IP Office

SIP Extension Support

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Chapter 1.
IP Office SIP Extensions

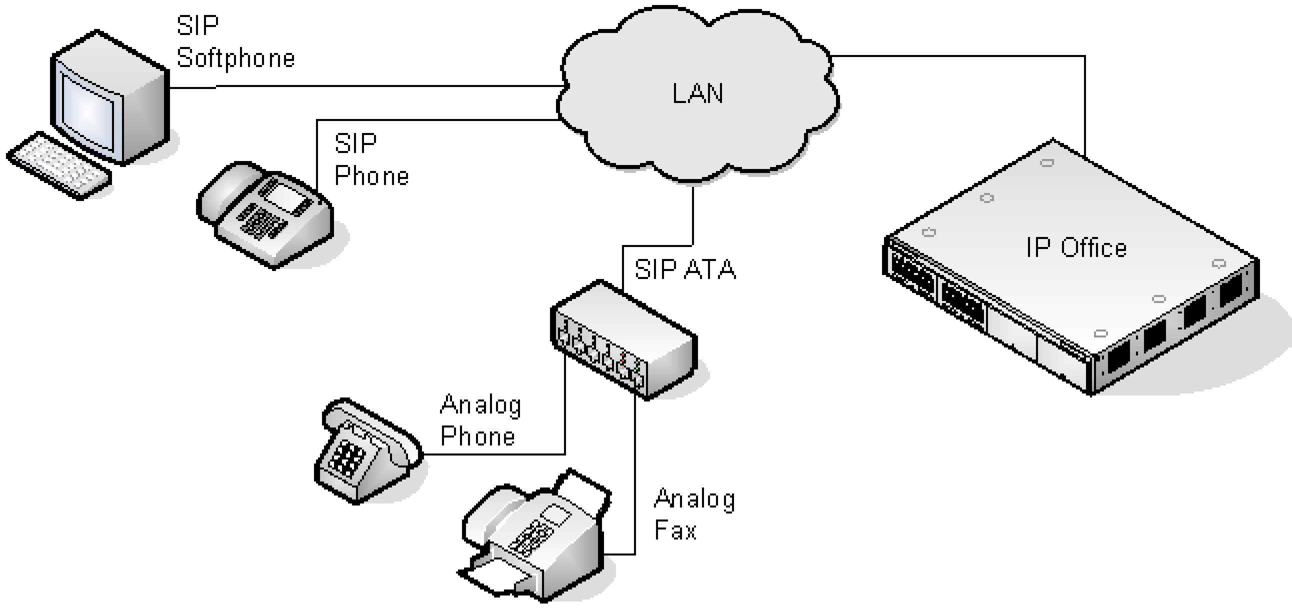
1. IP Office SIP Extensions

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- o nts** 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.

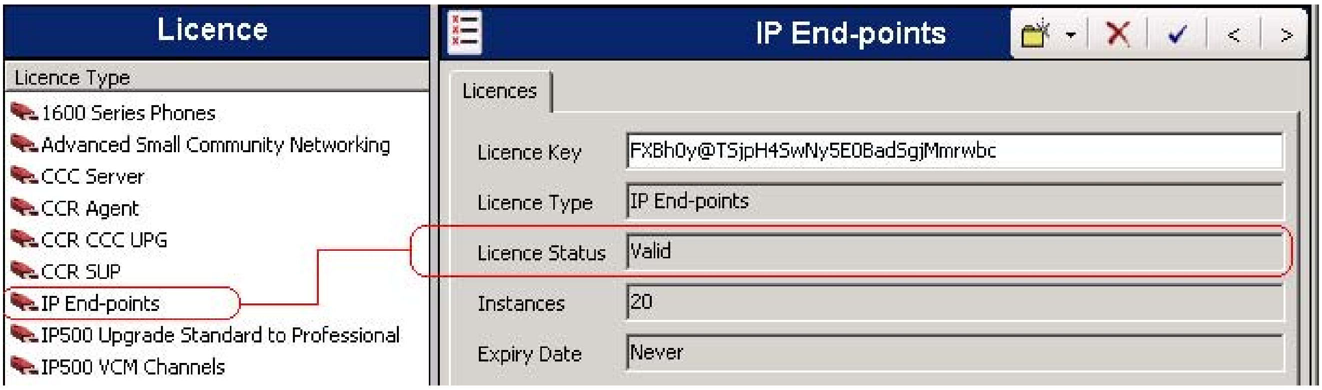


- **No NAT**
  Connection of SIP extension devices from locations where Network Address Translation (NAT) is applied to the connection is not supported. The IP Office does not provide NAT traversal services (for example STUN or TURN) for SIP extension devices.
- **Mult le L ne SIP Dev ces**
  Some SIP 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 Off ce s the SIP Reg strar 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 Off ce Vo ce Com ress on = SIP Aud o 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 Off ce Call Wa t ng = 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.**
  - **Make calls.**
  - **Hang U .**
  - **Hold.**
  - **Unsu erv sed Transfer.**
  - **Su erv sed Transfer.**
  - **Vo cema l Collect.**
  - **Set Forward ng/DND.**
  - **Park/Un ark.**

1.1 Licensing

SIP Extensions are within the IP Office configuration use **IP End- o nts** licenses. Successful registration consumes one license count.

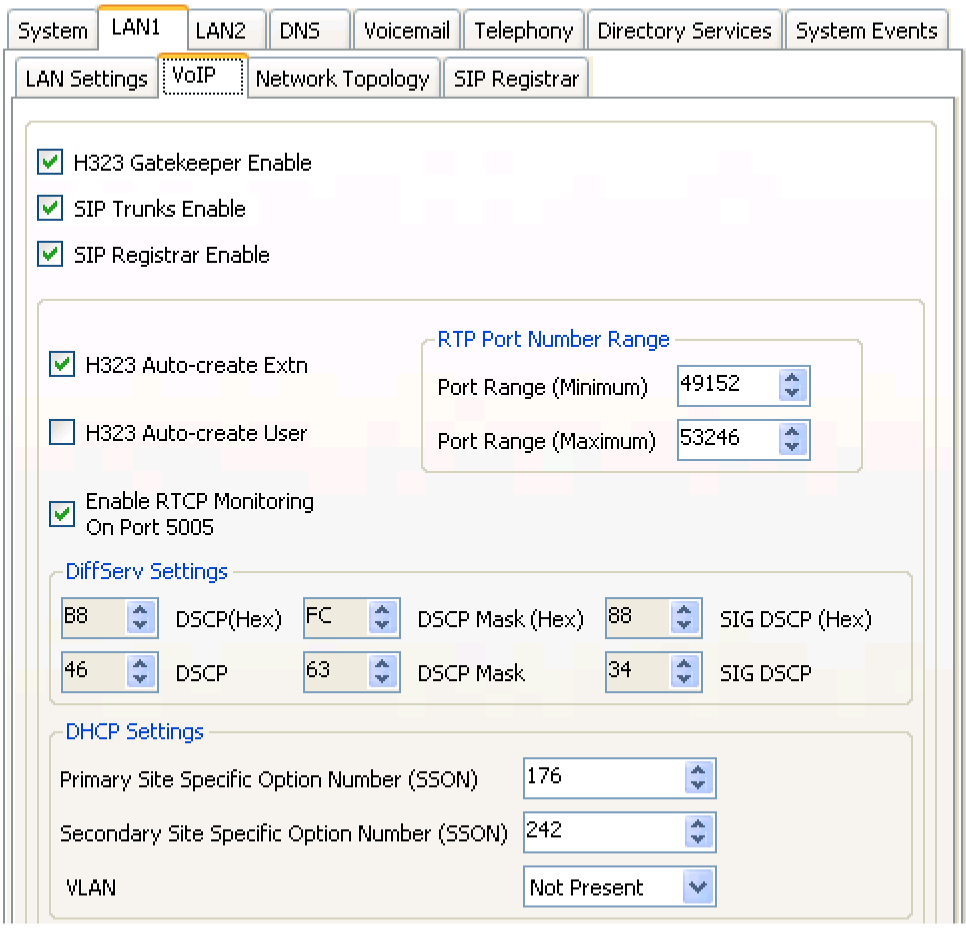
This license is also used for non-Avaya H323 IP extensions. There must be sufficient licenses for the number of extensions required.



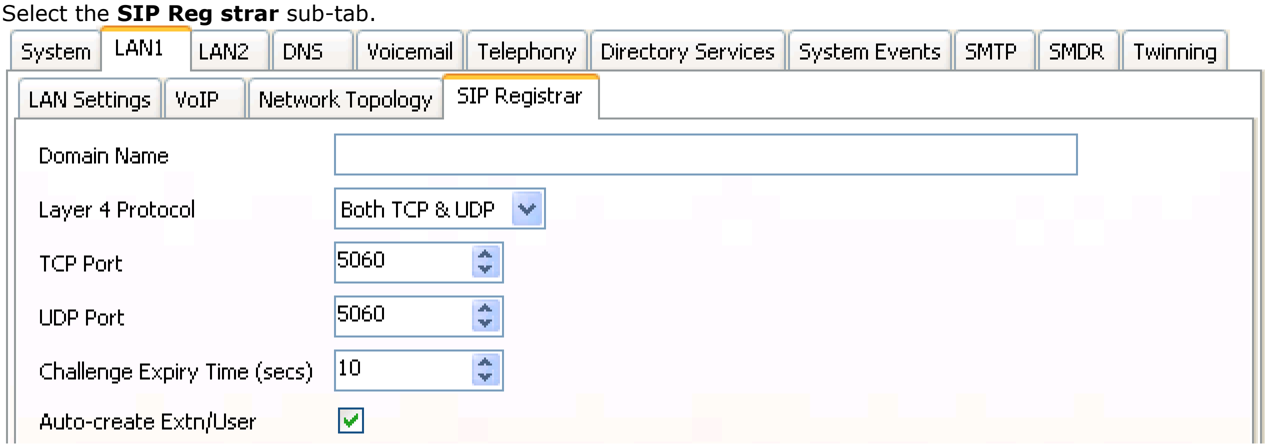
1.2 Enabling SIP Extension Support

Once the IP Office system has valid IP End-points licenses 8, it can support SIP extensions on its LAN1 and/or LAN2 interfaces.

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.



- **Doma n Name:** *De u t = B nk*
  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:** *De u t = B t TCP & DP*
  The transport protocol for SIP traffic between the IP Office and SIP extension devices. Both TCP and/or UDP can be used.

- **TCP Port:** *De u t = 5060*
  The SIP port if using TCP. The default is 5060.

- **UDP Port:** *De u t = 5060*
  The SIP port if using UDP. The default is 5060.

- **Challenge Ex ry T me (sec):** *De u t = 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:** *De u t = 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.

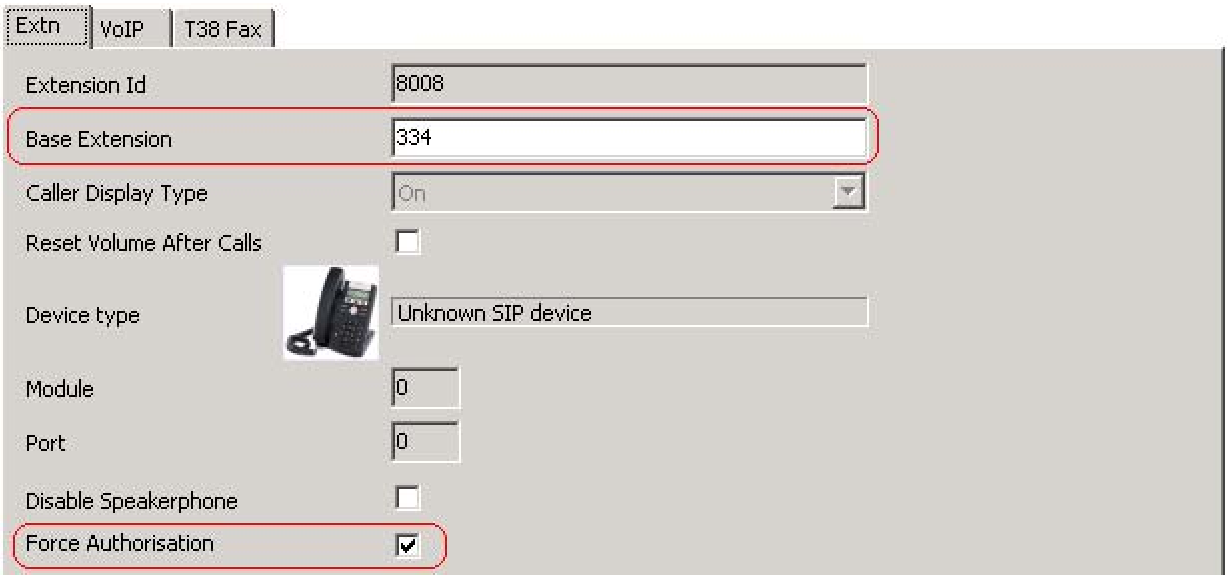
8. Send the configuration back to the IP Office.

1.3 SIP Extension Settings

SIP extensions can be created manually using **| SIP Extens on** or [automatically created] 16 during SIP device registration. Even if auto-created, the extension settings created in the IP Office configuration should be checked during installation.

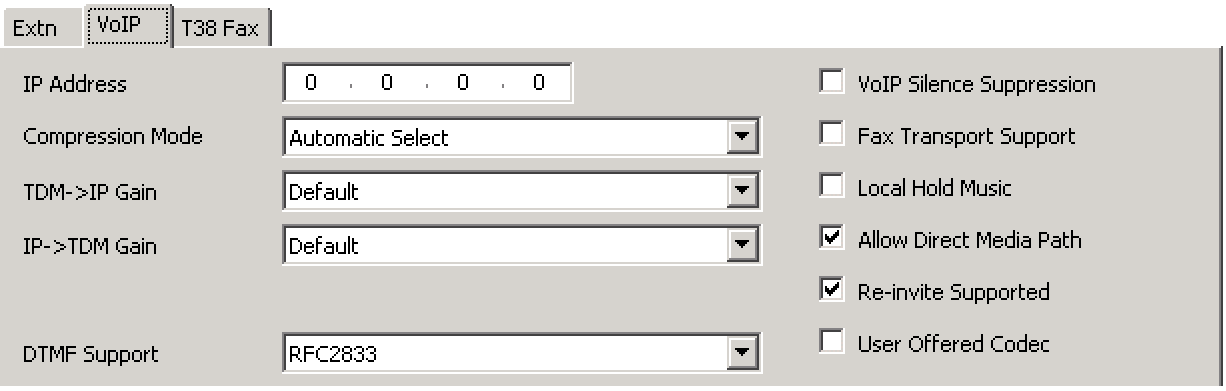
This section looks just at the key configuration settings that affect SIP extension devices.

1. Select **Extens ons** 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 u t = 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.



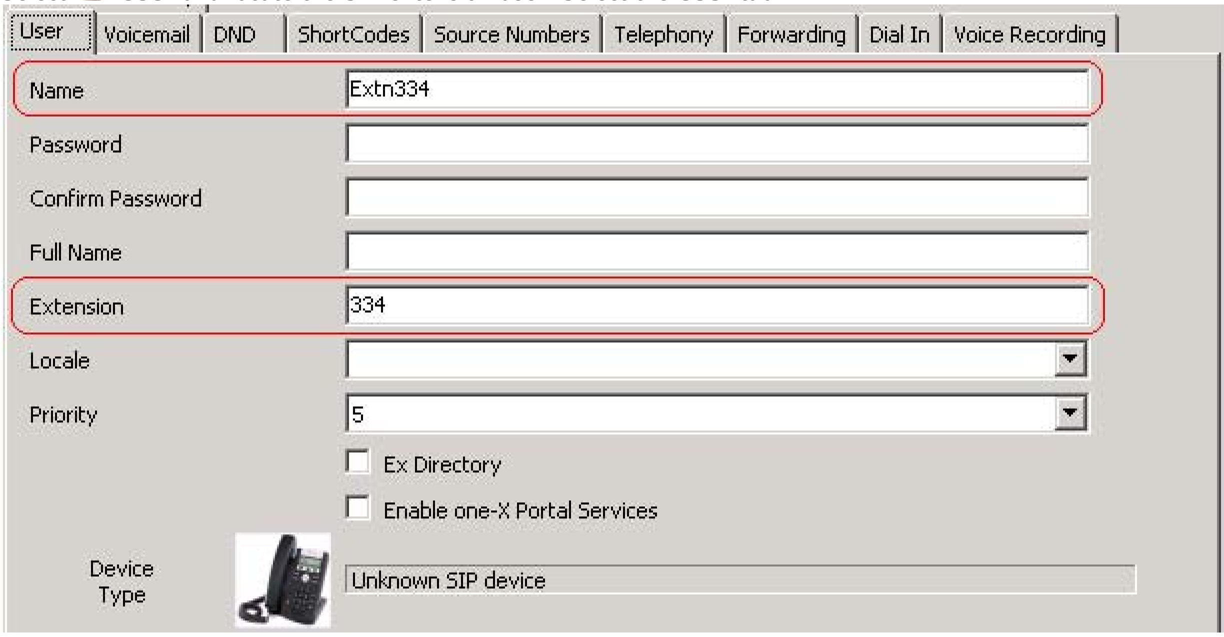
   - **Com ress on Mode**
     The selected mode must match an audio codec supported by the SIP device. If set to ***A tomat c Select***, then the codecs supported by the IP Office are set through the configuration option **System | Tele hony | Tele hony | Automat c Codec Preferences**.
     - **User Offered 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] 16 during SIP device registration. Even if auto-created, the user settings created in the IP Office configuration should be checked during installation.

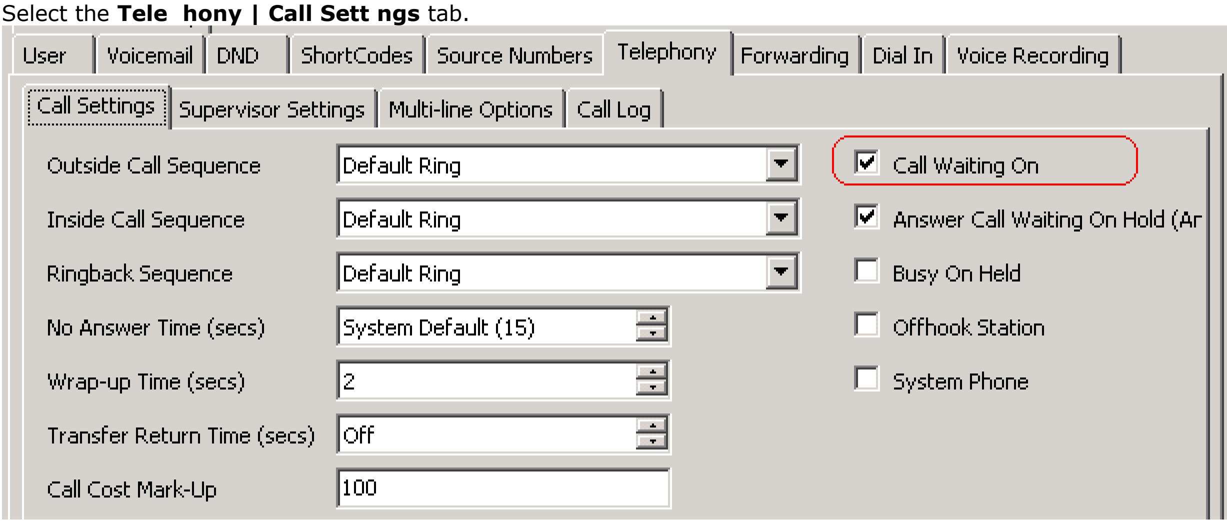
This section looks just at the key configuration settings that affect SIP extension devices.

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



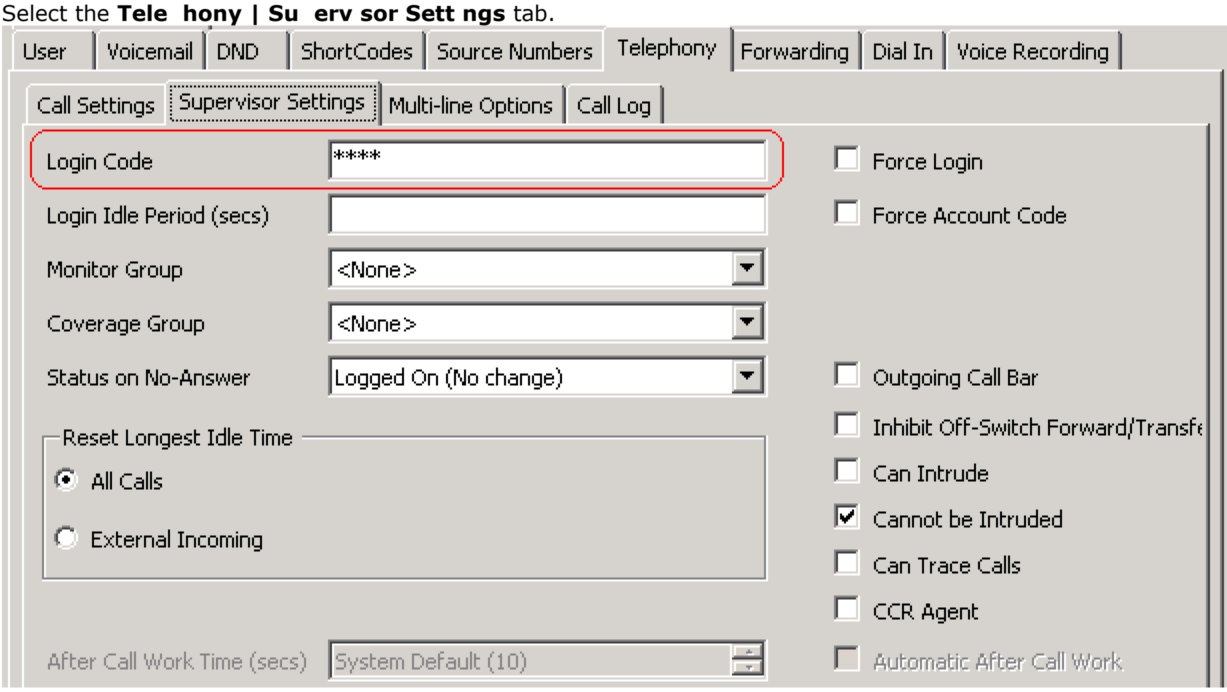
   - **Name**
     If the SIP extension is set to **Force Author at on** (the default), this field is used as the ***A thor zat on Name*** that must be set in the SIP device's configuration.
   - **Extens on**
     This should match the SIP ID of the SIP device and the Base Extension setting of the SIP 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.

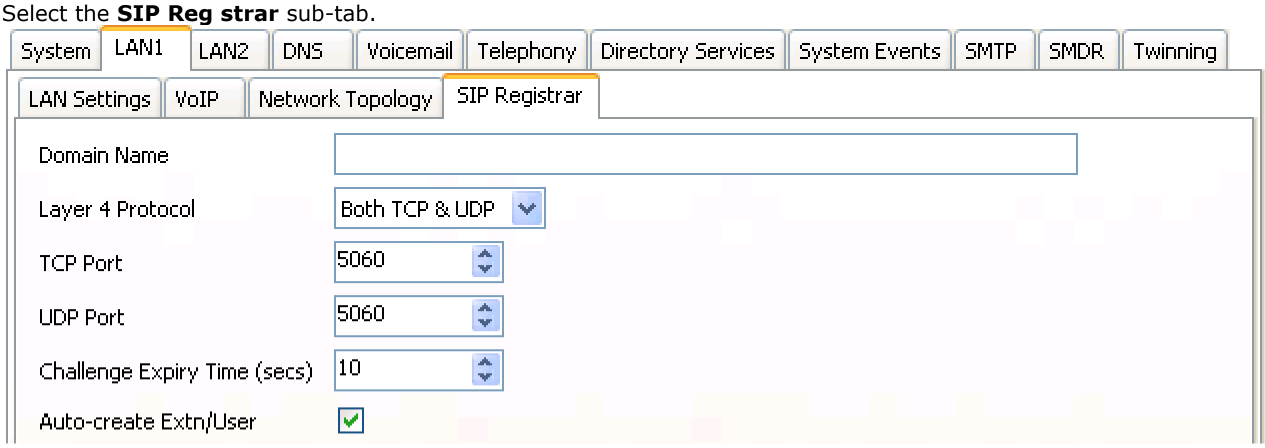
3. Select the **Tele hony | Su erv sor Sett ngs** tab.



- **Log n Code**
  If the SIP extension is set to **Force Author at on** (the default), this field is used as the ***A thor zat on Password*** that must be set in the SIP device's configuration.

1.5 Allowing SIP Extn/User Auto Creation

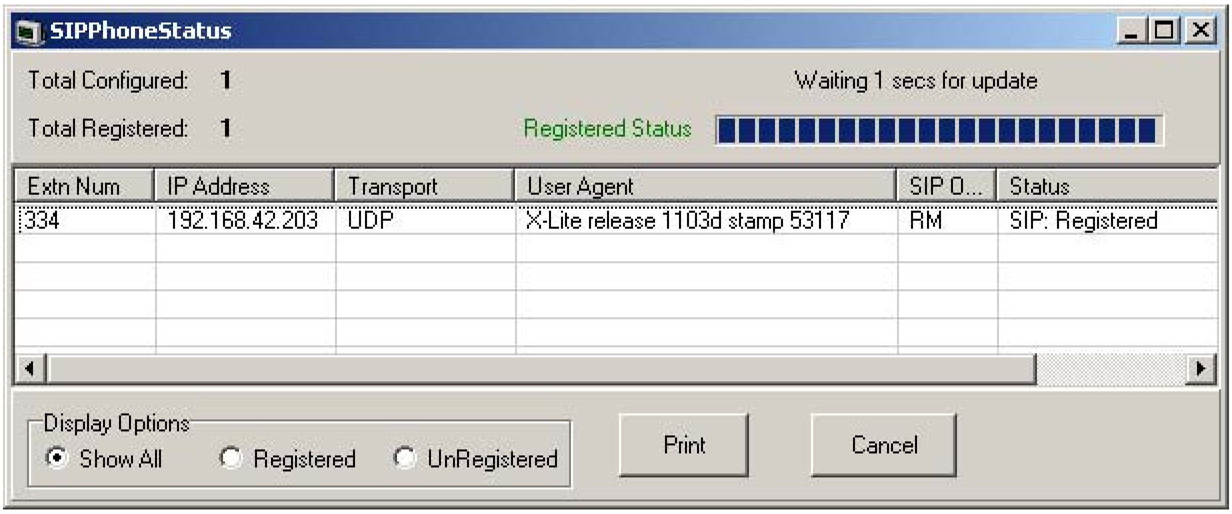
1. Note that changing the SIP registrar settings of an IP Office system requires the IP Office system to be rebooted.
2. Using IP Office Manager, receive the IP Office system configuration.
3. Select **System**.
4. Select either the **LAN1** or **LAN2** tab on which the SIP registrar is enabled.
5. Select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



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.



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-L te Soft hones** 20
- **Polycom Sound o nt** 23
- **Grandstream GXP 2000, GXP 2020** 25
- **Avaya A10 ATA** 28
- **Patton M cro ATA** 32
- **Nok a S60 v3 SIP Cl ent** 33
- **Innova hone IP22, IP24, IP28** 35

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

Us ng Auto Create	Us ng Manual Conf gurat on
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.	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/.

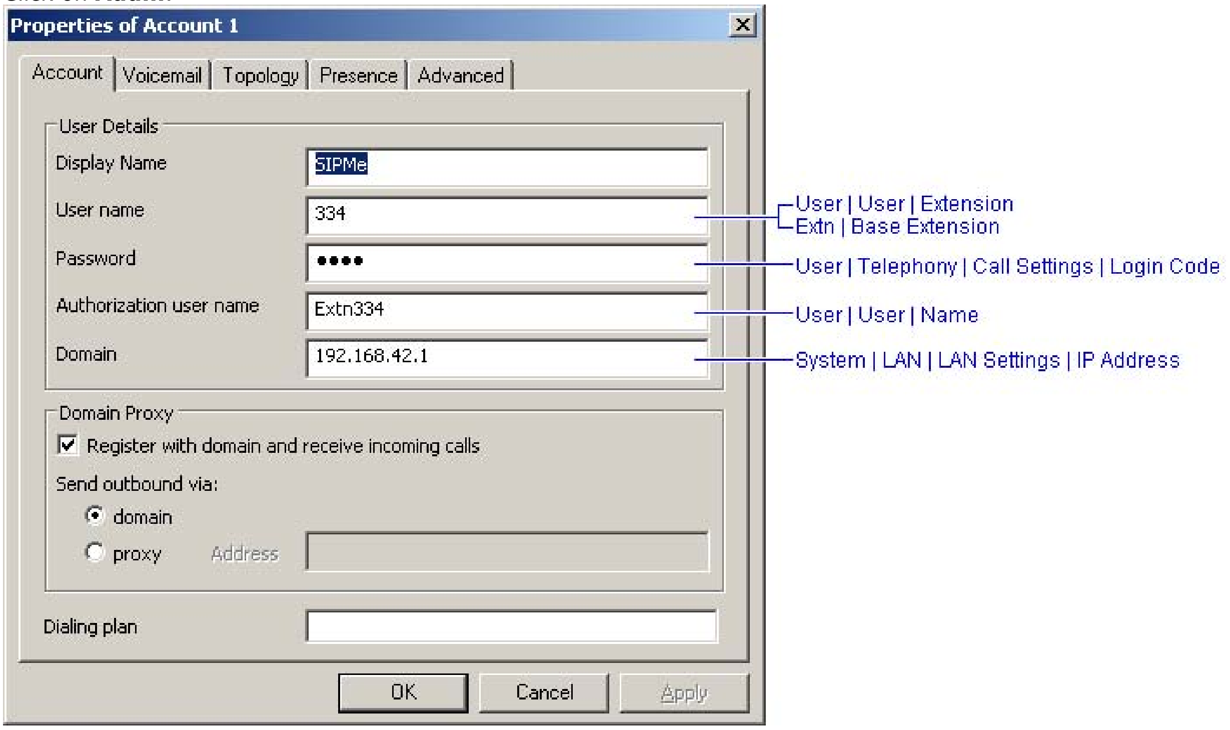
A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.

1. Either enable the IP Office to allow automatic creation 16 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 Sett ngs...**.

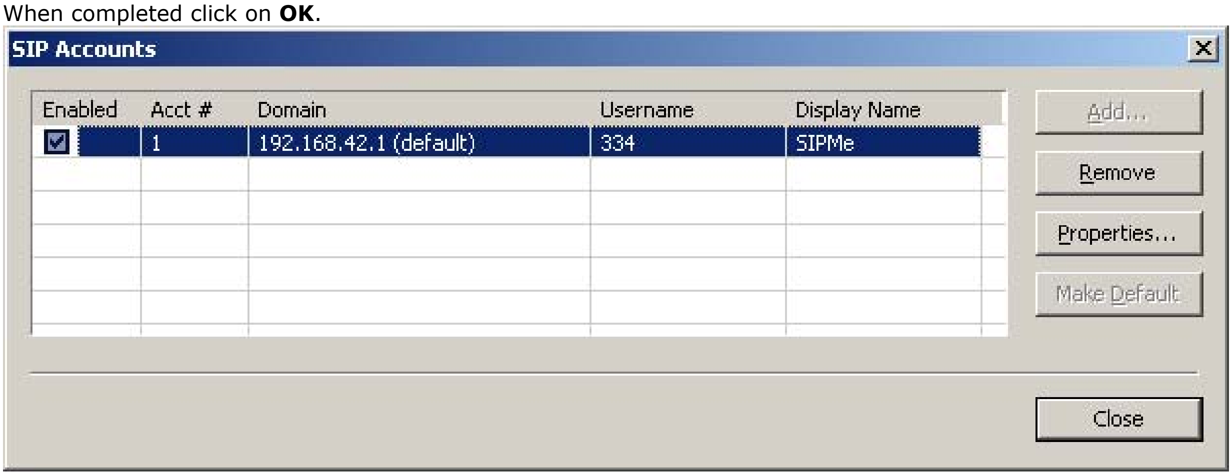
4. Click on **Add...**.



5. Set the fields to match the IP Office configuration settings are indicated above.

6. In the **Doma n Proxy** section enable **Reg ster w th doma n and rece ve ncom ng calls** and select **doma n**.

7. When completed click on **OK**.

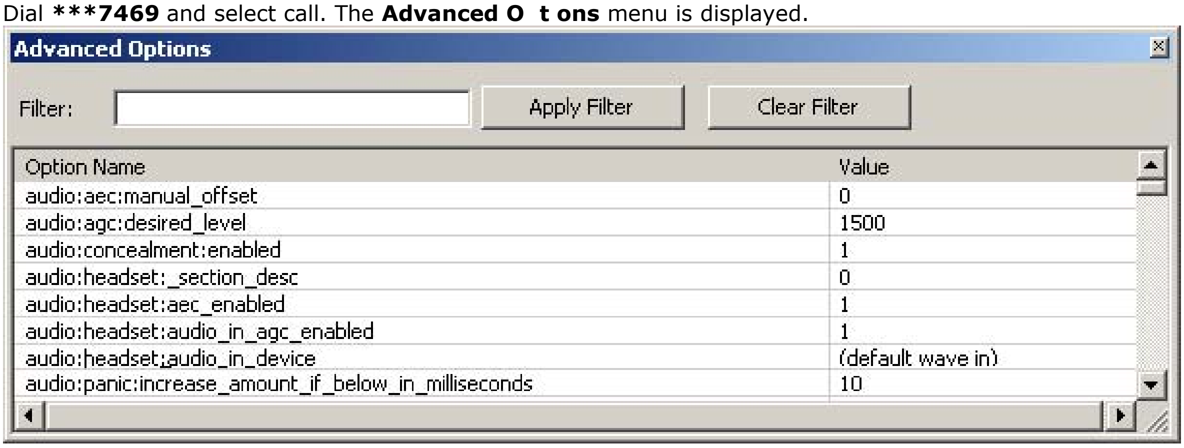


8. Ensure the the account is **Enabled**.

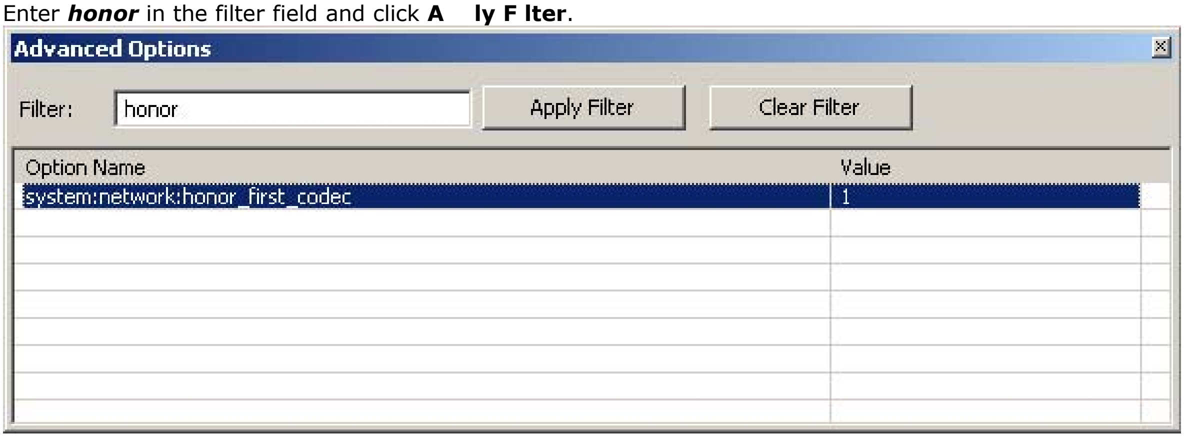
9. Click **Close**. The X-Lite client will now attempt to register with the IP Office. The success or failure of that process will be displayed by the client.

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 22 or to perform the following process.

a. Dial *****7469** and select call. The **Advanced O t ons** menu is displayed.

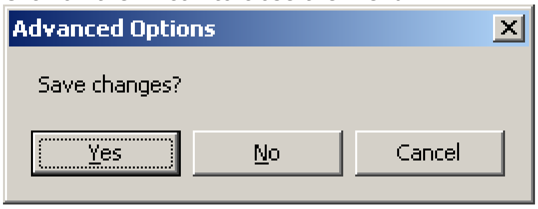


b. Enter ***honor*** in the filter field and click **A   ly F lter**.



c. Set the value for **system:network:honor f rst codec** to ***1***.

d. Click on the **X** icon to close the menu.



e. Click on   **es** 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.

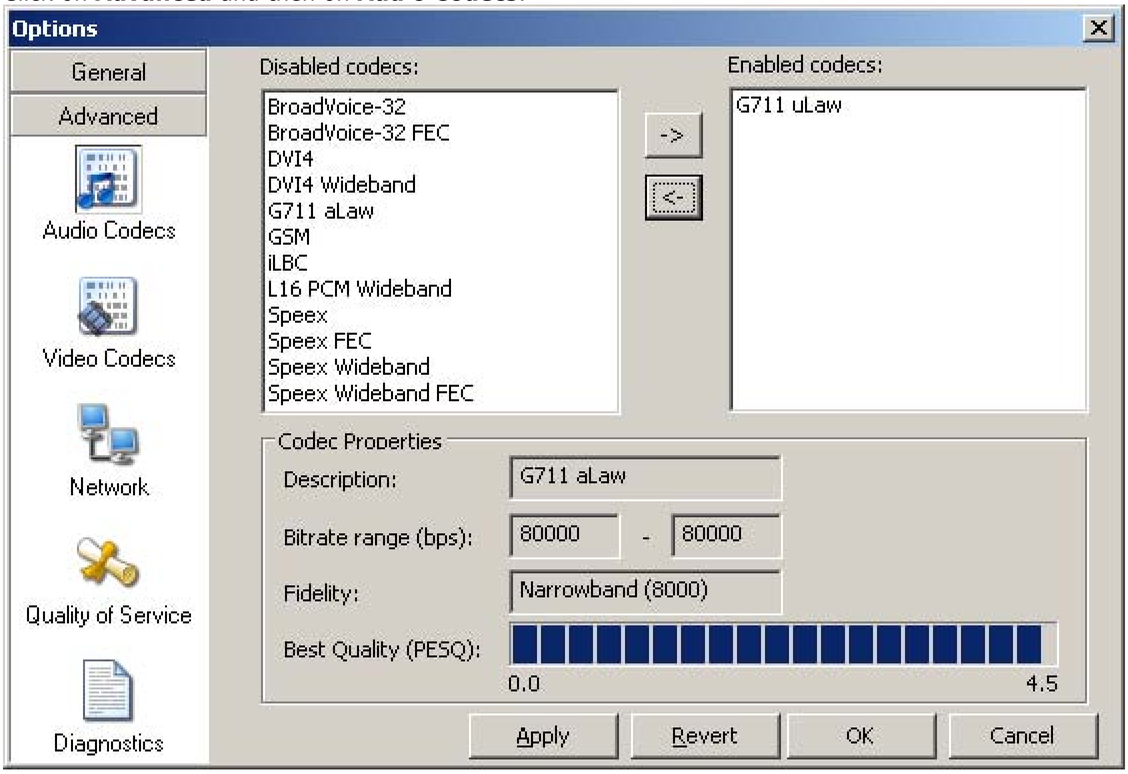
C. Make test calls from and to the SIP device.

D. If not installing any further SIP devices, **D sable 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 **O t ons**.
2. Click on **Advanced** and then on **Aud o 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**.

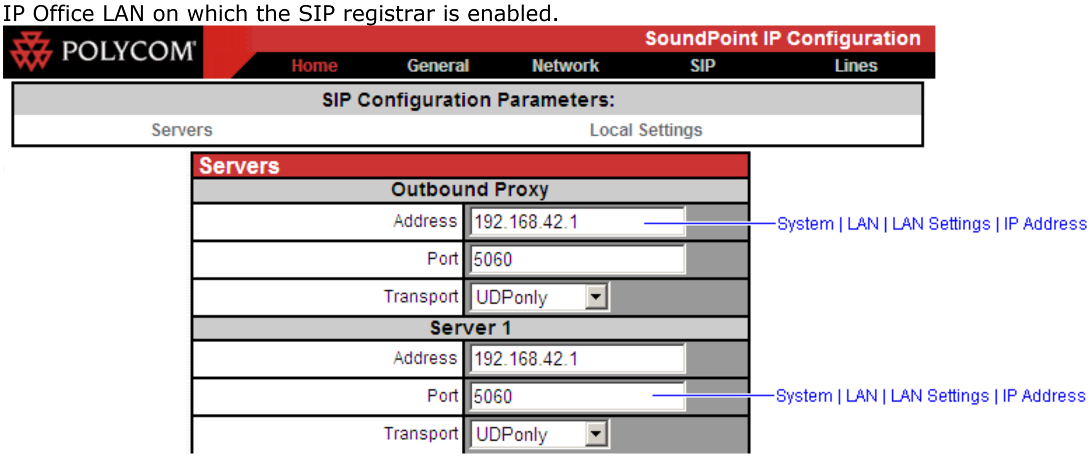
2.2 Polycom SoundPoint Phones

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.

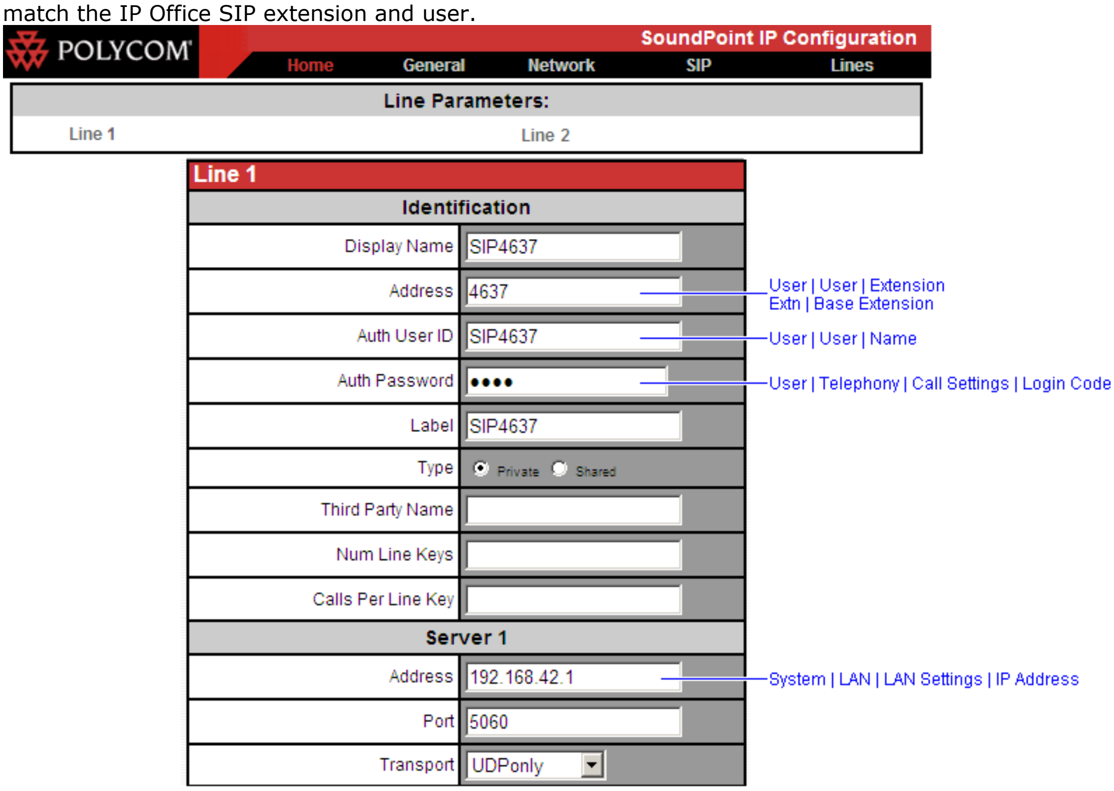
2. Select **SIP**. ou will be requested to enter the administrator name and password. The default values are ***Polycom*** and ***456***.

3. in the **Outbound Proxy** and **Server 1** sections, set the **Address**, **Port** and **Trans ort** details to match the IP Office LAN on which the SIP registrar is enabled.



4. Click **Subm t**. 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 **L nes**. In the Line 1 section, enter the details to match the IP Office SIP extension and user.



6. Click **Subm t**. The phone will reset and load the new settings. That will take up to 2 minutes.

7. Select **Network** and then **Aud o Process ng**. Check that the codecs match those configured for the SIP extension on the IP Office. If you make any changes click **Subm t** 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.

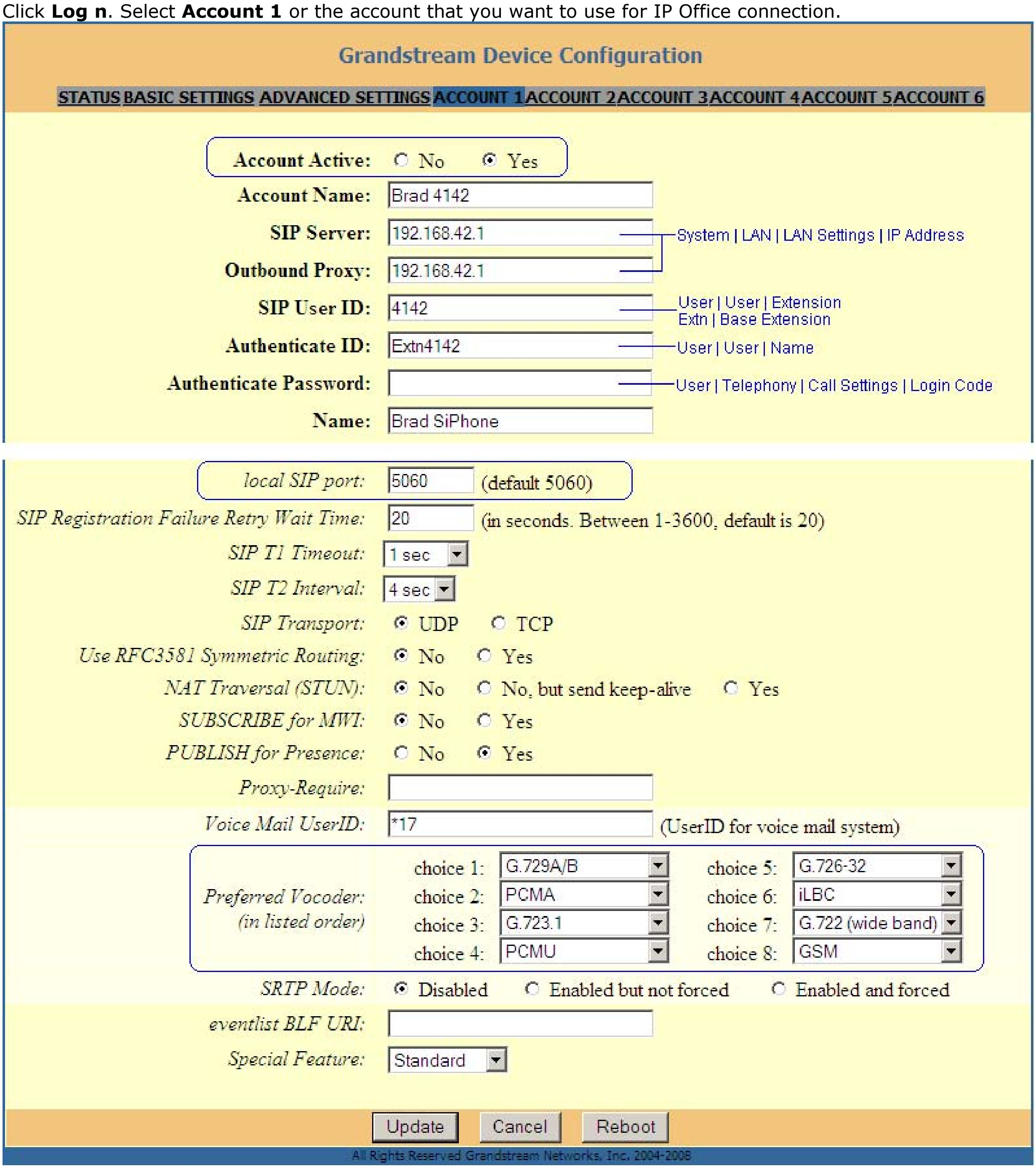
D. If not installing any further SIP devices, **D sable Auto-Create Extn/User** if it is enabled.

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.

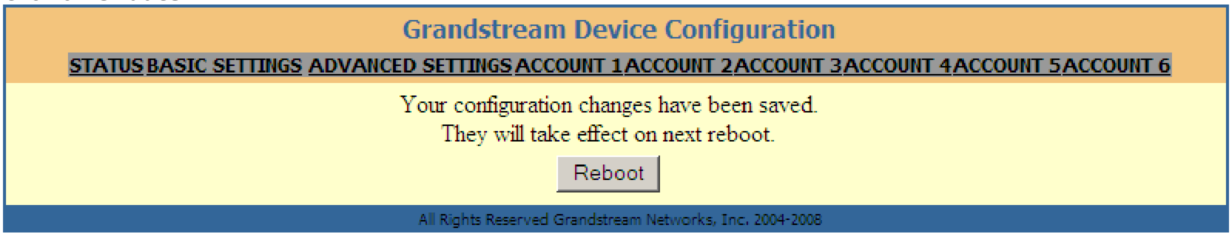
A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.

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.

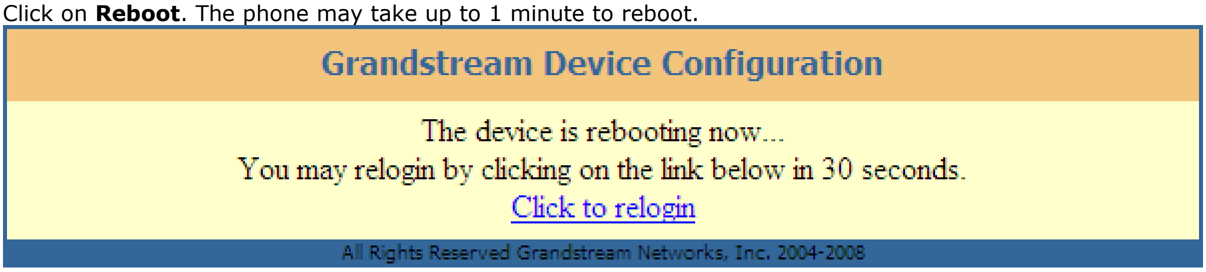


3. Set the fields indicated above to match those required for the IP Office system.

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.

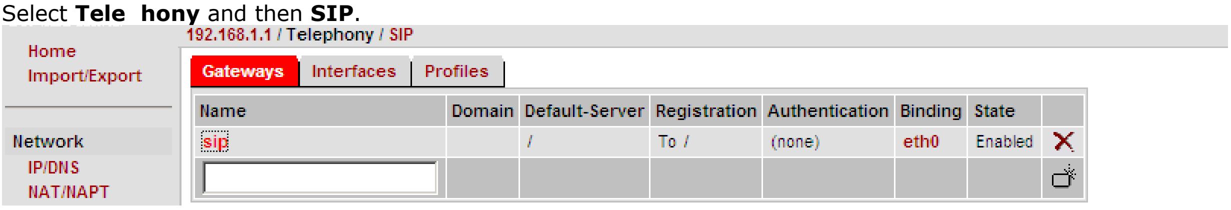
D. If not installing any further SIP devices, **D sable Auto-Create Extn/User** if it is enabled.

2.4 Avaya A10 ATA

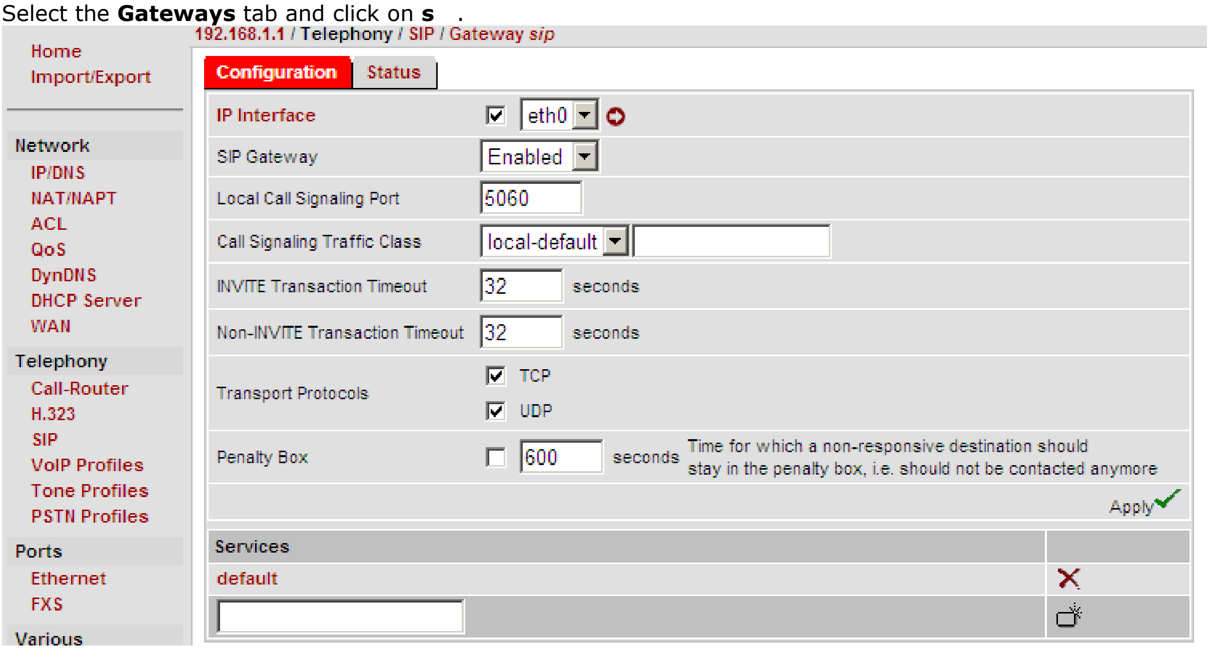
The Avaya A10 Analog Telephone Adapter provides 4 Phone/FXS ports on its rear plus a LAN port. It can be used to connect analog phone devices to the IP Office via the LAN, with the extensions appearing in the IP Office configuration as SIP extensions.

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

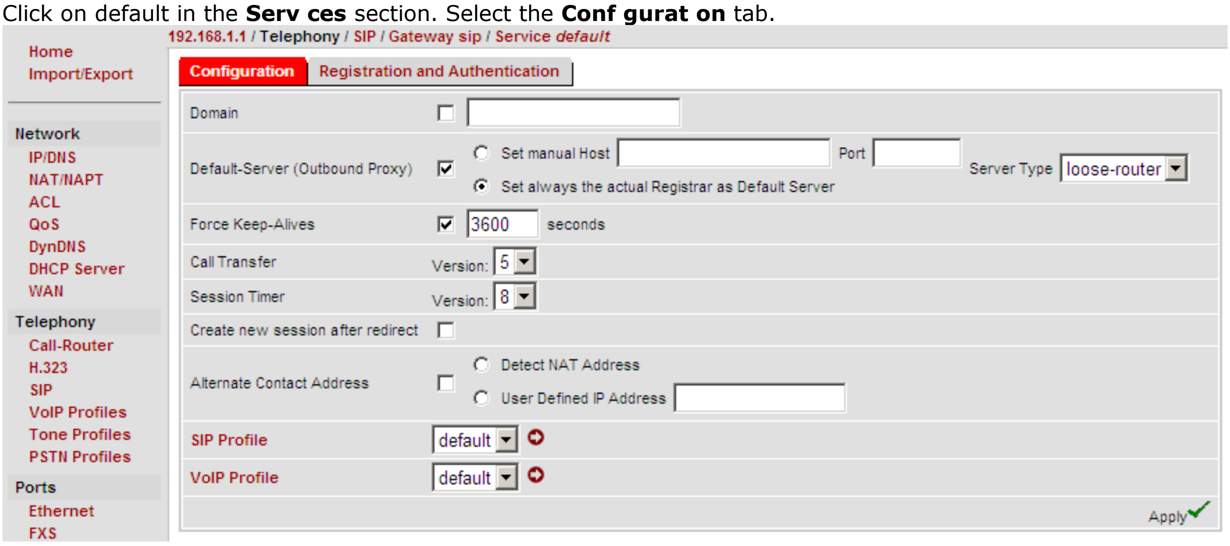
1. Browse to the IP address of the A10.
2. Enter the administrator name and password. The defaults are ***n mdbg*** and ***54321***.
3. Select **Tele hony** and then **SIP**.



4. Select the **Gateways** tab and click on **s** .

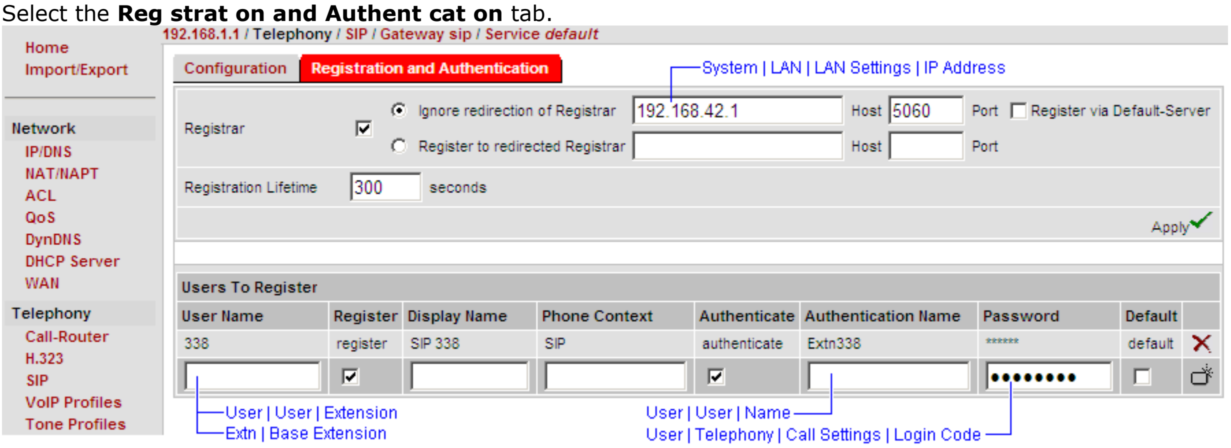


5. Click on default in the **Serv ces** section. Select the **Conf gurat on** tab.



- Ensure that the **Doma n** field is empty and the check box not selected.
- Enable the check box for **Default-Server (Outbound Proxy)** and select **Set always the actual Reg strar as Default Server**.
- Click **A   ly**.

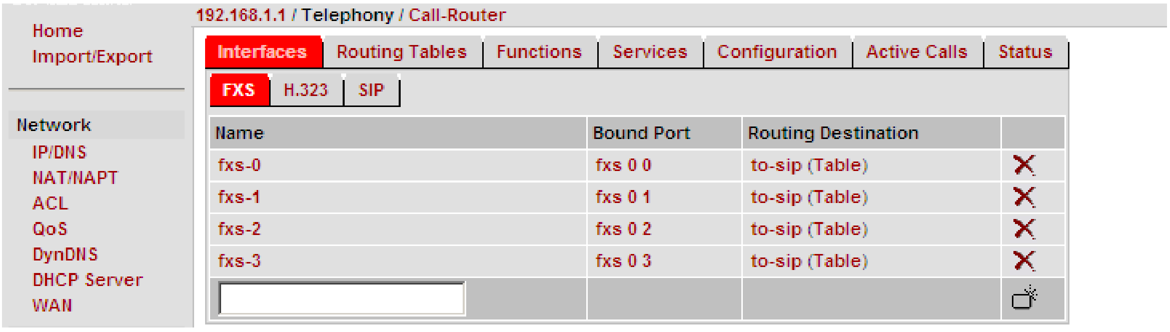
6. Select the **Reg strat on and Authent cat on** tab.



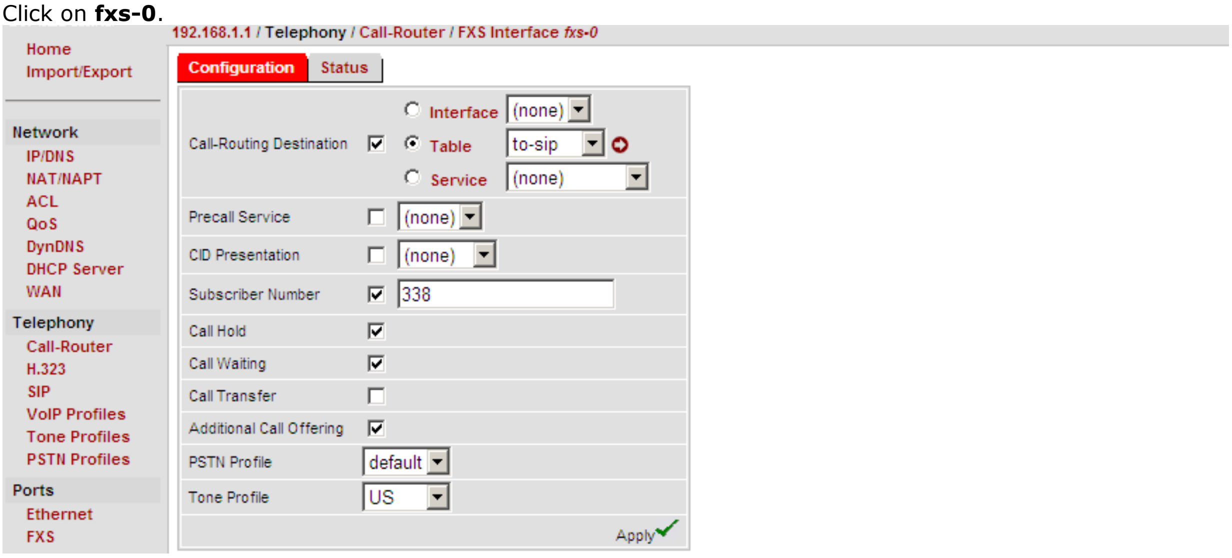
- Enable the Registrar checkbox. Select **Ignore red rect on of Reg strar** and enter the IP address and SIP port of the IP Office LAN on which the SIP registrar is enabled. Click **A ly**.

7. In the **Users To Reg ster** 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**.

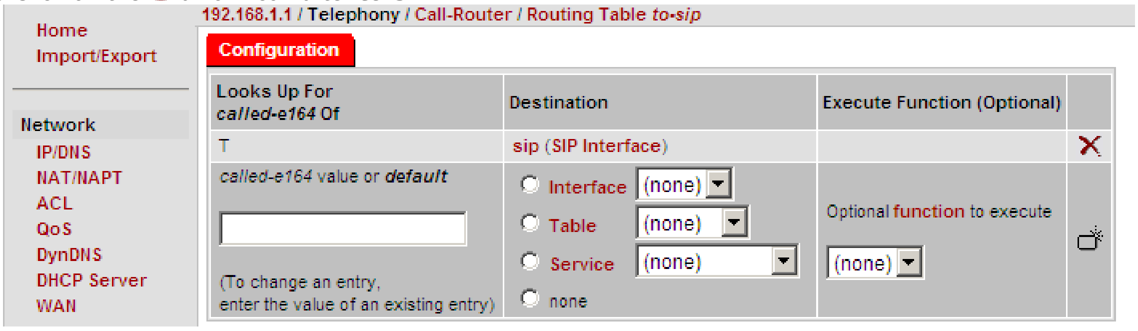


9. Click on **fxs-0**.



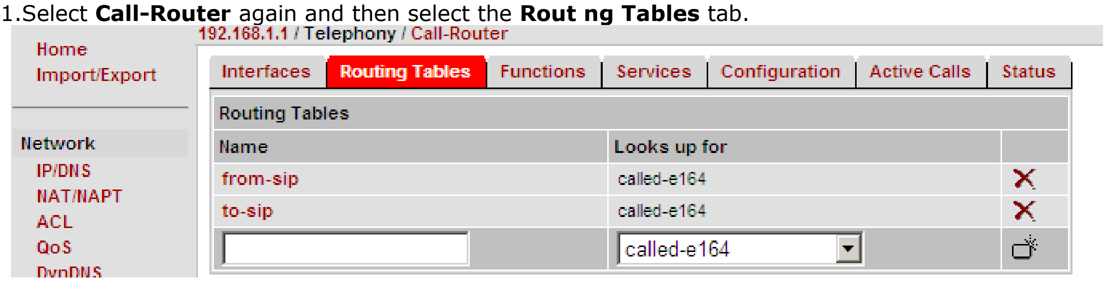
- Enable the **Call-Rout ng Dest nat on** checkbox. Select **Table** and in the adjacent drop down list select ***to-s*** .
- Enable the **Subscr ber Number** checkbox and enter the IP Office extension number for the SIP extension and user.
- Click **A ly**.

10.Click on the arrow icon after ***to-s*** .

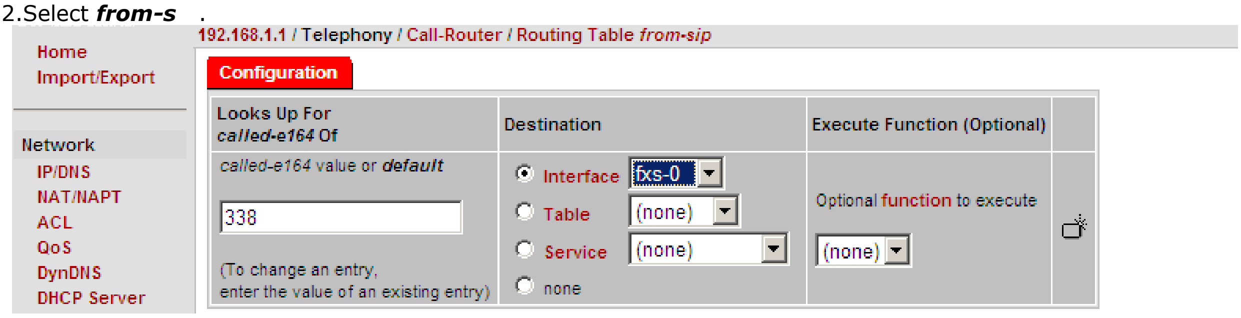


- Ensure that the table contains **T** with the destination **s** **(SIP Interface)**.

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

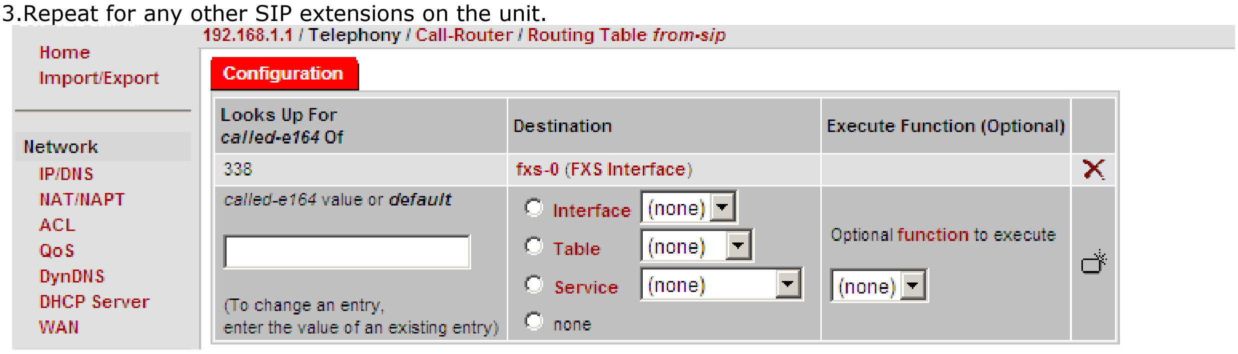


12.Select ***from-s*** .

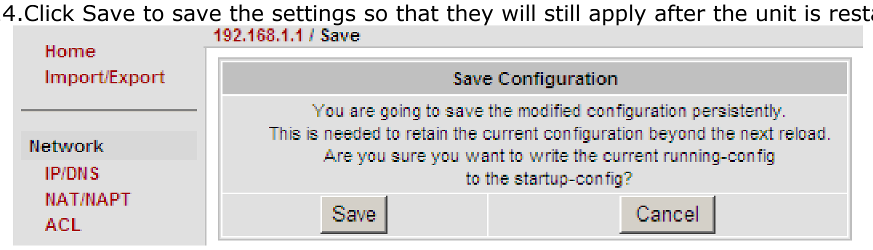


- Enter the IP Office SIP extension number.
- For the **Dest nat on** select **Interface** and select the matching fxs port for that extension number.
- Click .

13.Repeat for any other SIP extensions on the unit.



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



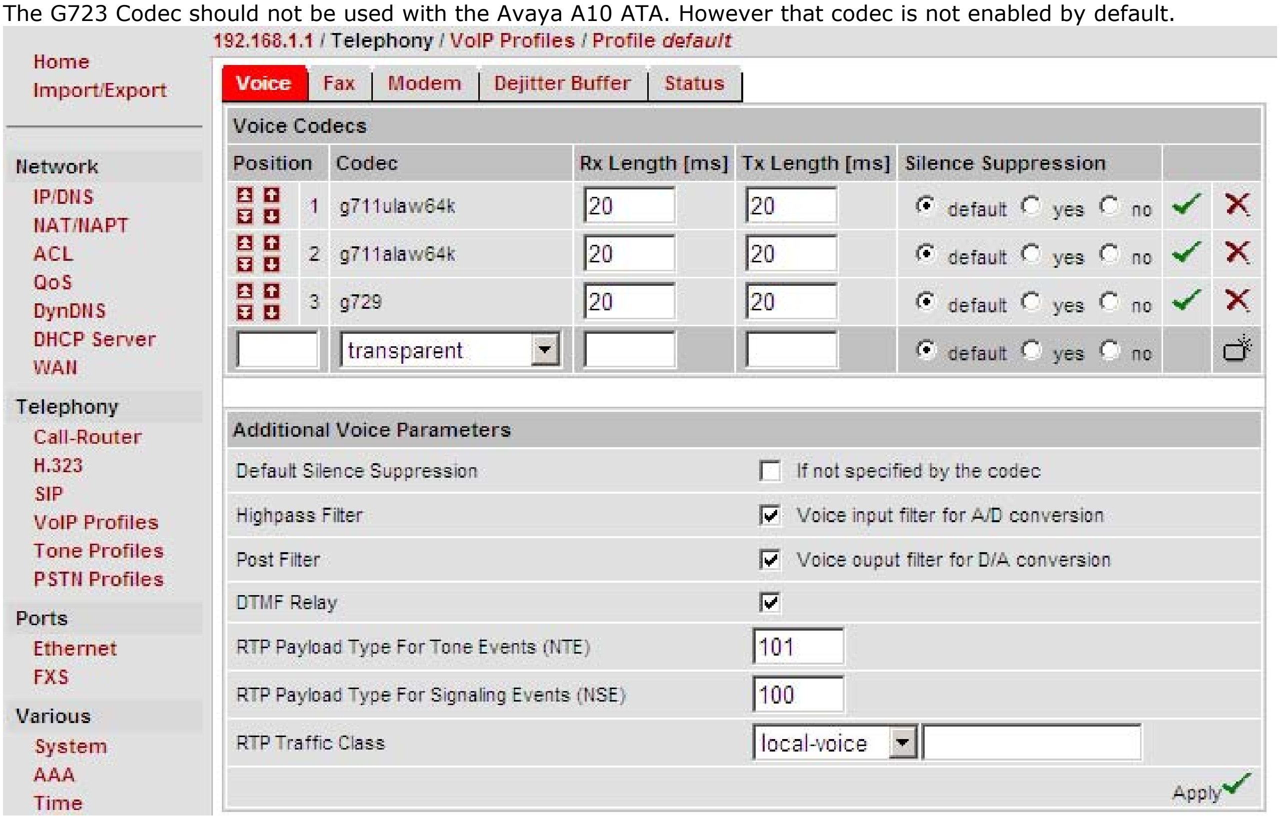
B.If installed using extension and user auto-creation, check the settings of the IP Office SIP extension and user created by the SIP devices registration.

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.

Notes

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

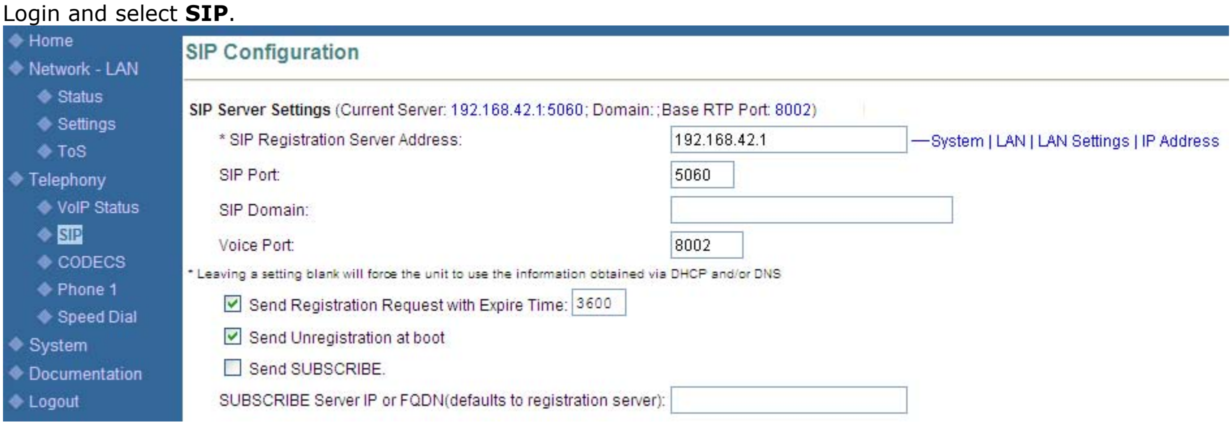


2.5 Patton Micro ATA

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**.

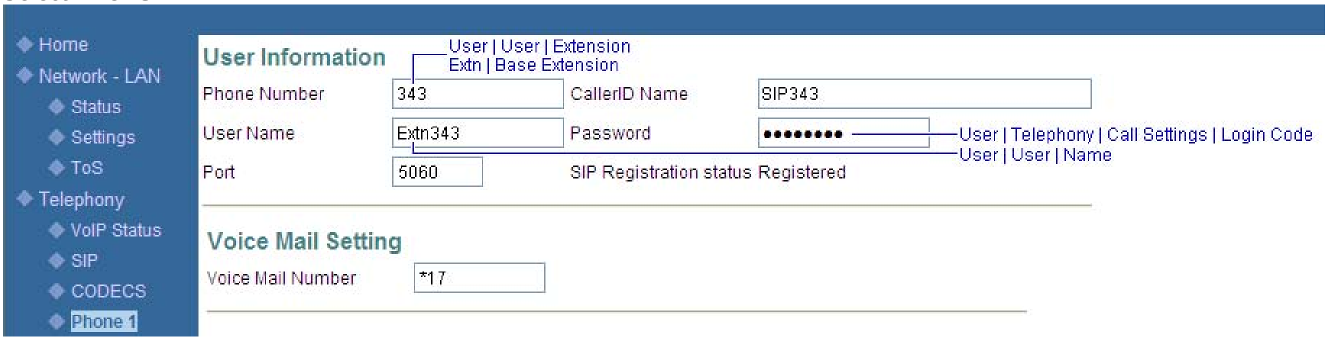


3. Enter the values to match the settings of the IP Office LAN on which the SIP Registrar is enabled. Click **Save**.

4. Select **CODECS**.

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

6. Select **Phone 1**.



7. Enter the values to match the IP Office SIP extension and user settings. Click **Save**.

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

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

D. If not installing any further SIP devices, **D sable 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 | Sett ngs | Connect on | S sett ngs | New SIP rof le**.

   2. Enter the following settings:

      - **Prof le name:** Give the profile a name that indicates its function.
      - **Serv ce rof le:** Select ***IE P***.
      - **Default access o nt:** Enter your access point.
      - **Publ c user name:** Enter an address of the form ***<IP Off ce extens on n mber>@<IP Off ce SIP Enabled LAN IP address>***, for example ***338@192.168.42.1***.
      - **Use com ress on:** Select ***no***.
      - **Reg strat on:** Select ***always on***.
      - **Use secur ty:** Select ***no***.
      - **Proxy server:** Leave blank.
      - **Reg strar server:**
        - **Reg strar server adress:** Enter the IP Office SIP Enabled LAN IP address.
        - **Realm:** Enter an address of the form ***<IP Off ce ser name>@<IP Off ce 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.
        - **Trans ort ty e:** Select auto.
        - **Port:** Match the port set on the IP Office LAN **SIP Reg strar** tab, by default this is ***5060***.

   3. Select **Menu | Tools | Sett ngs | Connect on | Internet tele hone | New rof le**.

      - Select the SIP profile just created above.

   4. Select **Menu | Commun cat on | Internet tel. | O t ons | Sett ngs**.

      - Change the **Default call ty e** 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, **D sable Auto-Create Extn/User** if it is enabled.

2.7 Innovaphone IP22, IP24, IP28

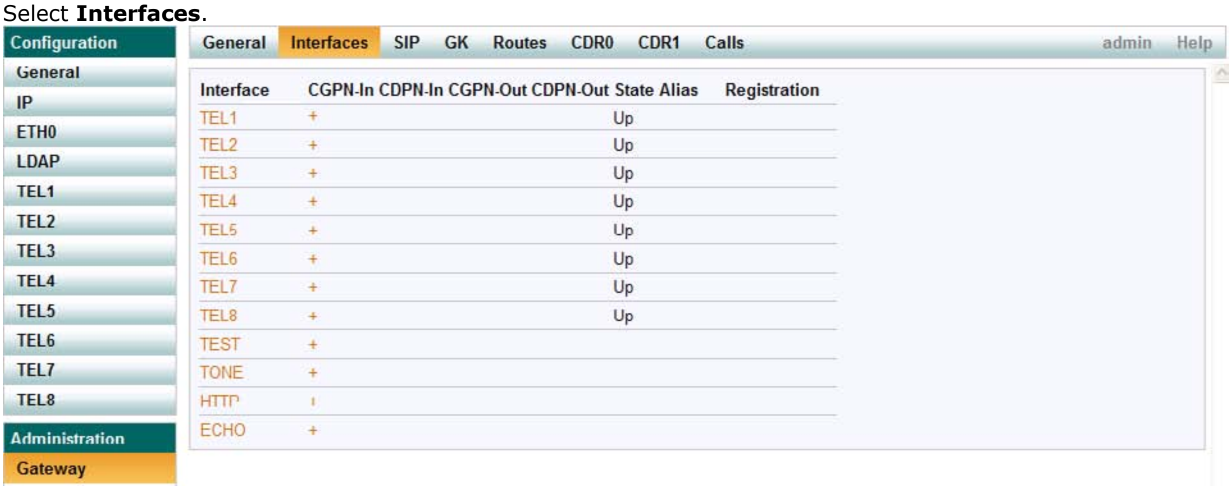
A. Either enable **Auto-Create Extn/User** or otherwise manually add SIP extensions and users to the IP Office configuration.

1. Browse to the IP address of the unit.

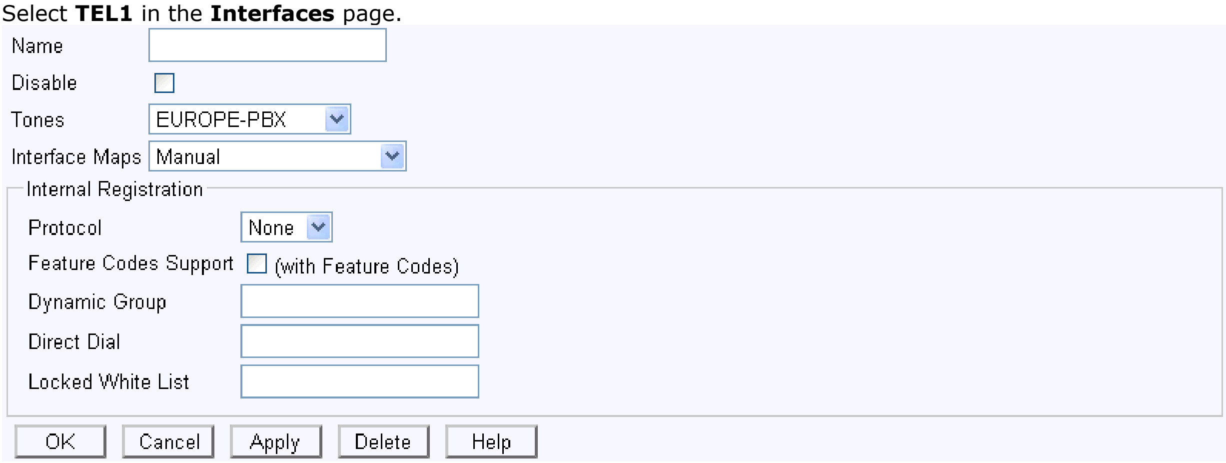
2. In the left hand column select **GATEWA** .

3. ou will be prompted to login. The default user name is ***adm n***. The default password is ***22***, ***24*** or ***28*** depending on the unit type.

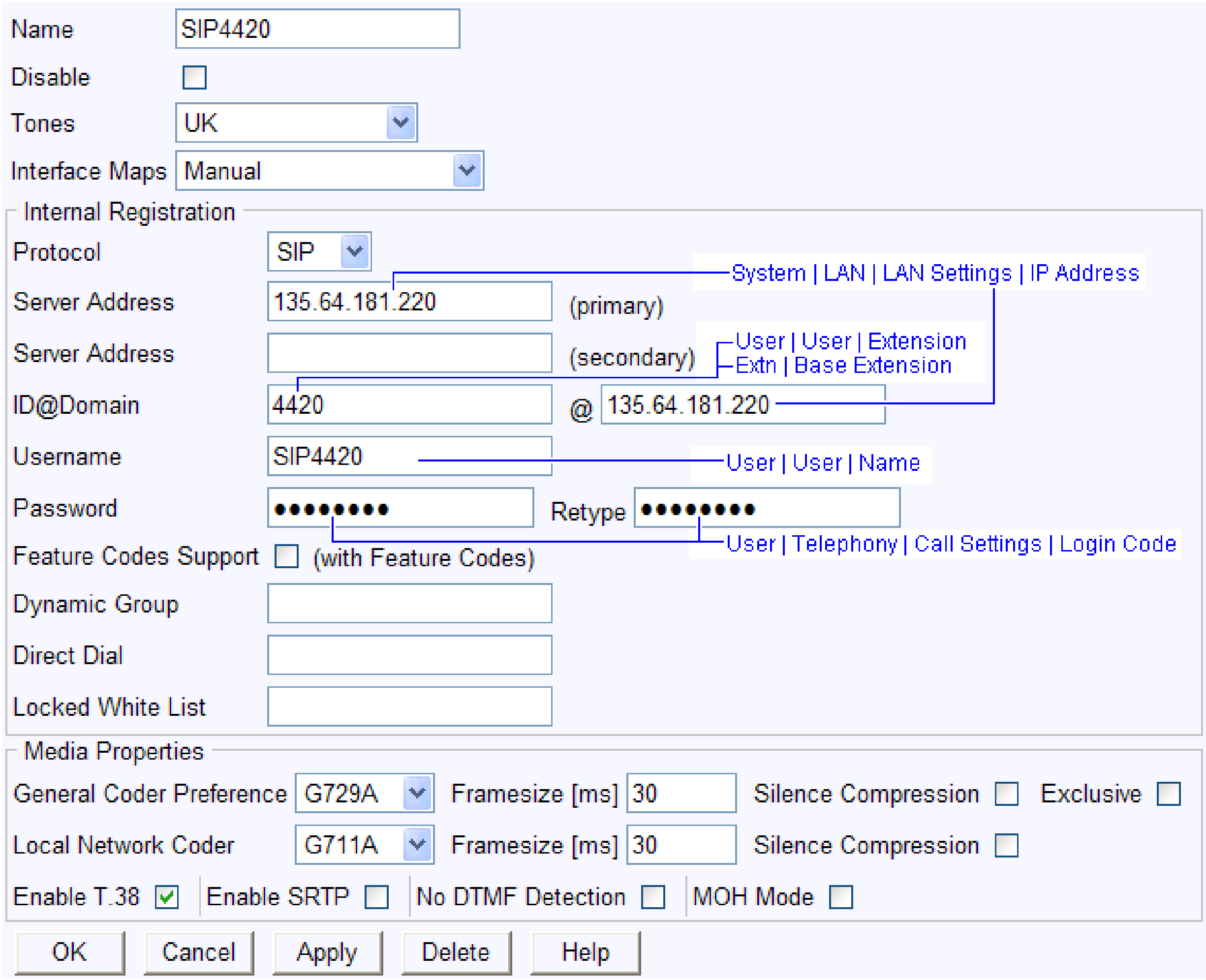
4. Select **Interfaces**.



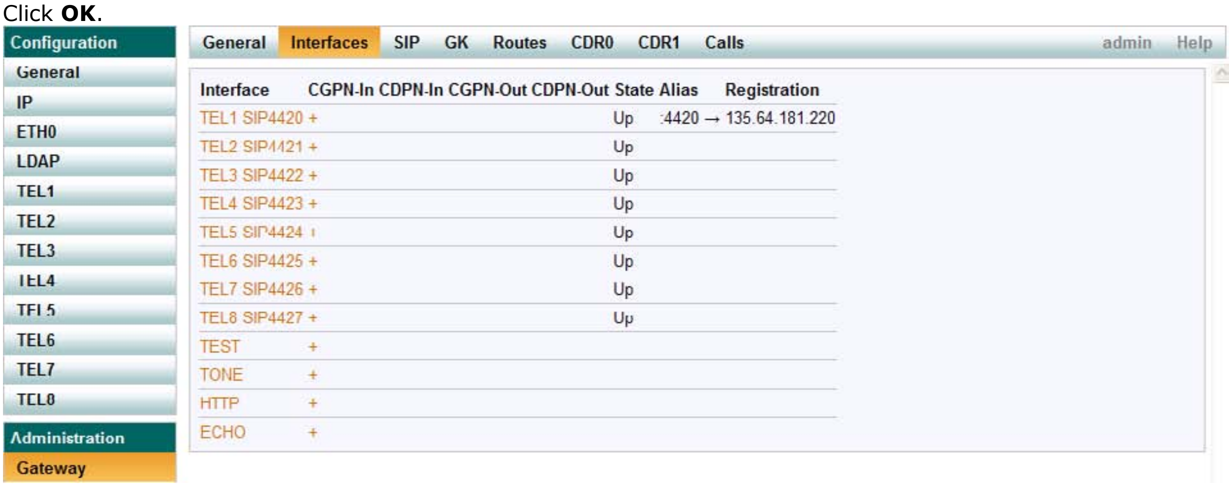
5. Select **TEL1** in the **Interfaces** page.



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



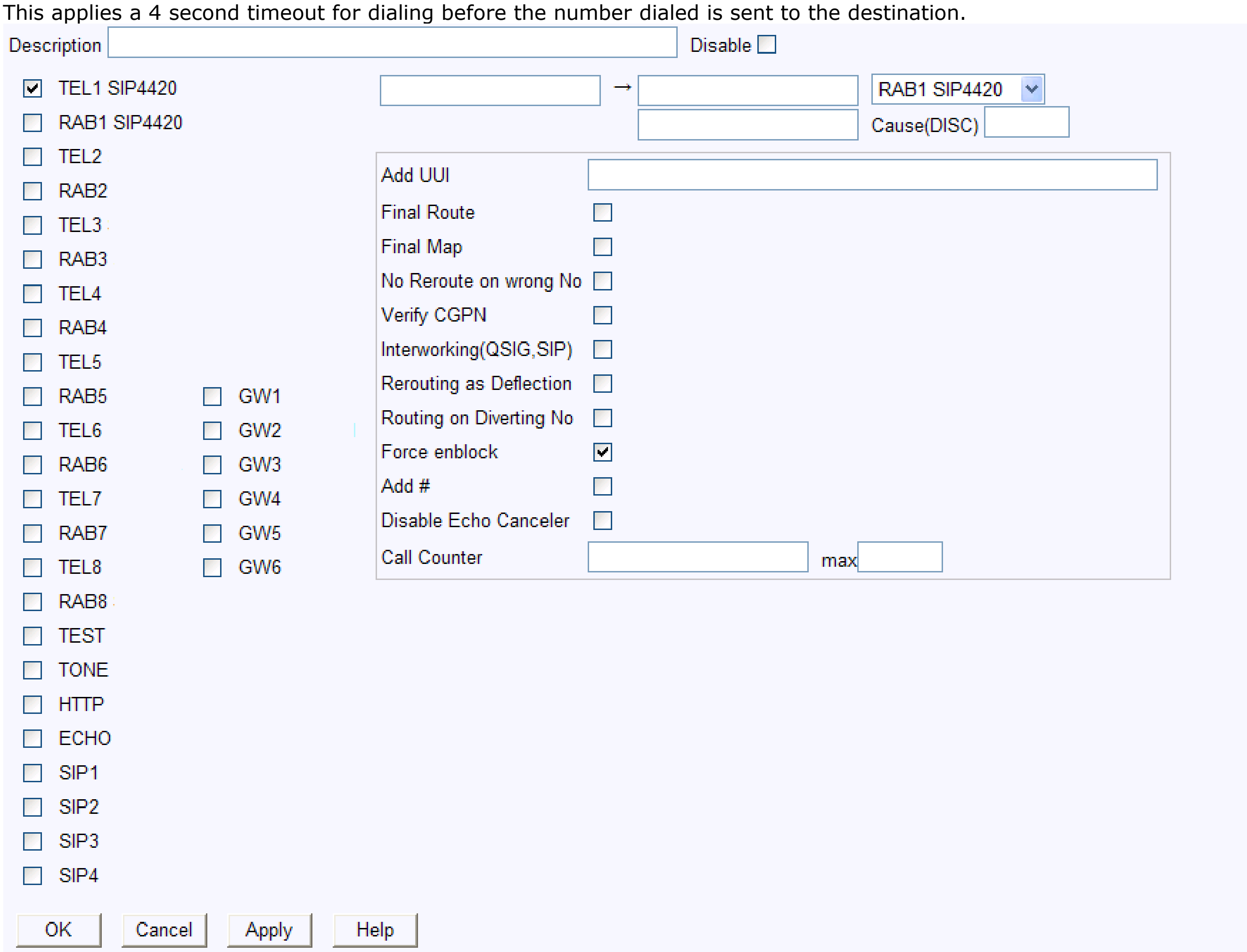
7. Click **OK**.



8. Select **Routes**.

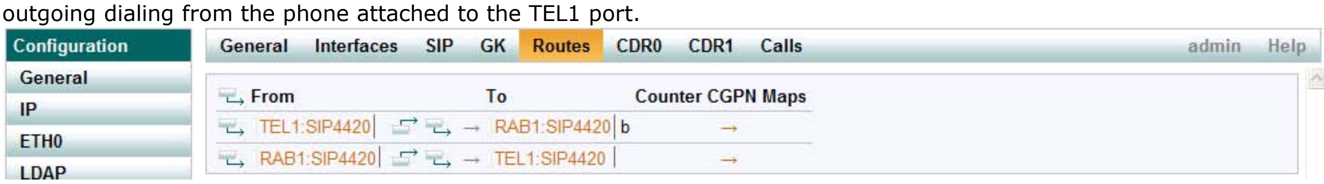
9. Two new routes are needed, one for dialing from the phone attached to the TEL port and one for incoming calls to the SIP account registered with the TEL port.

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.



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.



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, **D sable Auto-Create Extn/User** if it is enabled.

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