



# **IP Office**

## SIP Extension Support

#### Notice

While reasonable efforts were made to ensure that the information in this document was complete and accurate at the time of printing, Avaya Inc. can assume no liability for any errors. Changes and corrections to the information in this document may be incorporated in future releases.

#### Documentation Disclaimer

Avaya Inc. is not responsible for any modifications, additions, or deletions to the original published version of this documentation unless such modifications, additions, or deletions were performed by Avaya.

#### Link Disclaimer

Avaya Inc. is not responsible for the contents or reliability of any linked Web sites referenced elsewhere within this Documentation, and Avaya does not necessarily endorse the products, services, or information described or offered within them. We cannot guarantee that these links will work all of the time and we have no control over the availability of the linked pages.

#### License

USE OR INSTALLATION OF THE PRODUCT INDICATES THE END USER'S ACCEPTANCE OF THE TERMS SET FORTH HEREIN AND THE GENERAL LICENSE TERMS AVAILABLE ON THE AVAYA WEBSITE AT <http://support.avaya.com/LicenseInfo/> ("GENERAL LICENSE TERMS"). IF YOU DO NOT WISH TO BE BOUND BY THESE TERMS, YOU MUST RETURN THE PRODUCT(S) TO THE POINT OF PURCHASE WITHIN TEN (10) DAYS OF DELIVERY FOR A REFUND OR CREDIT.

Avaya grants End User a license within the scope of the license types described below. The applicable number of licenses and units of capacity for which the license is granted will be one (1), unless a different number of licenses or units of capacity is specified in the Documentation or other materials available to End User. "Designated Processor" means a single stand-alone computing device. "Server" means a Designated Processor that hosts a software application to be accessed by multiple users. "Software" means the computer programs in object code, originally licensed by Avaya and ultimately utilized by End User, whether as stand-alone Products or pre-installed on Hardware. "Hardware" means the standard hardware Products, originally sold by Avaya and ultimately utilized by End User.

License Type(s): Designated System(s) License (DS).

End User may install and use each copy of the Software on only one Designated Processor, unless a different number of Designated Processors is indicated in the Documentation or other materials available to End User. Avaya may require the Designated Processor(s) to be identified by type, serial number, feature key, location or other specific designation, or to be provided by End User to Avaya through electronic means established by Avaya specifically for this purpose.

#### Copyright

Except where expressly stated otherwise, the Product is protected by copyright and other laws respecting proprietary rights. Unauthorized reproduction, transfer, and or use can be a criminal, as well as a civil, offense under the applicable law.

#### Third-Party Components

Certain software programs or portions thereof included in the Product may contain software distributed under third party agreements ("Third Party Components"), which may contain terms that expand or limit rights to use certain portions of the Product ("Third Party Terms"). Information identifying Third Party Components and the Third Party Terms that apply to them is available on Avaya's web site at: <http://support.avaya.com/ThirdPartyLicense/>

#### Avaya Fraud Intervention

If you suspect that you are being victimized by toll fraud and you need technical assistance or support, call Technical Service Center Toll Fraud Intervention Hotline at +1-800-643-2353 for the United States and Canada. Suspected security vulnerabilities with Avaya Products should be reported to Avaya by sending mail to: [securityalerts@avaya.com](mailto:securityalerts@avaya.com). For additional support telephone numbers, see the Avaya Support web site (<http://www.avaya.com/support>).

#### Trademarks

Avaya and the Avaya logo are registered trademarks of Avaya Inc. in the United States of America and other jurisdictions. Unless otherwise provided in this document, marks identified by "®," "TM" and "SM" are registered marks, trademarks and service marks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners.

#### 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/>).

#### Avaya Support

Avaya provides a telephone number for you to use to report problems or to ask questions about your contact center. The support telephone number is 1 800 628 2888 in the United States. For additional support telephone numbers, see the Avaya Web site: <http://www.avaya.com/support>.

---

# Contents

## 1. IP Office SIP Extensions

1.1 Licensing .....	8
1.2 Enabling SIP Extension Support.....	10
1.3 SIP Extension Settings.....	12
1.4 SIP User Settings.....	14
1.5 Allowing SIP Extn/User Auto Creation.....	16
1.6 System Monitor .....	16

## 2. SIP Device Configuration

2.1 CounterPath eyeBeam/X-Lite.....	20
2.2 Polycom SoundPoint Phones.....	23
2.3 Grandstream.....	25
2.4 Avaya A10 ATA.....	28
2.5 Patton Micro ATA.....	32
2.6 Nokia S60 v3 SIP Client.....	33
2.7 Innovaphone IP22, IP24, IP28.....	35
Index .....	0



# **Chapter 1.**

## **IP Office SIP Extensions**



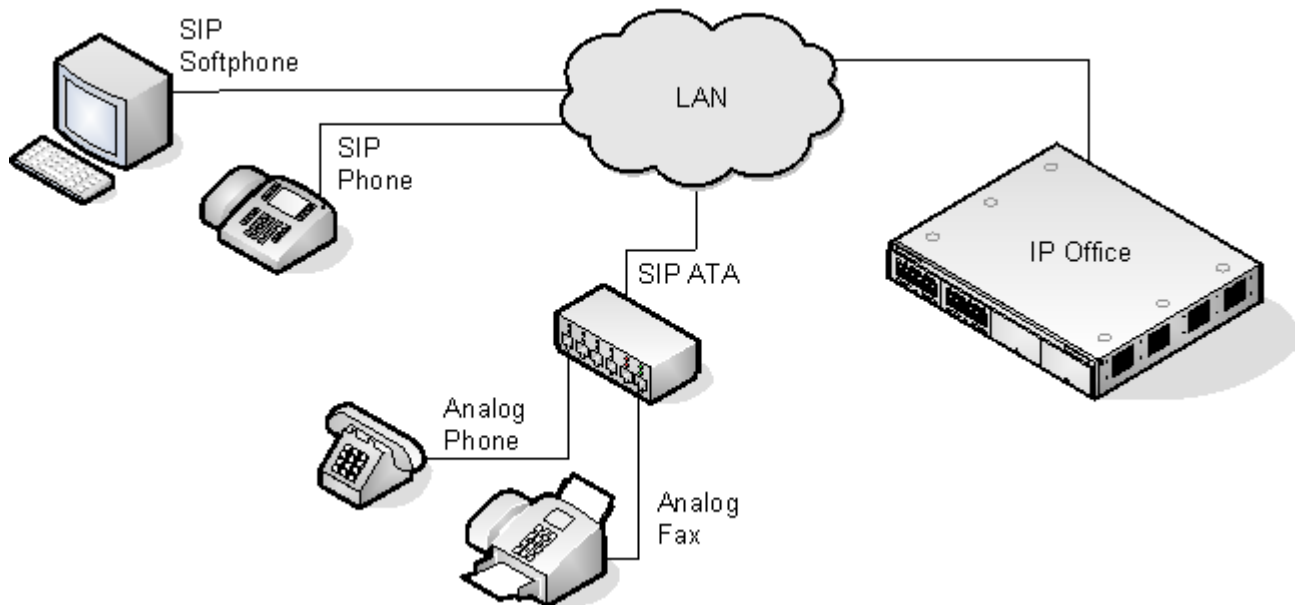
# 1. IP Office SIP Extensions

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 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.

- |                        |                                 |                              |
|------------------------|---------------------------------|------------------------------|
| • <b>Answer calls.</b> | • <b>Hold.</b>                  | • <b>Voicemail Collect.</b>  |
| • <b>Make calls.</b>   | • <b>Unsupervised Transfer.</b> | • <b>Set Forwarding/DND.</b> |
| • <b>Hang Up.</b>      | • <b>Supervised Transfer.</b>   | • <b>Park/Unpark.</b>        |

# 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.

Licence

Licence Type

1600 Series Phones

Advanced Small Community Networking

CCC Server

CCR Agent

CCR CCC UPG

CCR SUP

IP End-points

IP500 Upgrade Standard to Professional

IP500 VCM Channels

IP End-points

Licences

Licence Key

FXBh0y@TSjph4SwNy5E0BadSgjMmrwbc

Licence Type

IP End-points

Licence Status

Valid

Instances

20

Expiry Date


Never

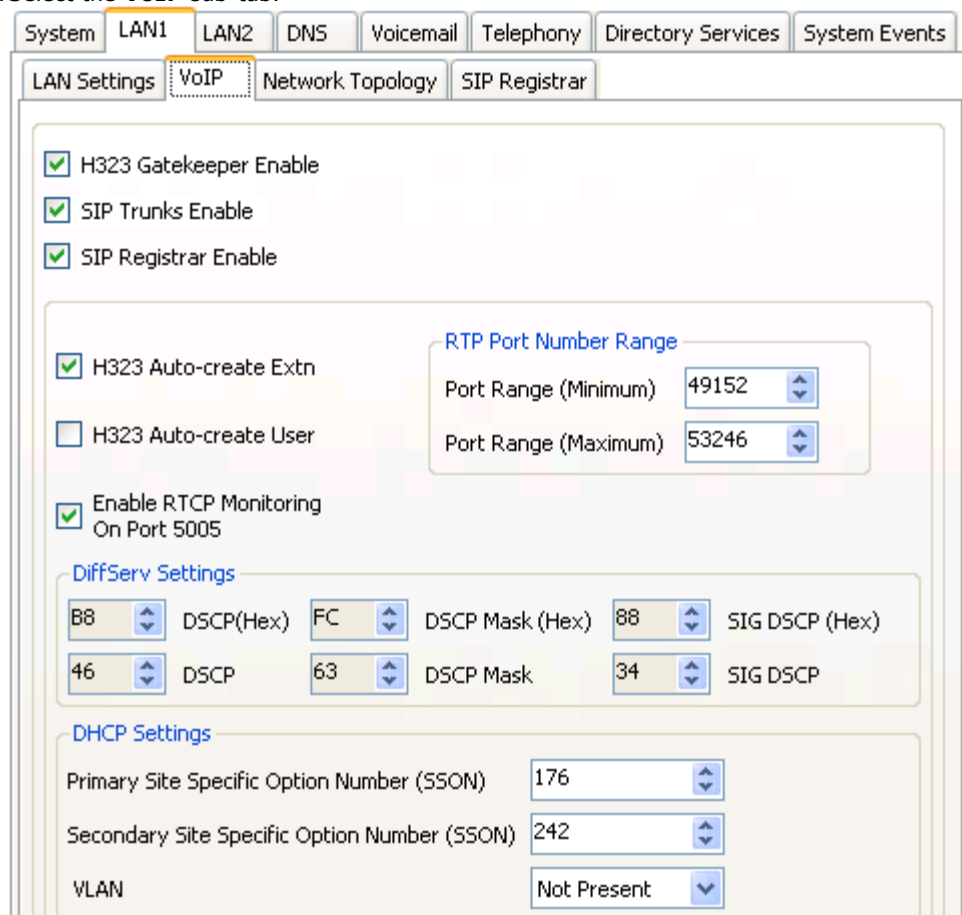




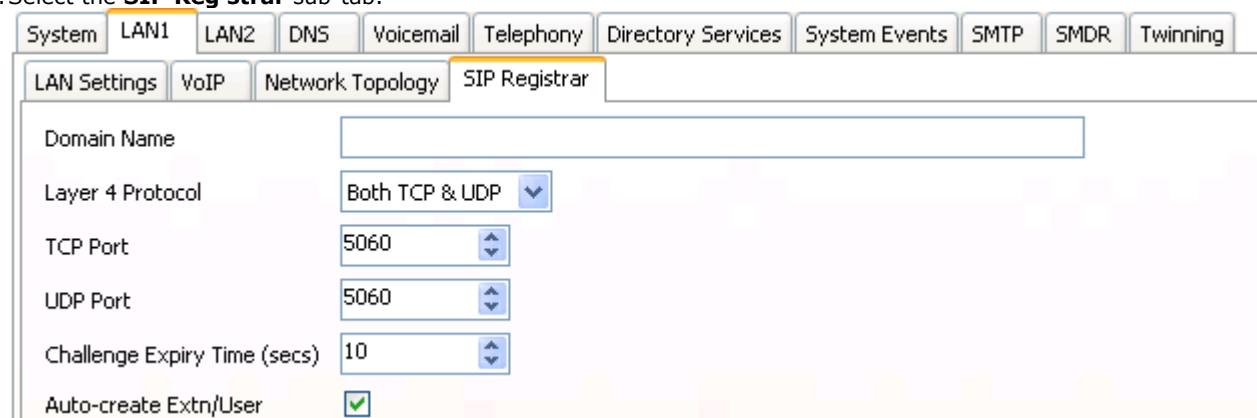
## 1.2 Enabling SIP Extension Support

Once the IP Office system has [valid IP End-points licenses](#)<sup>8</sup>, it can support SIP extensions on its LAN1 and/or LAN2 interfaces.

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 as required.
5. Select the **VoIP** sub-tab.



6. Check that **SIP Reg strar Enable** is selected.
7. Select the **SIP Reg strar** sub-tab.



- **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 Extens on** or [automatically created](#) <sup>16</sup> during SIP device registration. Even if auto-created, the extension settings created in the IP Office configuration should be checked during installation.

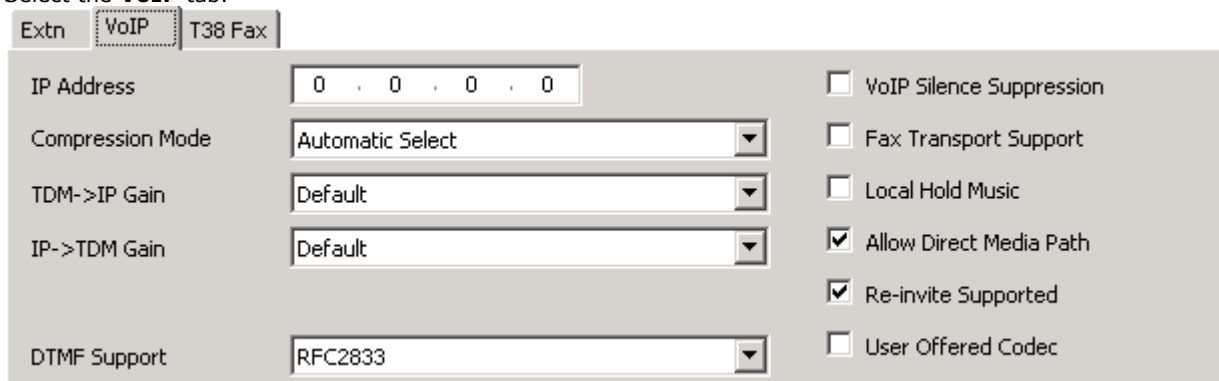
This section looks just at the key configuration settings that affect SIP extension devices.

1. Select  **Extens on** and locate the SIP extension. Select the **Extn** tab.



- **Base Extens on**  
This should match the **Extens on** setting of the SIP user added to the IP Office configuration.
- **Force Author at on: De ut = On**  
If enabled, SIP devices are required to register with the IP Office system using the **Name** and **Log n Code** configured for the user within the IP Office configuration.


2. Select the **VoIP** tab.




- **Com ress on Mode**  
The selected mode must match an audio codec supported by the SIP device. If set to **A tomat c Select**, then the codecs supported by the IP Office are set through the configuration option **System | Tele hony | Tele hony | Automat c 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 Su ort**  
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 D rect Med a Path** allows a VCM channel to be used for DTMF support when necessary.
- **Local Hold Mus c**  
Select this option if the SIP device supports its own hold music source.
- **Re- nv te Su orted**  
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](#) <sup>16</sup> during SIP device registration. Even if auto-created, the user settings created in the IP Office configuration should be checked during installation.

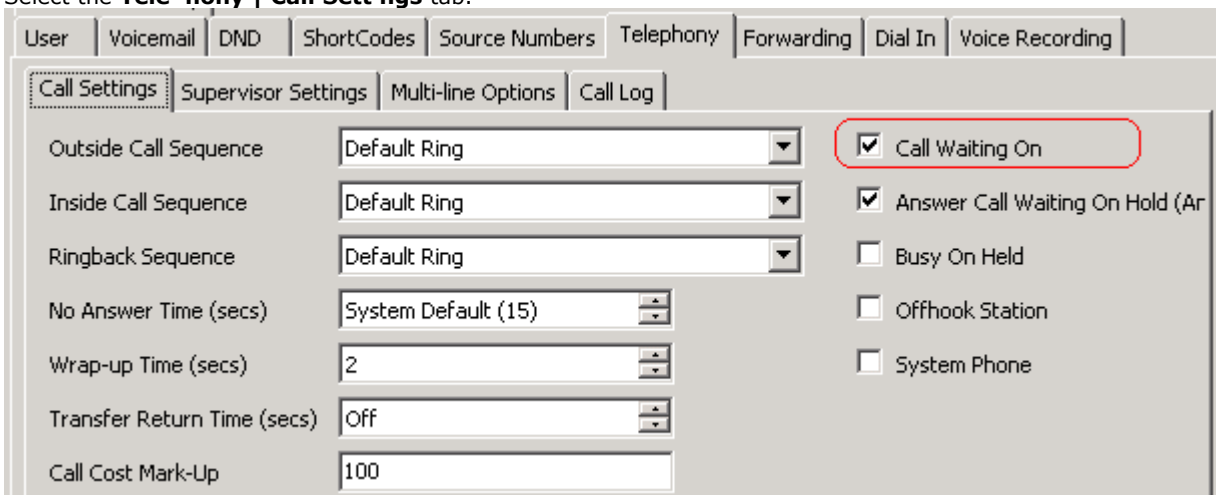
This section looks just at the key configuration settings that affect SIP extension devices.

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



- **Name**  
If the SIP extension is set to **Force Author at on** (the default), this field is used as the **A thor zat on Name** that must be set in the SIP device's configuration.
- **Extens on**  
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 **Tele hony | Call Sett ngs** tab.



- **Call Wa t ng 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.

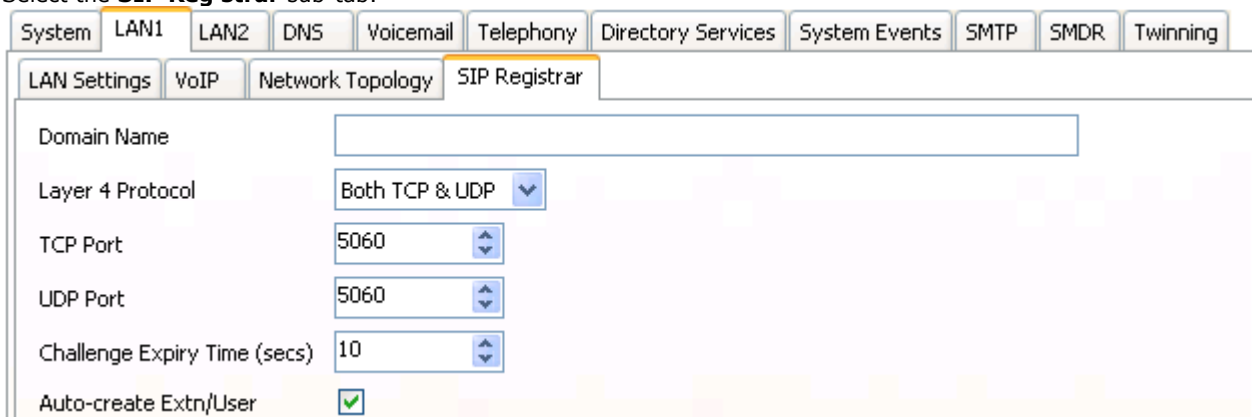
The screenshot shows the 'Supervisor Settings' tab in the SIP User Settings interface. The 'Login Code' field is highlighted with a red rectangle and contains the text '\*\*\*\*'. Below it are fields for 'Login Idle Period (secs)', 'Monitor Group' (set to '<None>'), 'Coverage Group' (set to '<None>'), and 'Status on No-Answer' (set to 'Logged On (No change)'). There is a section for 'Reset Longest Idle Time' with two radio buttons: 'All Calls' (selected) and 'External Incoming'. At the bottom left is 'After Call Work Time (secs)' set to 'System Default (10)'. On the right side, there are several checkboxes: 'Force Login', 'Force Account Code', 'Outgoing Call Bar', 'Inhibit Off-Switch Forward/Transfer', '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 Authorize on** (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 ▲▼

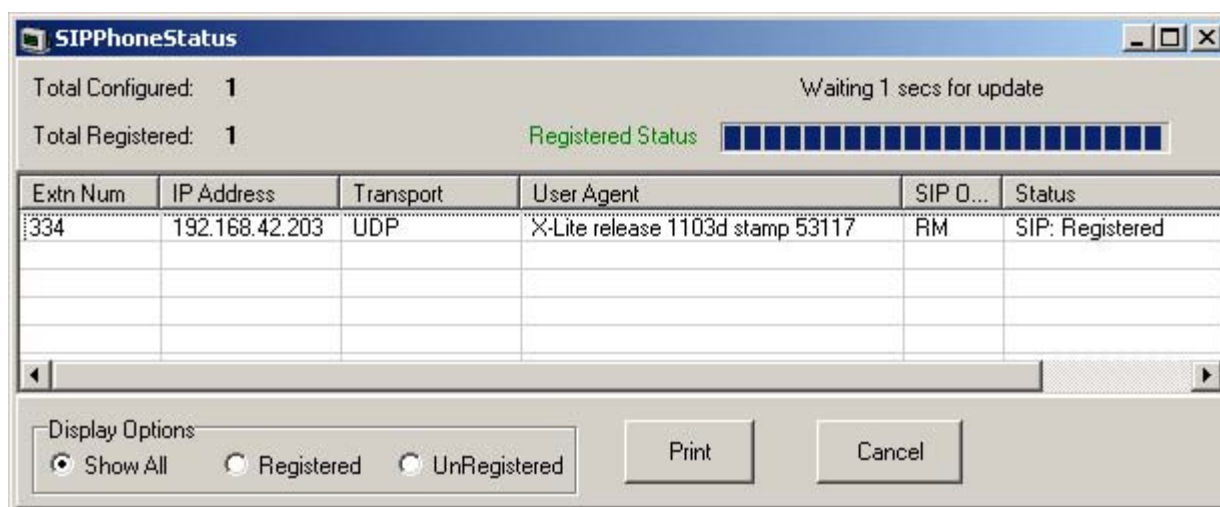
Challenge Expiry Time (secs) 10 ▲▼

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

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:

- [CounterPath Eyebeam/X-Lite Softphones](#) <sup>[20]</sup>
- [Polycom SoundPoint](#) <sup>[23]</sup>
- [Grandstream GXP 2000, GXP 2020](#) <sup>[25]</sup>
- [Avaya A10 ATA](#) <sup>[28]</sup>
- [Patton Micro ATA](#) <sup>[32]</sup>
- [Nokia S60 v3 SIP Client](#) <sup>[33]</sup>
- [Innovaphone IP22, IP24, IP28](#) <sup>[35]</sup>

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



## 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](#) <sup>16</sup> 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....**

Properties of Account 1

Account | Voicemail | Topology | Presence | Advanced

User Details

Display Name: SIPMe

User name: 334

Password: ••••

Authorization user name: Extn334

Domain: 192.168.42.1

Domain Proxy

☒ Register with domain and receive incoming calls

Send outbound via:

☒ domain

☐ proxy Address:

Dialing plan:

OK Cancel Apply

Callouts:

- User | User | Extension
- Extn | Base Extension
- User | Telephony | Call Settings | Login Code
- User | User | Name
- System | LAN | LAN Settings | IP Address

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**.

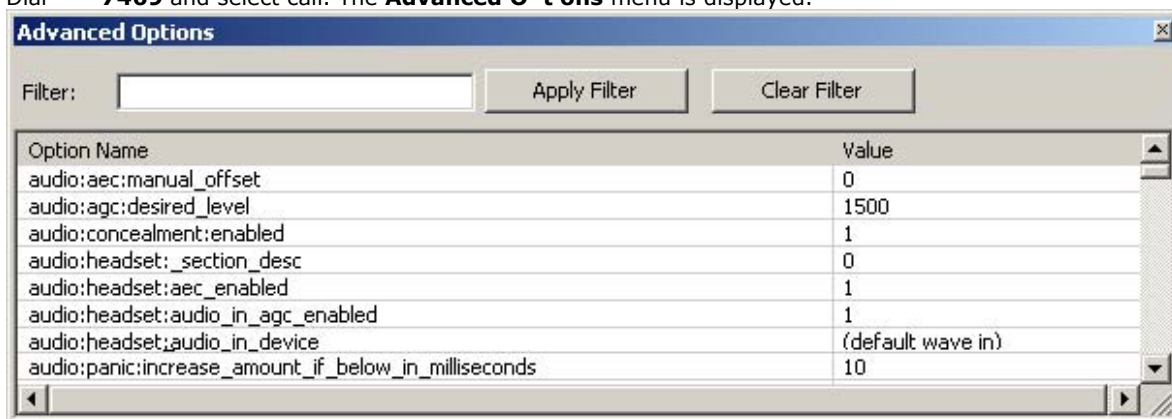
SIP Accounts

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"/>				
<input type="checkbox"/>				

Add... Remove Properties... Make Default Close

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](#) <sup>22</sup> or to perform the following process.

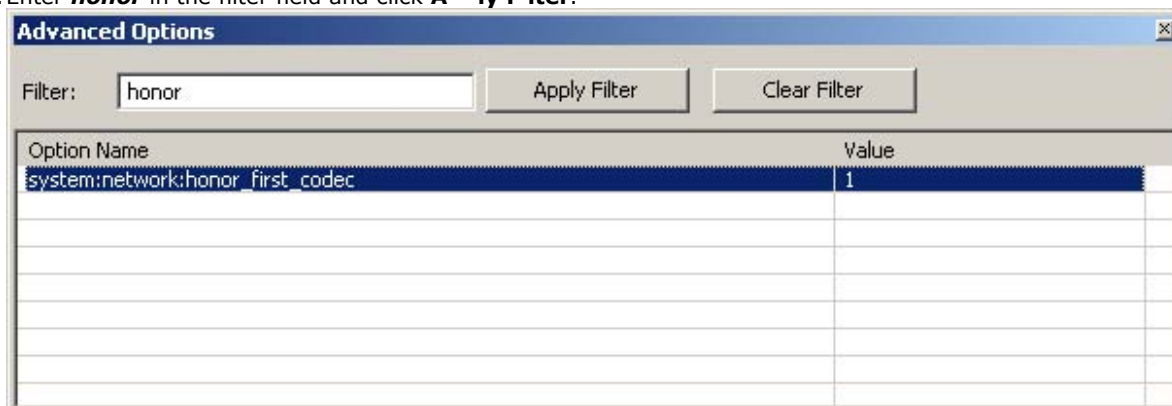
- a. Dial \*\*\*7469 and select call. The **Advanced Options** menu is displayed.



The screenshot shows the 'Advanced Options' window with a filter field and a list of options. The filter field is empty. The list contains the following options and values:

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

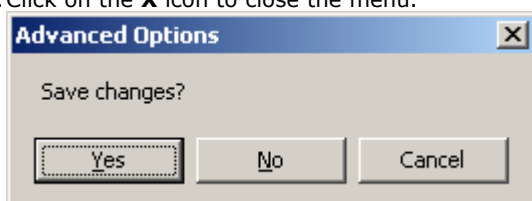
- b. Enter **honor** in the filter field and click **Apply Filter**.



The screenshot shows the 'Advanced Options' window with the filter field set to 'honor'. The list now only contains one option:

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 **X** icon to close the menu.



The screenshot shows a 'Save changes?' dialog box with three buttons: Yes, No, and Cancel. The 'Yes' button is highlighted.

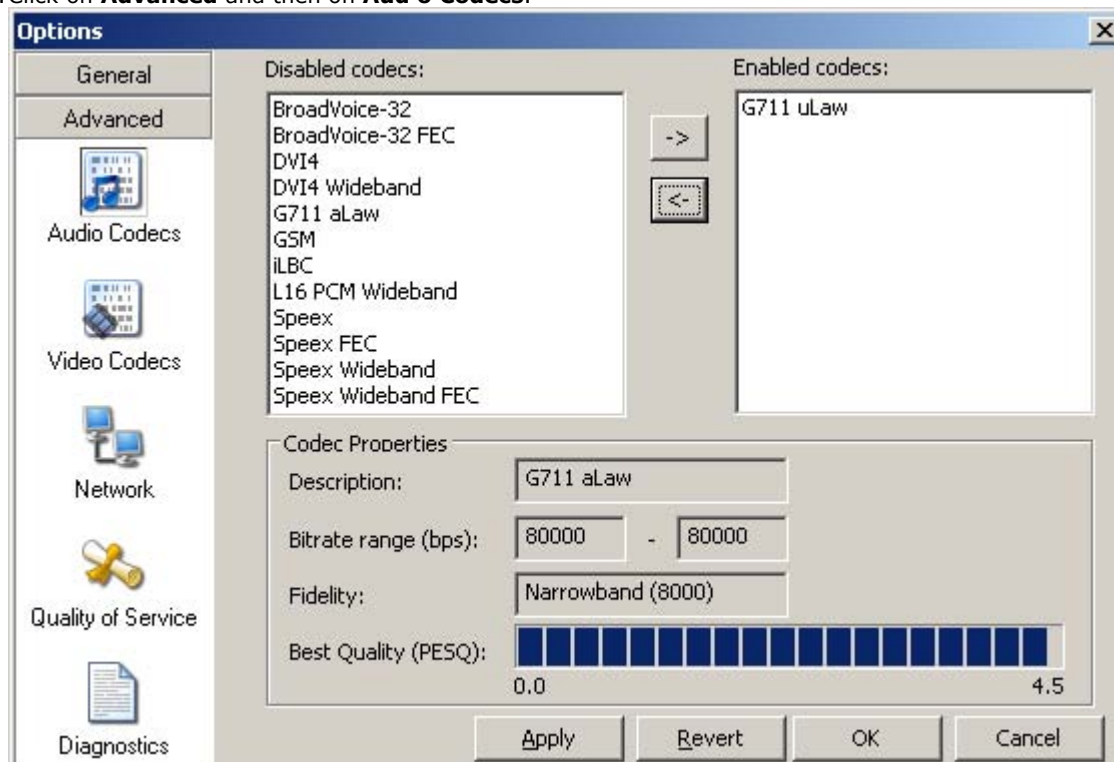
- e. Click on **Yes** to save the change.

- 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 Select on

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: UDPOonly

**Server 1**

Address: 192.168.42.1

Port: 5060

Transport: UDPOonly

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: UDPOonly

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 **adm n**).
2. Click **Log n**. Select **Account 1** or the account that you want to use for IP Office connection.

**Grandstream Device Configuration**

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

**Account Active:** ☐ No ☒ 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 Extn | Base Extension

**Authenticate ID:** Extn4142 User | User | Name

**Authenticate Password:** User | Telephony | Call Settings | Login 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 ☐ 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:** \*17 (UserID for voice mail system)

**Preferred Vocoder:** (in listed order)

choice 1:	G.729A/B	choice 5:	G.726-32
choice 2:	PCMA	choice 6:	iLBC
choice 3:	G.723.1	choice 7:	G.722 (wide band)
choice 4:	PCMU	choice 8:	GSM

**SRTP Mode:** ☒ Disabled ☐ Enabled but not forced ☐ Enabled and forced

**eventlist BLF URI:**

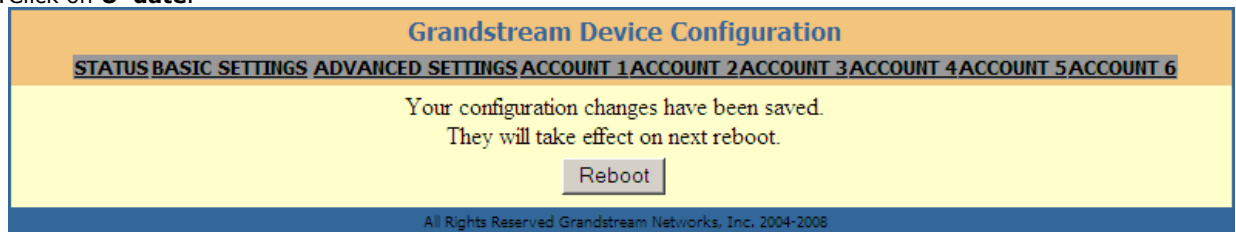
**Special Feature:** Standard

**Update Cancel Reboot**

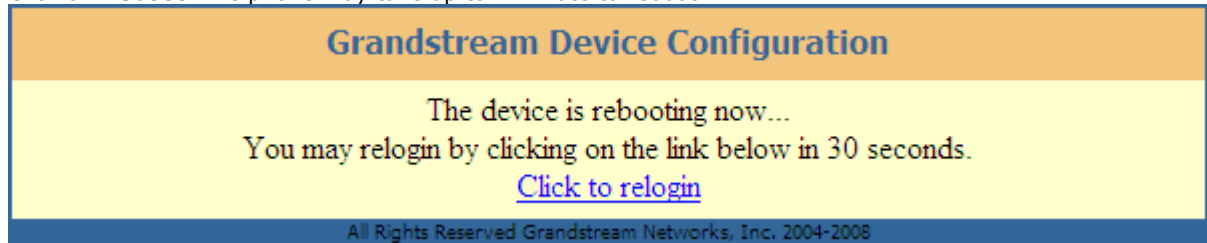
All Rights Reserved Grandstream Networks, Inc. 2004-2008

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

4. Click on **U date**.



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, **D sable 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 **n mdbg** 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	X

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

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

- 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.

192.168.1.1 / Telephony / SIP / Gateway sip / Service default

Configuration **Registration and Authentication**

System | LAN | LAN Settings | IP Address

Registrar ☒ ☐ Ignore redirection of Registrar 192.168.42.1 Host 5060 Port ☐ Register via Default-Server  
☐ Register to redirected Registrar Host Port

Registration Lifetime 300 seconds

Apply ✓

Users To Register

User Name	Register	Display Name	Phone Context	Authenticate	Authentication Name	Password	Default	
338	register	SIP 338	SIP	authenticate	Extn338	*****	default	✗
<input type="text"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text"/>	<input checked="" type="checkbox"/>	<input type="text"/>	.....	<input type="checkbox"/>	

User | User | Extension  
Extn | Base Extension

User | User | Name  
User | Telephony | Call Settings | Login Code

- 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**.

192.168.1.1 / Telephony / Call-Router

Interfaces Routing Tables Functions Services Configuration Active Calls Status

**FXS** H.323 SIP

Name	Bound Port	Routing Destination	
fxs-0	fxs 0 0	to-sip (Table)	✗
fxs-1	fxs 0 1	to-sip (Table)	✗
fxs-2	fxs 0 2	to-sip (Table)	✗
fxs-3	fxs 0 3	to-sip (Table)	✗
<input type="text"/>			

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

192.168.1.1 / Telephony / Call-Router / FXS Interface fxs-0

Configuration Status

Interface (none)

Call-Routing Destination ☒ ☐ Table to-sip ☐ Service (none)

Precall Service ☐ (none)

CID Presentation ☐ (none)

Subscriber Number ☒ 338

Call Hold ☒

Call Waiting ☒

Call Transfer ☐

Additional Call Offering ☒

PSTN Profile default



Tone Profile US

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  arrow icon after **to-s**.


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

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

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


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


192.168.1.1 / Telephony / Call-Router

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-s**.



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

Configuration			
Looks Up For	Destination	Execute Function (Optional)	
called-e164 Of	Interface fxs-0		
called-e164 value or default	<input type="radio"/> Table (none) <input type="radio"/> Service (none) <input type="radio"/> none	Optional function to execute	
338		(none)	
(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**

Configuration			
Looks Up For	Destination	Execute Function (Optional)	
called-e164 Of	fxs-0 (FXS Interface)		
called-e164 value or default	<input type="radio"/> Interface (none) <input type="radio"/> Table (none) <input type="radio"/> Service (none) <input type="radio"/> none	Optional function to execute	
338		(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

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?
<input type="button" value="Save"/> <input type="button" value="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

Network  
IP/DNS  
NAT/IAPT  
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

**Voice** Fax Modem Dejitter Buffer Status

Voice Codecs

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) 101

RTP Payload Type For Signaling Events (NSE) 100

RTP Traffic Class local-voice

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**.

The screenshot shows the 'SIP Configuration' page. On the left is a navigation menu with 'SIP' selected. The main content area has a title 'SIP Configuration' and a subtitle 'SIP Server Settings (Current Server: 192.168.42.1:5060; Domain:; Base RTP Port: 8002)'. Below this are several fields: '\* SIP Registration Server Address' (192.168.42.1), 'SIP Port' (5060), 'SIP Domain' (empty), and 'Voice Port' (8002). A note states: '\* Leaving a setting blank will force the unit to use the information obtained via DHCP and/or DNS'. There are three checkboxes: 'Send Registration Request with Expire Time: 3600' (checked), 'Send Unregistration at boot' (checked), and 'Send SUBSCRIBE' (unchecked). At the bottom is a field for '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**.

The screenshot shows the 'Audio/CODEC Configuration' page. On the left is a navigation menu with 'CODECS' selected. The main content area has a title 'Audio/CODEC Configuration' and a subtitle 'CODECS'. Below this is a table with three columns: 'Selected', 'Silence Suppression', and 'Preferred-Codec'. The table lists five codecs: G711U, G711A, G723, G726, and G729. For each codec, there is a checkbox in the 'Selected' column, a dropdown menu in the 'Silence Suppression' column (all set to 'ON'), and a radio button in the 'Preferred-Codec' column (G729 is selected).

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

6. Select **Phone 1**.

The screenshot shows two pages. The top page is 'User Information' with a subtitle 'User | User | Extension Extn | Base Extension'. It has fields for 'Phone Number' (343), 'CallerID Name' (SIP343), 'User Name' (Extn343), 'Password' (masked with dots), and 'Port' (5060). The 'SIP Registration status' is 'Registered'. The bottom page is 'Voice Mail Setting' with a subtitle 'Voice Mail Setting'. It has a field for '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 Author at on** 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 | Connect on | Settings | New SIP profile**.

2. Enter the following settings:

- **Profile name:** Give the profile a name that indicates its function.
- **Service profile:** Select **IE P**.
- **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 server 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 | Connect on | 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.

Configuration	Info	Admin	License	Update	NTP	Sync	HTTP-Server	HTTP-Client	Logging	SNMP	Telnet	Certificates
<b>General</b>	<b>Version</b> 7.00 hntfix3 IP28[09-703000 11], Rmtcode[09-703000 11], Hardware[402] <b>SerialNo</b> 00-90-33-21-01-7d (9e) <b>DRAM</b> 16 MB <b>FLASH</b> 8 MB <b>Coder</b> 8 Channels of G.711, G.726, G.729 <b>Sync</b> - <b>SNTP Server</b> 135.64.181.220 <b>Time</b> 05.06.2009 07:13 <b>Uptime</b> 17d 11h 37m 29s											

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 **22**, **24** or **28** depending on the unit type.

Configuration	General	Interfaces	SIP	GK	Routes	CDR0	CDR1	Calls	admin	Help						
<b>General</b>	Call Logging <input type="checkbox"/> Route Logging <input type="checkbox"/> Billing CDRs only <input type="checkbox"/> Logging Filter(GW:Nr) : Licenses <table border="1"> <thead> <tr> <th>Name</th> <th>Count</th> <th>Usage</th> </tr> </thead> <tbody> <tr><td> </td><td> </td><td> </td></tr> </tbody> </table> OK Cancel										Name	Count	Usage			
Name	Count	Usage														

4. Select **Interfaces**.

Configuration	General	Interfaces	SIP	GK	Routes	CDR0	CDR1	Calls	admin	Help																																																																																																								
<b>General</b>	<table border="1"> <thead> <tr> <th>Interface</th> <th>CGPN-In</th> <th>CDPN-In</th> <th>CGPN-Out</th> <th>CDPN-Out</th> <th>State</th> <th>Alias</th> <th>Registration</th> </tr> </thead> <tbody> <tr><td>TEL1</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEL2</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEL3</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEL4</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEL5</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEL6</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEL7</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEL8</td><td>+</td><td></td><td></td><td></td><td>Up</td><td></td><td></td></tr> <tr><td>TEST</td><td>+</td><td></td><td></td><td></td><td></td><td></td><td></td></tr> <tr><td>TONE</td><td>+</td><td></td><td></td><td></td><td></td><td></td><td></td></tr> <tr><td>HTTP</td><td>+</td><td></td><td></td><td></td><td></td><td></td><td></td></tr> <tr><td>ECHO</td><td>+</td><td></td><td></td><td></td><td></td><td></td><td></td></tr> </tbody> </table>										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	+						
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.

Configuration	General	Interfaces	SIP	GK	Routes	CDR0	CDR1	Calls	admin	Help
<b>General</b>	Name Disable <input type="checkbox"/> Tones EUROPE-PBX Interface Maps Manual Internal Registration Protocol None Feature Codes Support <input type="checkbox"/> (with Feature Codes) Dynamic Group Direct Dial Locked White List OK Cancel Apply Delete Help									

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

☐ TEL2

☐ RAB2

☐ TEL3

☐ RAB3

☐ TEL4

☐ RAB4

☐ TEL5

☐ RAB5

☐ TEL6

☐ RAB6

☐ TEL7

☐ RAB7

☐ TEL8

☐ RAB8

☐ TEST

☐ TONE

☐ HTTP

☐ ECHO

☐ SIP1

☐ SIP2

☐ SIP3

☐ SIP4

☐ GW1

☐ GW2

☐ GW3

☐ GW4

☐ GW5

☐ GW6

Add UII

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										

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.





Performance figures and data quoted in this document are typical, and must be specifically confirmed in writing by Avaya before they become applicable to any particular order or contract. The company reserves the right to make alterations or amendments to the detailed specifications at its discretion. The publication of information in this document does not imply freedom from patent or other protective rights of Avaya or others.

Intellectual property related to this product (including trademarks) and registered to Lucent Technologies have been transferred or licensed to Avaya.

All trademarks identified by the ® or ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners.

This document contains proprietary information of Avaya and is not to be disclosed or used except in accordance with applicable agreements.

Any comments or suggestions regarding this document should be sent to "wgctechpubs@avaya.com".

© 2010 Avaya Inc. All rights reserved.

Avaya  
Unit 1, Sterling Court  
15 - 21 Mundells  
Welwyn Garden City  
Hertfordshire  
AL7 1LZ  
England.

Tel: +44 (0) 1707 392200  
Fax: +44 (0) 1707 376933

Web: <http://marketingtools.avaya.com/knowledgebase>