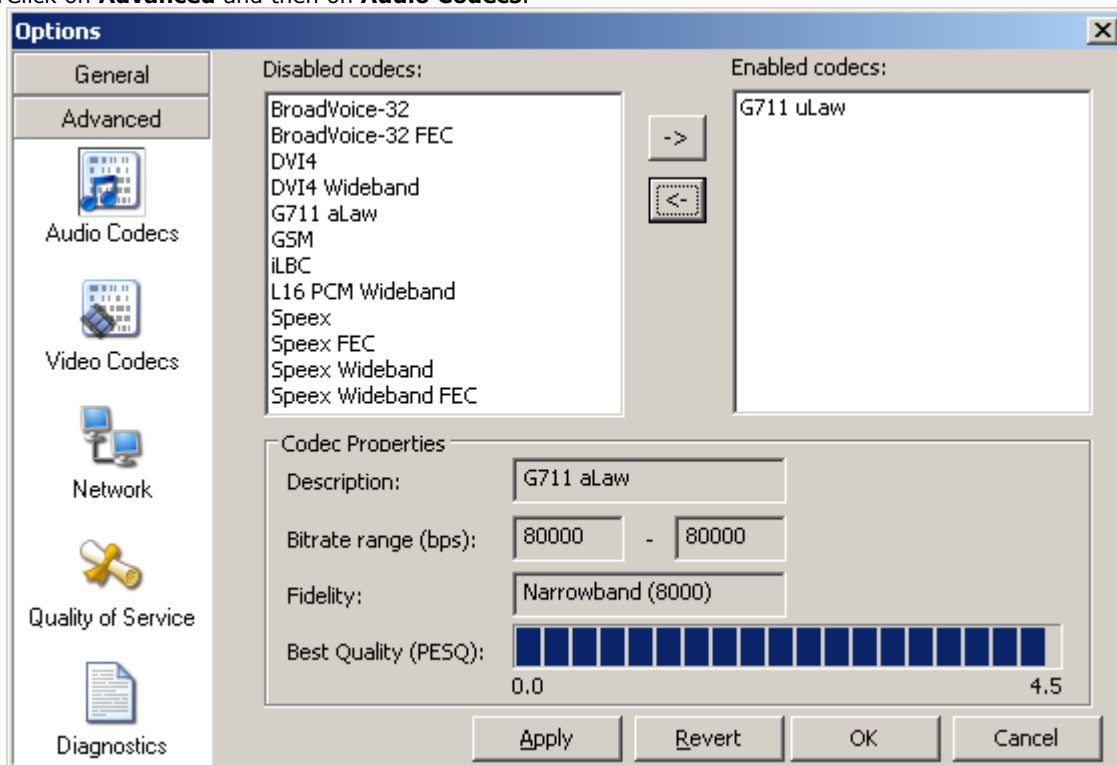
- e. Click on **Yes** to save the change.

- 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**.



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**.

2.2 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.

POLYCOM SoundPoint IP Configuration

Home General Network SIP Lines

SIP Configuration Parameters:

Servers Local Settings

Servers

Outbound Proxy

Address: 192.168.42.1

Port: 5060

Transport: UDPonly

Server 1

Address: 192.168.42.1

Port: 5060

Transport: UDPonly

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.

POLYCOM SoundPoint IP Configuration

Home General Network SIP Lines

Line Parameters:

Line 1 Line 2

Line 1

Identification

Display Name: SIP4637

Address: 4637

Auth User ID: SIP4637

Auth Password: ●●●●

Label: SIP4637

Type: Private Shared

Third Party Name:

Num Line Keys:

Calls Per Line Key:

Server 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.

2.3 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.

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ACCOUNT 1
ACCOUNT 2
ACCOUNT 3
ACCOUNT 4
ACCOUNT 5
ACCOUNT 6

Account Active: No Yes

Account Name:

SIP Server: System | LAN | LAN Settings | IP Address

Outbound Proxy:

SIP User ID: User | User | Extension
Extn | Base Extension

Authenticate ID: User | User | Name

Authenticate Password:

Name: User | Telephony | Call Settings | Login Code

local SIP port: (default 5060)

SIP Registration Failure Retry Wait Time: (in seconds. Between 1-3600, default is 20)

SIP T1 Timeout:

SIP T2 Interval:

SIP Transport: UDP TCP

Use RFC3581 Symmetric Routing: No Yes

NAT Traversal (STUN): No No, but send keep-alive Yes

SUBSCRIBE for MWI: No Yes

PUBLISH for Presence: No Yes

Proxy-Require:

Voice Mail UserID: (UserID for voice mail system)

Preferred Vocoder: (in listed order)

choice 1: <input type="text" value="G.729A/B"/>	choice 5: <input type="text" value="G.726-32"/>
choice 2: <input type="text" value="PCMA"/>	choice 6: <input type="text" value="iLBC"/>
choice 3: <input type="text" value="G.723.1"/>	choice 7: <input type="text" value="G.722 (wide band)"/>
choice 4: <input type="text" value="PCMU"/>	choice 8: <input type="text" value="GSM"/>

SRTP Mode: Disabled Enabled but not forced Enabled and forced

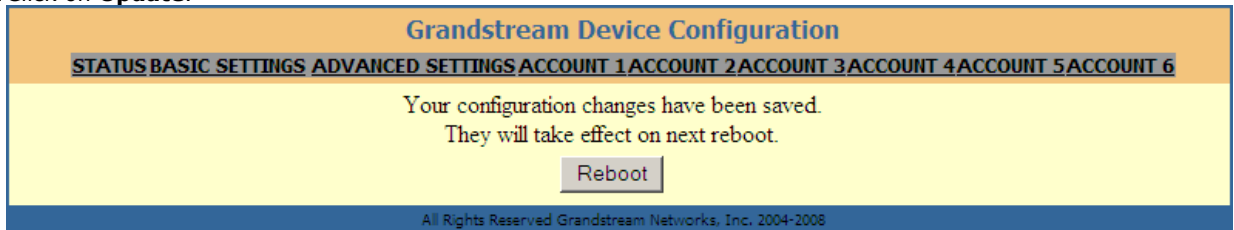
eventlist BLF URI:

Special Feature:

All Rights 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**.



5. Click on **Reboot**. The phone may take up to 1 minute to reboot.



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.

2.4 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**.

192.168.1.1 / Telephony / SIP

Home Import/Export

Gateways Interfaces Profiles

Name	Domain	Default-Server	Registration	Authentication	Binding	State
sip		/	To /	(none)	eth0	Enabled

Network
IP/DNS
NAT/NAPT

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

SIP Gateway Enabled

Local Call Signaling Port 5060

Call Signaling Traffic Class local-default

INVITE Transaction Timeout 32 seconds

Non-INVITE Transaction Timeout 32 seconds

Transport Protocols TCP UDP

Penalty Box 600 seconds Time for which a non-responsive destination should stay in the penalty box, i.e. should not be contacted anymore

Apply

Services

default

Various

Network
IP/DNS
NAT/NAPT
ACL
QoS
DynDNS
DHCP Server
WAN
Telephony
Call-Router
H.323
SIP
VoIP Profiles
Tone Profiles
PSTN Profiles
Ports
Ethernet
FXS
Various

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

192.168.1.1 / Telephony / SIP / Gateway sip / Service default

Home Import/Export

Configuration Registration and Authentication

Domain

Default-Server (Outbound Proxy) Set manual Host Port Server Type loose-router

Set always the actual Registrar as Default Server

Force Keep-Alives 3600 seconds

Call Transfer Version: 5

Session Timer Version: 8

Create new session after redirect

Alternate Contact Address Detect NAT Address User Defined IP Address

SIP Profile default

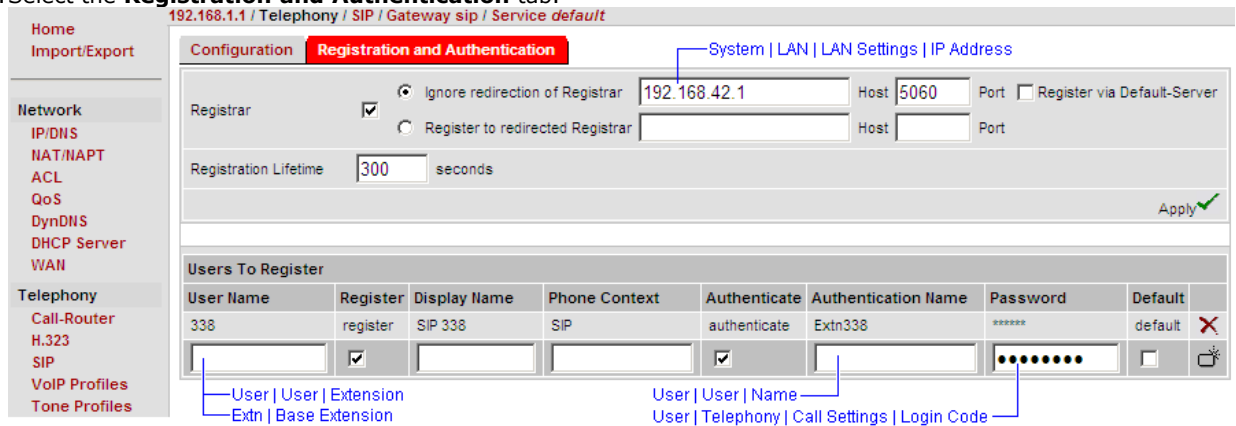
VoIP Profile default

Apply

Network
IP/DNS
NAT/NAPT
ACL
QoS
DynDNS
DHCP Server
WAN
Telephony
Call-Router
H.323
SIP
VoIP Profiles
Tone Profiles
PSTN Profiles
Ports
Ethernet
FXS

- 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** .

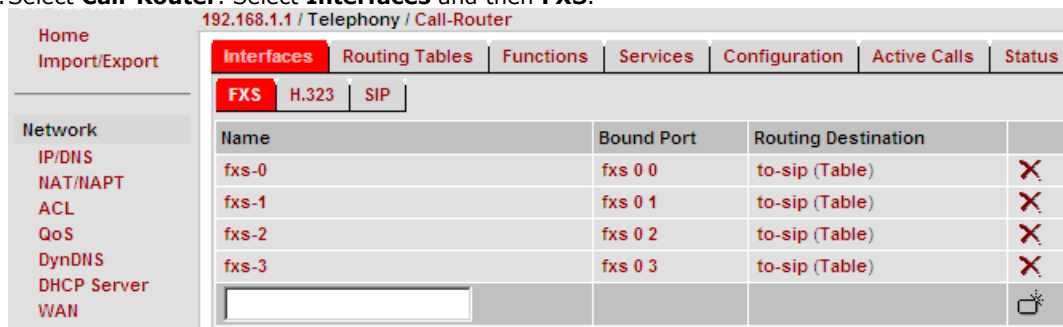
6. Select the **Registration and Authentication** tab.



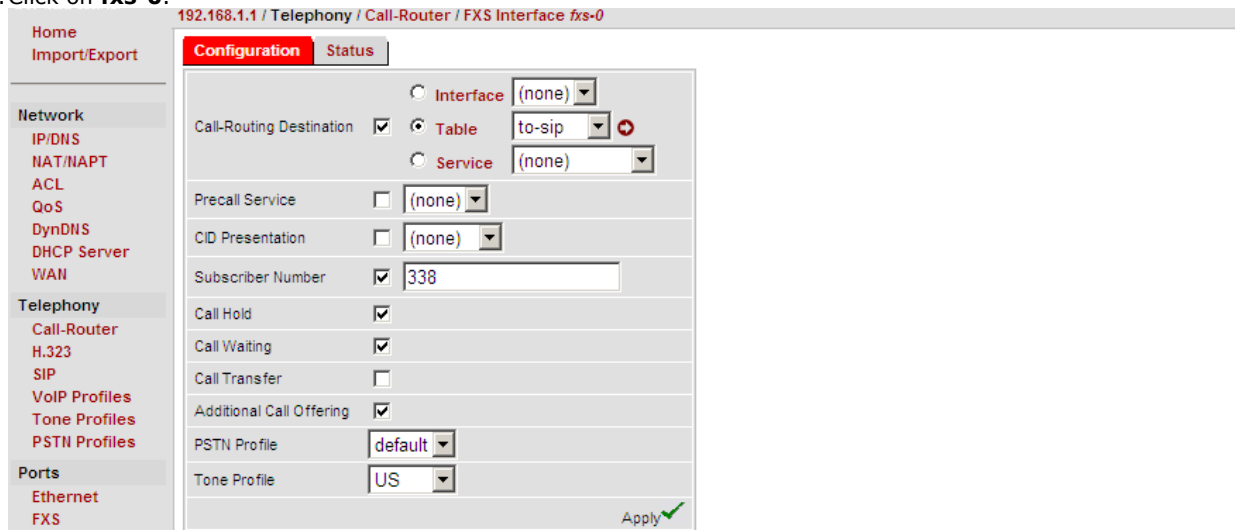
- Enable the Registrar checkbox. Select **Ignore redirection of Registrar** and enter the IP address and SIP port of the IP Office LAN on which the SIP registrar is enabled. Click **Apply** ✓.

7. In the **Users To Register** section, create a user matching the IP Office SIP extension and user. Enter the settings and click on

8. Select **Call-Router**. Select **Interfaces** and then **FXS**.



9. Click on **fxs-0**.



- 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  arrow icon after **to-sip**.

192.168.1.1 / Telephony / Call-Router / Routing Table *to-sip*

Home
Import/Export

Network
IP/DNS
NAT/NAPT
ACL
QoS
DynDNS
DHCP Server
WAN

Configuration

Looks Up For <i>called-e164 Of</i>	Destination	Execute Function (Optional)
T	sip (SIP Interface)	
<i>called-e164 value or default</i>	<input type="radio"/> Interface (none) <input type="radio"/> Table (none) <input type="radio"/> Service (none) <input type="radio"/> none	Optional function to execute (none)
(To change an entry, enter the value of an existing entry)		

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

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

192.168.1.1 / Telephony / Call-Router

Home
Import/Export

Network
IP/DNS
NAT/NAPT
ACL
QoS
DynDNS

Interfaces **Routing Tables** Functions Services Configuration Active Calls Status

Routing Tables

Name	Looks up for	
from-sip	called-e164	
to-sip	called-e164	
	called-e164	


12. Select **from-sip**.


192.168.1.1 / Telephony / Call-Router / Routing Table *from-sip*

Home
Import/Export

Network
IP/DNS
NAT/NAPT
ACL
QoS
DynDNS
DHCP Server

Configuration

Looks Up For <i>called-e164 Of</i>	Destination	Execute Function (Optional)
<i>called-e164 value or default</i>	<input checked="" type="radio"/> Interface fxs-0 <input type="radio"/> Table (none) <input type="radio"/> Service (none) <input type="radio"/> none	Optional function to execute (none)
338		
(To change an entry, enter the value of an existing entry)		

- 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

Network
IP/DNS
NAT/NAPT
ACL
QoS
DynDNS
DHCP Server
WAN

Configuration

Looks Up For <i>called-e164 Of</i>	Destination	Execute Function (Optional)
338	fxs-0 (FXS Interface)	
<i>called-e164 value or default</i>	<input type="radio"/> Interface (none) <input type="radio"/> Table (none) <input type="radio"/> Service (none) <input type="radio"/> none	Optional function to execute (none)
(To change an entry, enter the value of an existing entry)		

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

Network
IP/DNS
NAT/NAPT
ACL

Save Configuration

You are going to save the modified configuration persistently. 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?

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 a 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.

192.168.1.1 / Telephony / VoIP Profiles / Profile *default*

Home
Import/Export

Voice Fax Modem Dejitter Buffer Status

Network

- IP/DNS
- NAT/NAPT
- ACL
- QoS
- DynDNS
- DHCP Server
- WAN

Telephony

- Call-Router
- H.323
- SIP
- VoIP Profiles
- Tone Profiles
- PSTN Profiles

Ports

- Ethernet
- FXS

Various

- System
- AAA
- Time

Position	Codec	Rx Length [ms]	Tx Length [ms]	Silence Suppression		
1	g711ulaw64k	20	20	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	✓	✗
2	g711alaw64k	20	20	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	✓	✗
3	g729	20	20	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	✓	✗
	transparent			<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no		✗

Additional Voice Parameters

Default Silence Suppression If not specified by the codec

Highpass Filter Voice input filter for A/D conversion

Post Filter Voice output filter for D/A conversion

DTMF Relay

RTP Payload Type For Tone Events (NTE)

RTP Payload Type For Signaling Events (NSE)

RTP Traffic Class

Apply ✓

2.5 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**.

SIP Configuration

SIP Server Settings (Current Server: 192.168.42.1:5060; Domain:; Base RTP Port: 8002)

* SIP Registration Server Address: 192.168.42.1 —System | LAN | LAN Settings | IP Address

SIP Port: 5060

SIP Domain:

Voice Port: 8002

* Leaving a setting blank will force the unit to use the information obtained via DHCP and/or DNS

Send Registration Request with Expire Time: 3600

Send Unregistration at boot

Send SUBSCRIBE.

SUBSCRIBE Server IP or FQDN(defaults to registration server):

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**.

Audio/CODEC Configuration

CODECS

Selected	Silence Suppression	Preferred-Codec
<input checked="" type="checkbox"/> G711U	ON <input type="button" value="v"/>	<input type="radio"/>
<input checked="" type="checkbox"/> G711A	ON <input type="button" value="v"/>	<input type="radio"/>
<input checked="" type="checkbox"/> G723	ON <input type="button" value="v"/>	<input type="radio"/>
<input checked="" type="checkbox"/> G726	ON <input type="button" value="v"/>	<input type="radio"/>
<input checked="" type="checkbox"/> G729	ON <input type="button" value="v"/>	<input checked="" type="radio"/>

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

User Information

Phone Number: 343 —User | User | Extension
Extn | Base Extension

CallerID Name: SIP343

User Name: Extn343 —User | Telephony | Call Settings | Login Code
User | User | Name

Password: ●●●●●●

Port: 5060 —SIP Registration status Registered

Voice Mail Setting

Voice Mail Number: *17

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.

2.6 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 address:** 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 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.

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.

4. Select **Interfaces**.

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registration
TEL1	+				Up		
TEL2	+				Up		
TEL3	+				Up		
TEL4	+				Up		
TEL5	+				Up		
TEL6	+				Up		
TEL7	+				Up		
TEL8	+				Up		
TEST	+						
TONE	+						
HTTP	+						
ECHO	+						

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

6. In the **Protocol** drop down list select **SIP**. Enter the details as indicated below to match your IP Office SIP extension and user.

Name

Disable

Tones

Interface Maps

Internal Registration

Protocol

Server Address (primary) System | LAN | LAN Settings | IP Address

Server Address (secondary) User | User | Extension
- Extn | Base Extension

ID@Domain @

Username User | User | Name

Password Retype User | Telephony | Call Settings | Login Code

Feature Codes Support (with Feature Codes)

Dynamic Group

Direct Dial

Locked White List

Media Properties

General Coder Preference Framesize [ms] Silence Compression Exclusive

Local Network Coder Framesize [ms] Silence Compression

Enable T.38 Enable SRTP No DTMF Detection MOH Mode

7. Click **OK**.

Configuration	General	Interfaces	SIP	GK	Routes	CDR0	CDR1	Calls	admin	Help
General										
IP										
ETH0										
LDAP										
TEL1										
TEL2										
TEL3										
TEL4										
TEL5										
TEL6										
TEL7										
TEL8										
Administration										
Gateway										


Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registration
TEL1	SIP4420	+			Up	:4420 →	135.64.181.220
TEL2	SIP4421	+			Up		
TEL3	SIP4422	+			Up		
TEL4	SIP4423	+			Up		
TEL5	SIP4424	+			Up		
TEL6	SIP4425	+			Up		
TEL7	SIP4426	+			Up		
TEL8	SIP4427	+			Up		
TEST		+					
TONE		+					
HTTP		+					
ECHO		+					

8. Select **Routes**.

Configuration	General	Interfaces	SIP	GK	Routes	CDR0	CDR1	Calls	admin	Help
General										
IP										
ETH0										

From	To	Counter	CGPN Maps

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 the source select the checkbox for the **TEL** port just configured. For the destination use the drop down list to select the matching **RAB** entry. Ensure that **Force enblock** is selected. This applies a 4 second timeout for dialing before the number dialed is sent to the destination.

Description Disable

TEL1 SIP4420 → RAB1 SIP4420 Cause(DISC)

RAB1 SIP4420

TEL2

RAB2

TEL3

RAB3

TEL4

RAB4

TEL5

RAB5 GW1

TEL6 GW2

RAB6 GW3

TEL7 GW4

RAB7 GW5

TEL8 GW6

RAB8

TEST

TONE

HTTP

ECHO

SIP1

SIP2

SIP3

SIP4

Add UUI

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 max

OK Cancel Apply Help

11. Click **OK**. Click on the  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.

Configuration	General	Interfaces	SIP	GK	Routes	CDR0	CDR1	Calls	admin	Help															
General																									
IP																									
ETH0																									
LDAP																									
<table border="1"> <thead> <tr> <th>From</th> <th>To</th> <th>Counter</th> <th>CGPN</th> <th>Maps</th> </tr> </thead> <tbody> <tr> <td> TEL1:SIP4420</td> <td> RAB1:SIP4420</td> <td>b</td> <td></td> <td>→</td> </tr> <tr> <td> RAB1:SIP4420</td> <td> TEL1:SIP4420</td> <td></td> <td></td> <td>→</td> </tr> </tbody> </table>											From	To	Counter	CGPN	Maps	 TEL1:SIP4420	 RAB1:SIP4420	b		→	 RAB1:SIP4420	 TEL1:SIP4420			→
From	To	Counter	CGPN	Maps																					
 TEL1:SIP4420	 RAB1:SIP4420	b		→																					
 RAB1:SIP4420	 TEL1:SIP4420			→																					

13. To edit an existing route click on the  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.

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