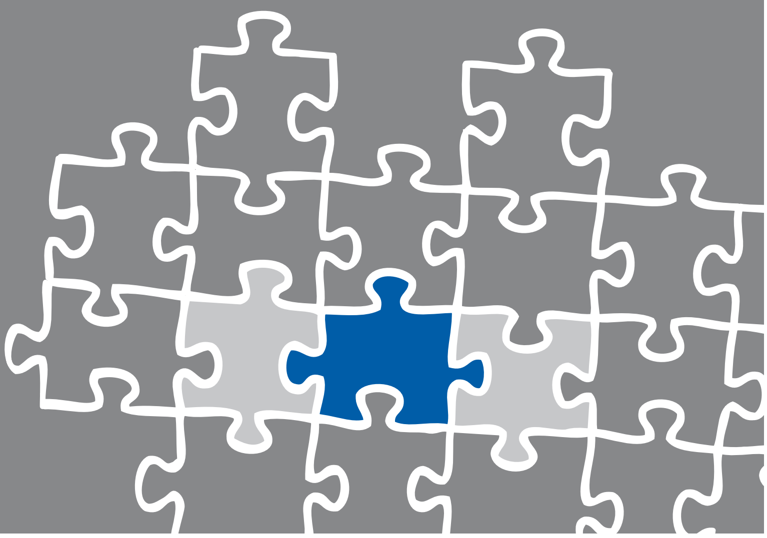
TechNote

Avaya IP Office 8.0 - SIP

September 10, 2012

XCAPI voiceoverIP



Introduction

This document is intended to support you with the integration of the XCAPI version 3.3.259 into an existing environment of the Avaya IP Office. Though being based on the 8.0 of Avaya IP Office it should be applicable to other versions, given a few adjustments. In the following sections we show the essential steps of configuration to allow for optimal cooperation of both the XCAPI and the Avaya IP Office. At this point we suppose that the PBX environment, the hardware the XCAPI is running on and both XCAPI and the CAPI application are already installed properly.

For detailed information about Avaya IP Office configuration procedures, please refer to the respective manuals. Additional XCAPI information and documents, for e.g. installation procedures, License on demand, Fax Transmission, Virtual Hardware ID, VMware Virtual Machines, can be found on the XCAPI Website within our community download section in the XCAPI TechNotes area.

XCAPI Configuration

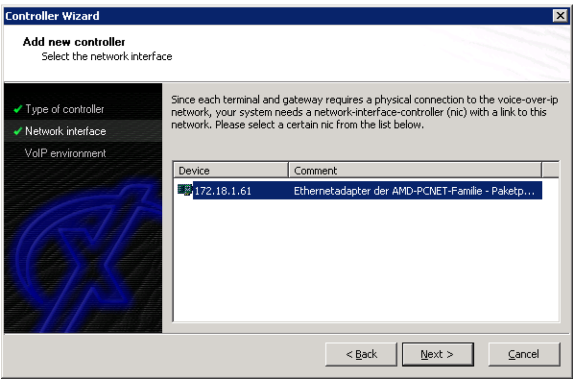
Please start up the XCAPI configuration to create a new controller assigned to the Avaya IP Office.

If you've just installed the XCAPI and start the configuration tool for the first time, the XCAPI controller wizard will pop up automatically. This will also happen if there's no controller configured at all. To start up the XCAPI controller wizard on your own, just click the hyperlink labelled **Click here to add a controller** on the main page of the XCAPI configuration tool.

On the first page of the controller wizard please select the **Add Voice-over-IP controller (VoIP)** option and continue by clicking on the **Next** button.

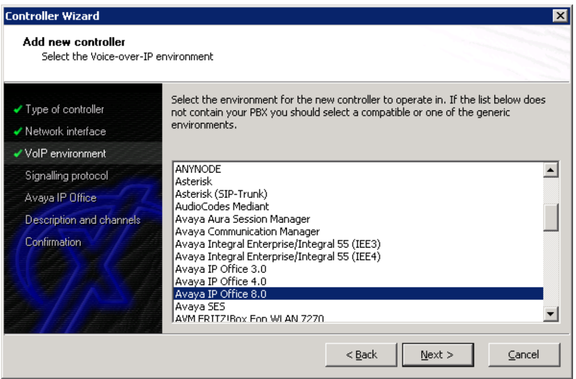
2.1 Network Interface

On this page of the XCAPI controller wizard you can select the network adapter you want to bind to the XCAPI controller.



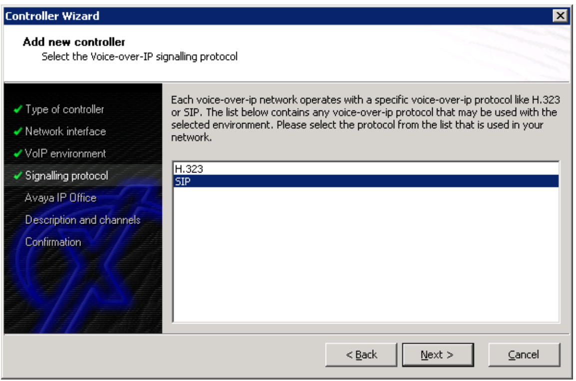
2.2 Voice over IP Environment

The next dialog of the configuration tool shows a list of some common Voice-over-IP environments. Selecting one of those will configure the XCAPI with a selection of near-optimal presets for the kind of environment you have, sparing you quite a lot of manual configuration.



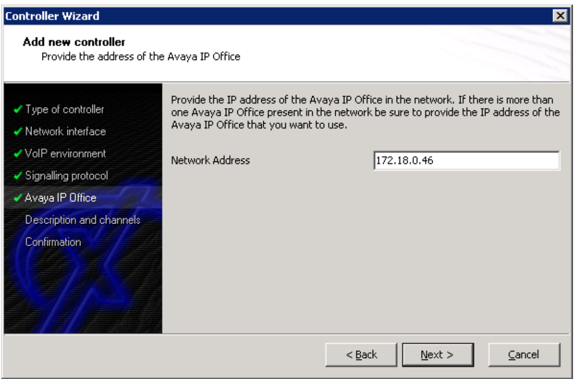
2.3 Signalling Protocol

The next dialog shows a list of signalling protocols which are supported for the given Voice-over-IP environment. For this example we use the SIP protocol, so please select that from the list.



2.4 Gateway Address

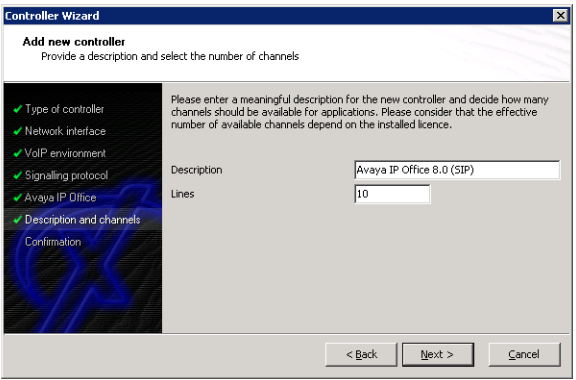
Next, please provide the host name or the IP address (In this example **172.18.0.46**) of the SIP listening Avaya IP Office Ethernet interface.



2.5 Description and Channels

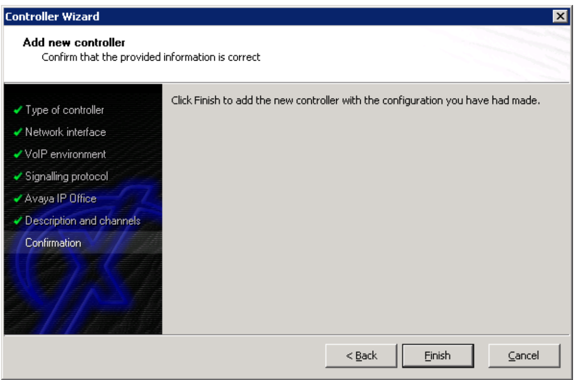
The next-to-final dialog of the controller wizard allows you to configure a meaningful description for the controller you're going to create.

This dialog, however, also allows configuring the number of channels that the new controller will be able to provide. Please enter how many simultaneous connections the XCAPI should handle when communicating with the Avaya IP Office.



2.6 Confirmation

The final dialog of the controller wizard performs some checks on the configuration parameters you've made. If everything is correct, please use the **Finish** button in order to create the new controller.



Finally you can save the new created controller which appears now on main view of the XCAPI configuration.

You always need to restart the bound CAPI application, in meaning of its services, for the changes to take effect.

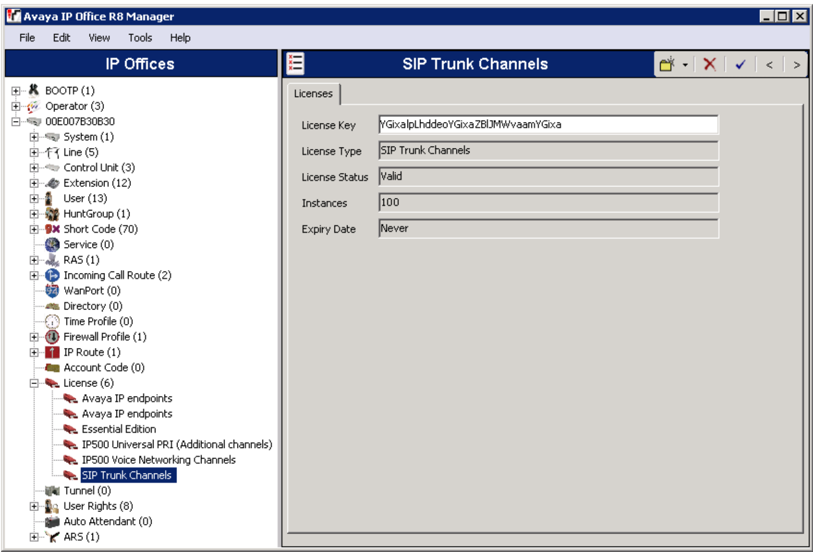
Configuring the Avaya IP Office

In order to establish a connection between the XCAPI and the Avaya IP Office you need to setup the XCAPI as SIP trunk with all its appropriate configurations. The next chapters show a basic configuration which can't be assigned one-to-one to the environment.

The according configuration dialogs have to be adapted to the PBX environment and hardware and the according CAPI application. Especially the dialing plan and its related trunk group settings (access codes, DID and DNIS mapping, trunk digit manipulation) must reflect the local circumstances.

3.1 License

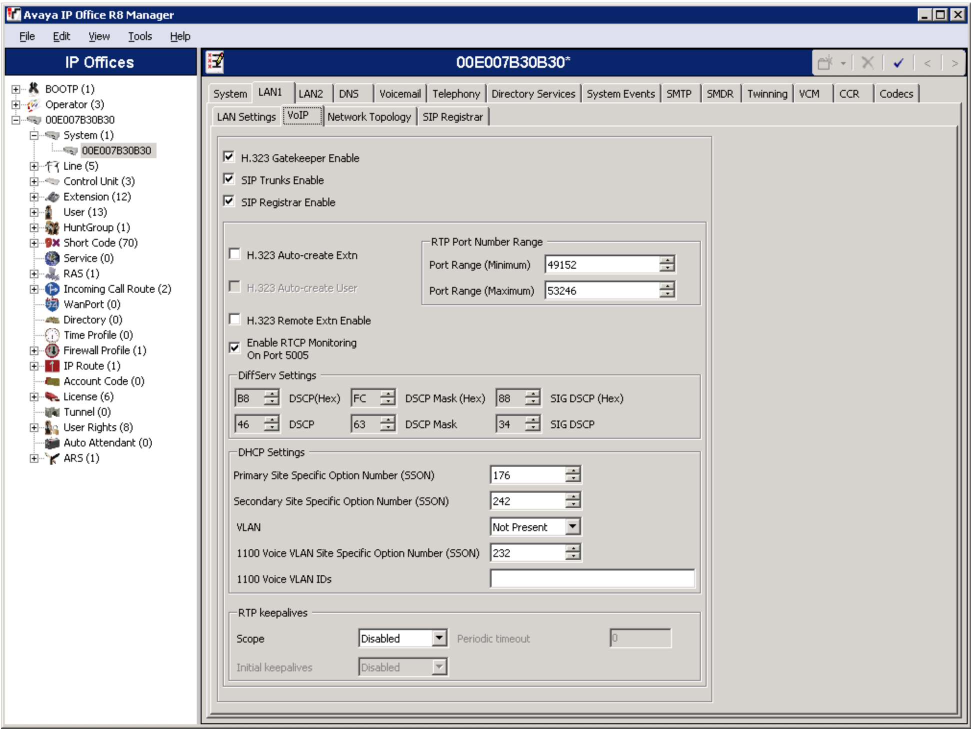
First, please ensure that the **Avaya IP Office SIP Trunk Channel** license key is available for allowing SIP trunking.



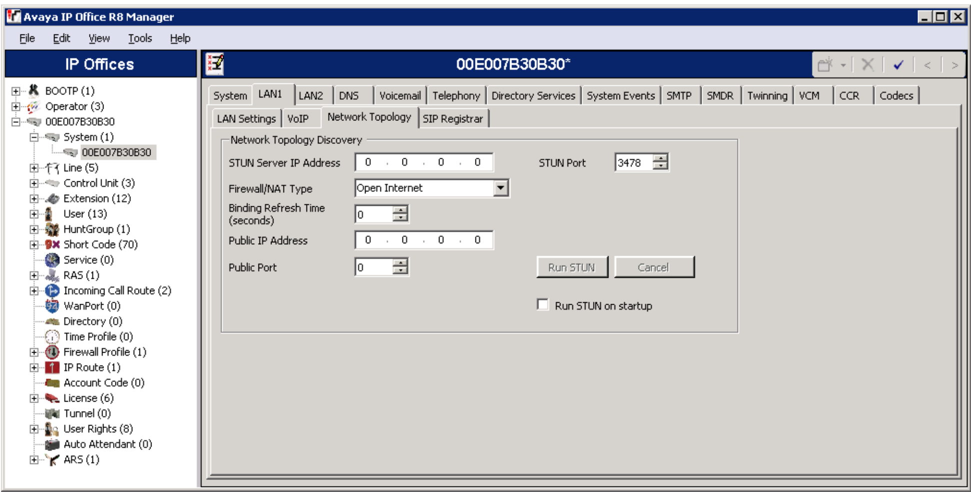
Additional voice networking channel licenses may be necessary, depending on the firmware and hardware.

3.2 System

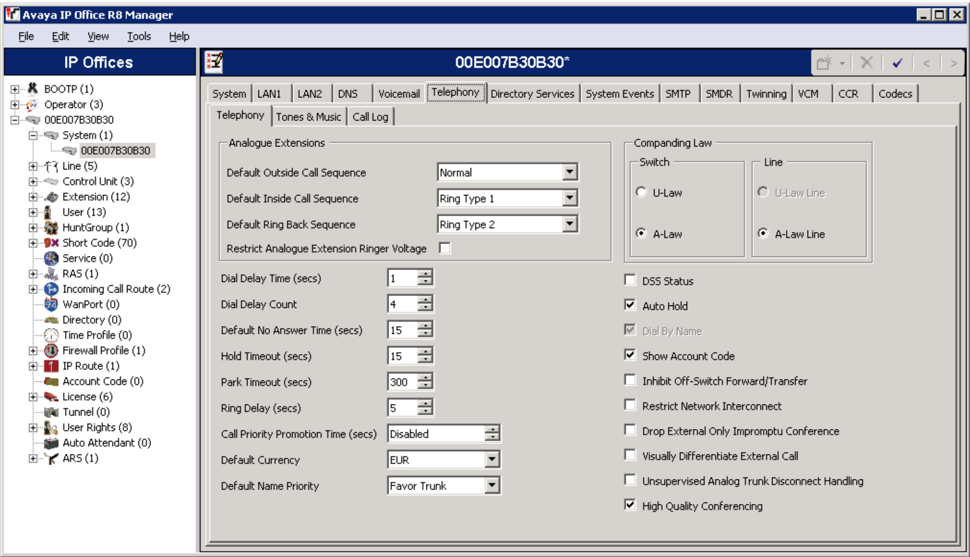
Next, the **VoIP** tab within the **System** configuration is going to be reviewed. Please ensure that the SIP trunk support is enabled, see checkbox below **SIP Trunks Enable**.



This example environment doesn't use any STUN or Firewall. The firewall type is related to the systems **Open Internet** profile.

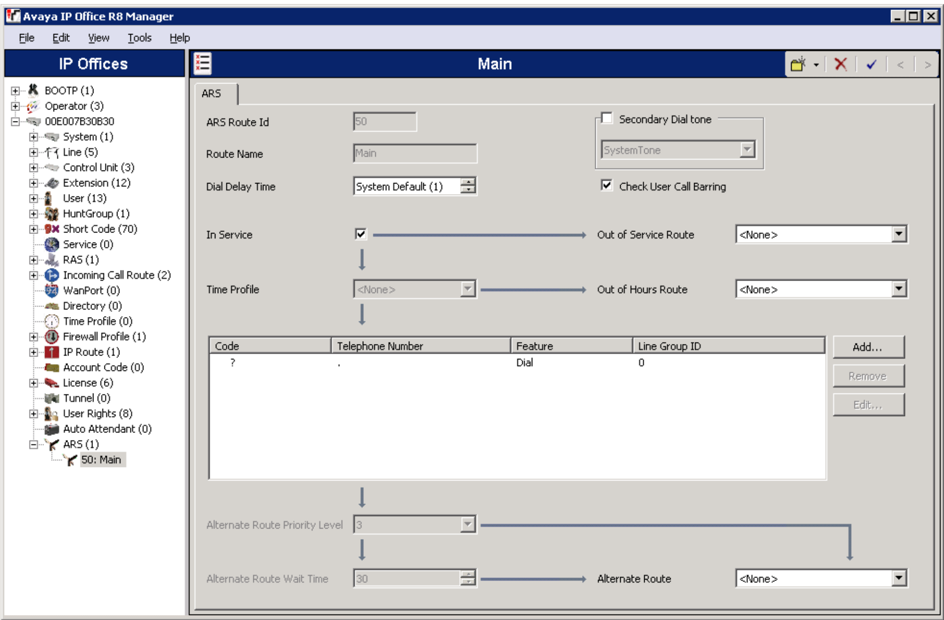


The **Telephony** settings are used with their default values. You might have to disable **Inhibit Off-Switch Forward/Transfer** for allowing call forwarding or call transfer scenarios through the PSTN.



3.3 ARS

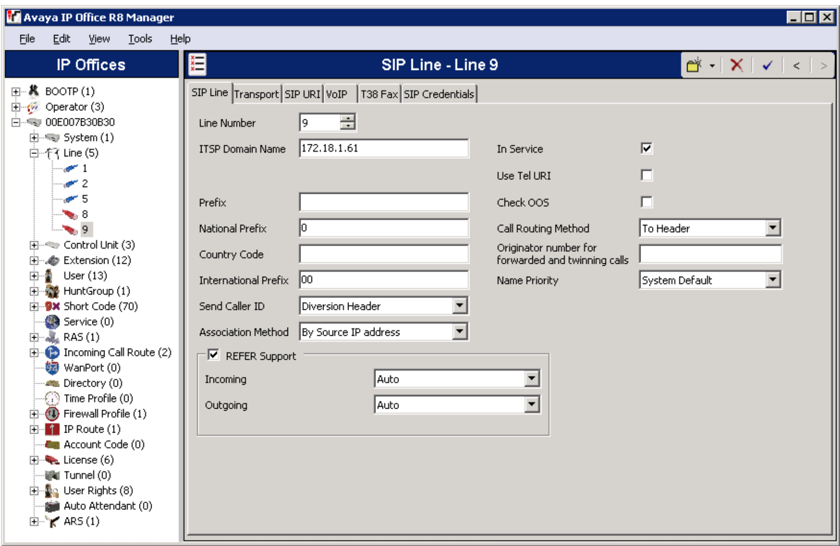
This environments **ARS** configuration is used with its default main entry.



3.4 SIP Line

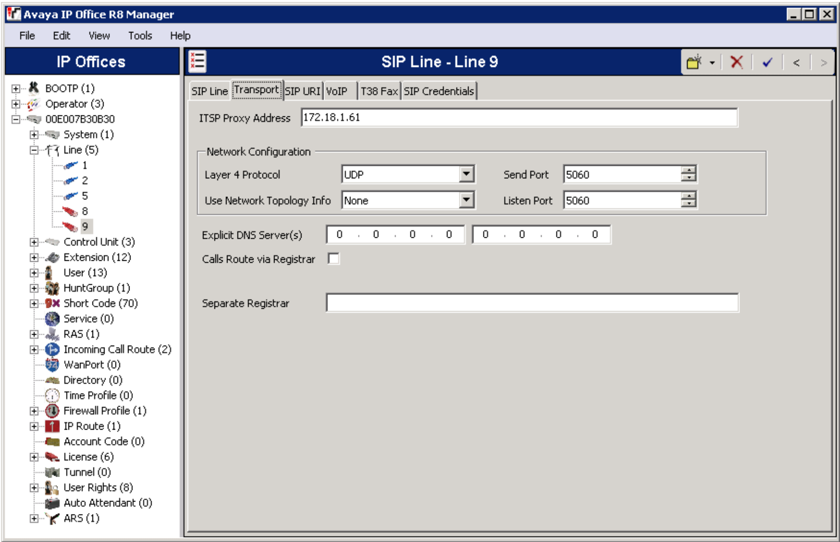
The XCAPI's related **SIP Line** is defined as **Line Number 9**. Its configuration in detail:

- The **ITSP Domain Name** is set to the IP address **172.18.1.61**, which determines the Ethernet interface address of the XCAPI and its application server.
- For appropriate use of the redirection number the **Send Caller ID** parameter must be set to **Diversion Header**.
- **REFER Support** has to be enabled for call transfer via SIP refer.
- Ensure that the **SIP Line** is **In Service**.
- For appropriate interworking **Use TEL URI** must be disabled.
- If required the **Check OOS** box can be enabled. The IP Office will use the SIP OPTIONS method, which is also supported by the XCAPI, to periodically check the SIP Line.

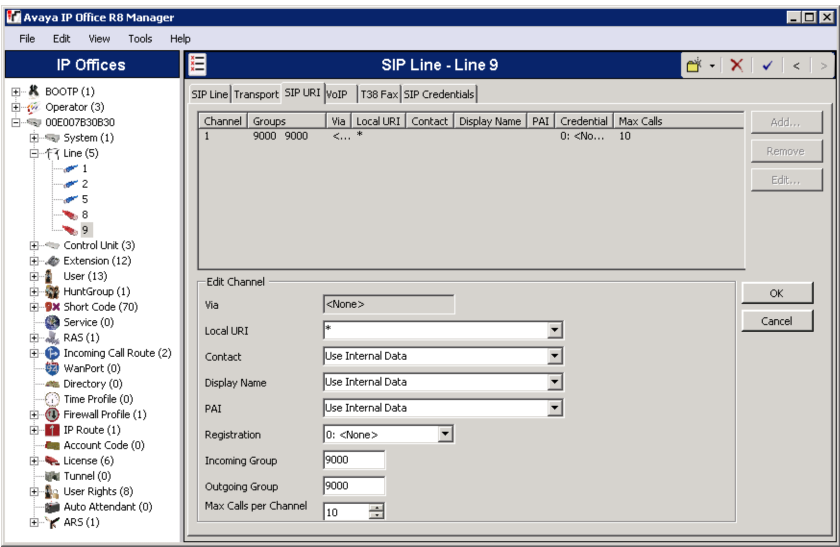


Within the **Transport** tab the IP address **172.18.1.61** is again related to the XCAPI's Ethernet interface. The **Network Configuration** parameters are used with their default values. Nevertheless, please ensure that those settings conform to the XCAPI controller configuration.

In detail **Layer 4 Protocol** is set to **UDP**, **Use Network Topology Info** is set to **None** and **Send Port** and **Listen Port** is set to **5060**.

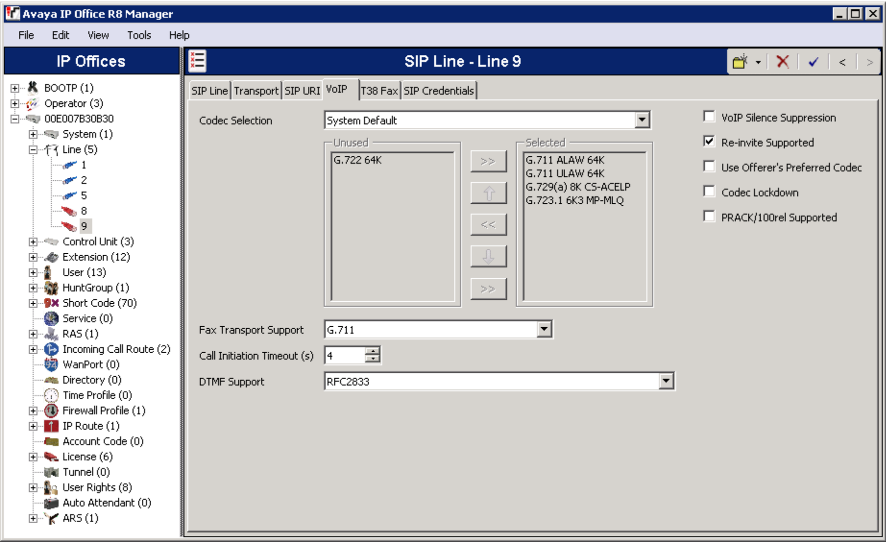


The **SIP URI** tab is configured as follows. The **Local URI** is used with wildcard * which determines that the SIP trunk will accept any incoming SIP call. **Contact**, **Display Name** and **PAI** credentials are here used with their defaults (**Use Internal Data**). The **Incoming** and **Outgoing Group** ID is set to **9000** which is the related short codes ID for the call routing.



The **VoIP** tabs **Codec Selection** is assigned to **Systems Default** profile. For appropriate interworking the **Re-Invite Supported** flag is set. All others parameters are unchecked.

Fax Transport and **DTMF Support** are set to **G.711** and **RFC2833**. This is also used by default within the XCAPI controller configuration. Additional information about facsimile can be reviewed in the according chapters starting on page 16.

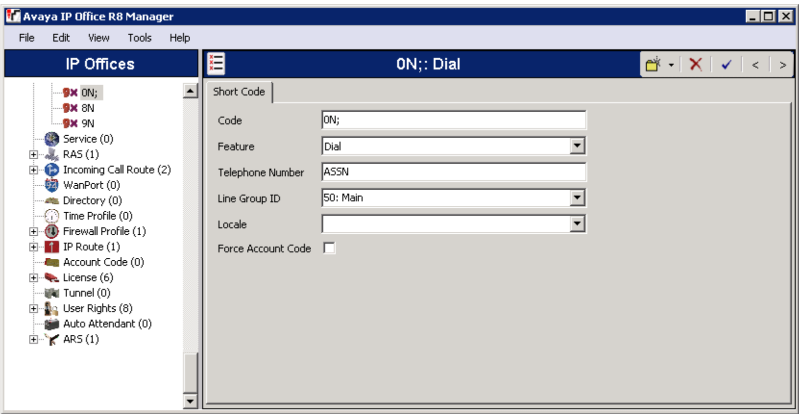


3.5 Short Codes

In this chapter we are going to review the most important short codes for appropriate PSTN and SIP line routing.

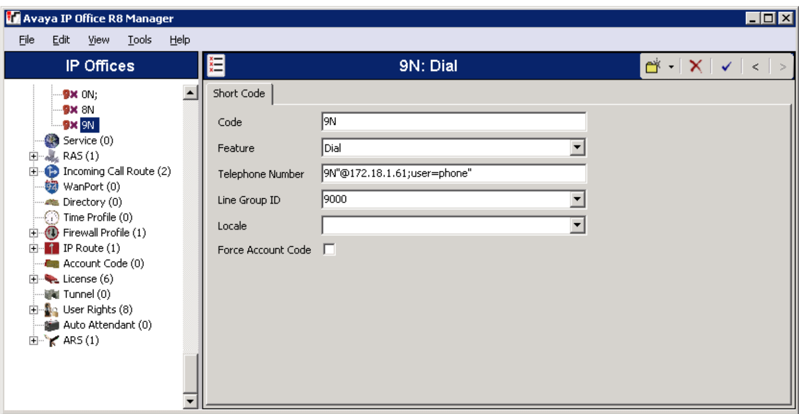
For matching the PSTN line (here related to **Line Group ID 50: Main**) the short code **0N;** is used and set to the **Dial Feature**.

The **Telephone Number** is set to **ASSN**. In detail code **A** set an outgoing cli, **SS** pass through the calling number and **N** matches any prefix.



The SIP line using **Line Group ID 9000** will match any prefix starting with 9 (**Code 9N**) and uses also the **Dial Feature**.

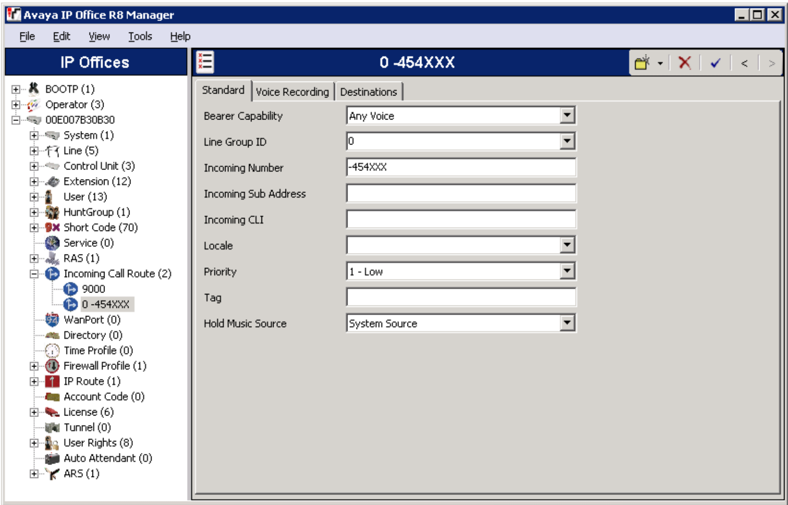
Telephone Number is set to **9N"@172.18.1.61;user=phone"** for transmitting an appropriate SIP-to header (for e.g. **To: <sip:900@172.18.1.61;user=phone>**).



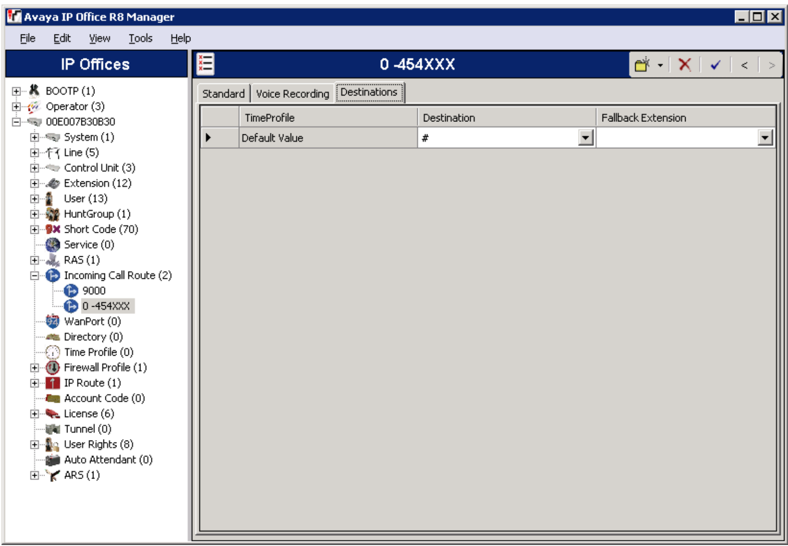
3.6 Incoming Call Route

This example uses two incoming call routes. The first call route is related to the main (PSTN) line.

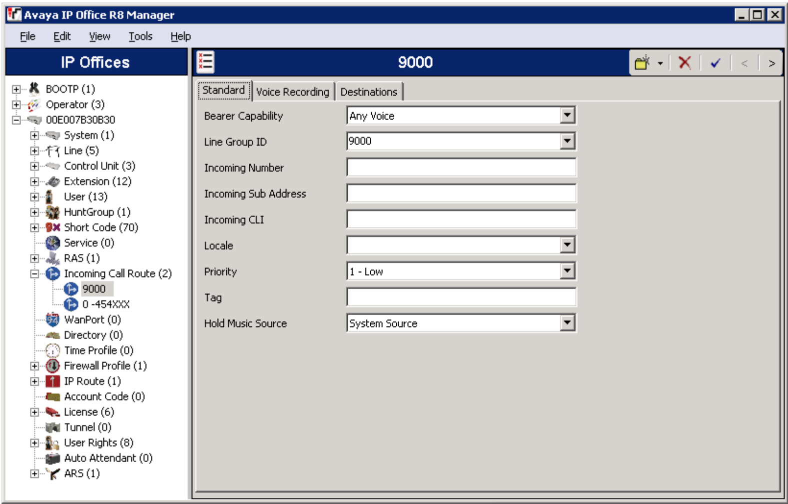
The **Incoming Number** is set in accordance with this environments used PRI interface **-454XXXX** and its prefix relations. Please replace our example prefix **454XXXX** with the real PSTN provided prefix.



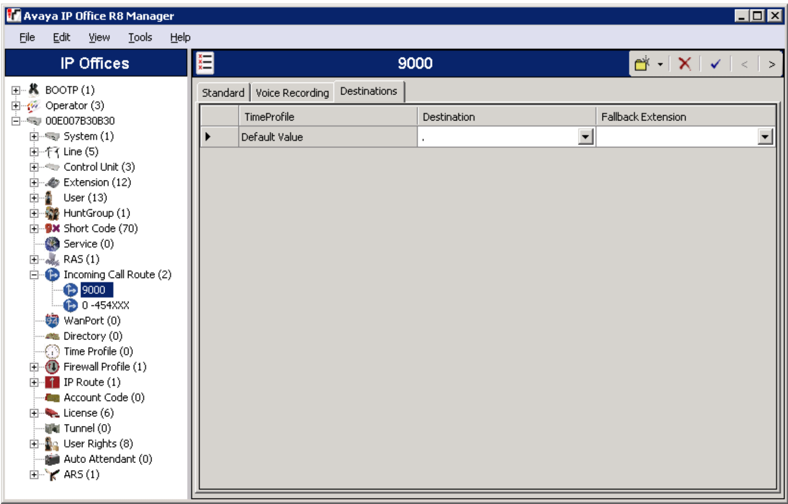
Within the **Destinations** tab the ♯ symbol is used for matching all **X** wildcards used in the incoming number field.



The second call route is related to the XCAPI's SIP line using ID 9000. **Incoming Number** and **Incoming CLI** aren't set at all here.



For this call route the **Destination** value is set to dot (**.**) for matching any value of the incoming number field.



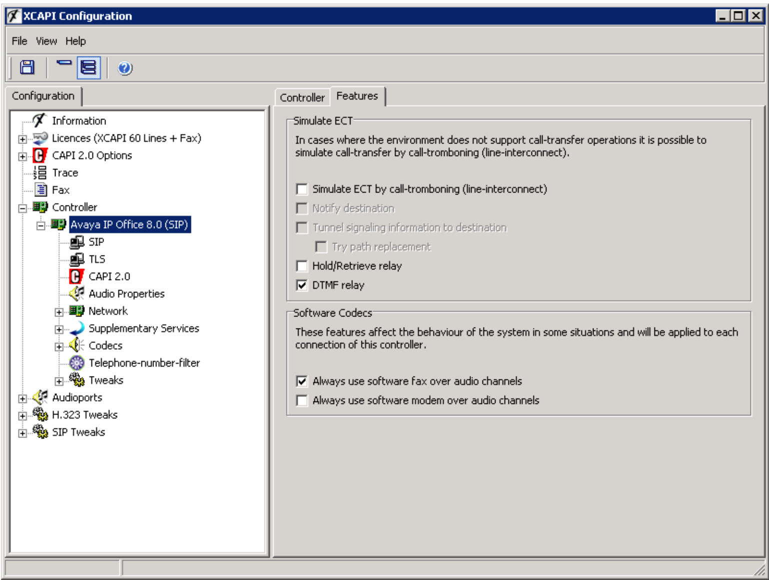
Fax Services

This chapter shows the necessary settings when using facsimile services via the SoftFax (G.711) method which is also set by default within the XCAPI controller

4.1 SoftFax

In the SoftFax mode, the XCAPI simulates an analog Fax device by transmitting modulated Facsimile signals modem-like via audio-channels. For this the **Always use software fax over audio channels option** within the **Features** configuration dialog must be enabled.

Ensure that the **Fax Transport Support** within the SIP line **VoIP** tab is set to **G.711** as shown in the according screenshot in the chapter **SIP Line** starting on page 10.



If using facsimile via G.711 you might have to set the according short code to **Dial 3K1** for appropriate interworking. This dial feature sets the ISDN bearer capabilities to 3.1KHz audio call and improves the compatibility with specific destinations which are only capable detecting facsimile with such bearer capability.

Supplementary Services & Features

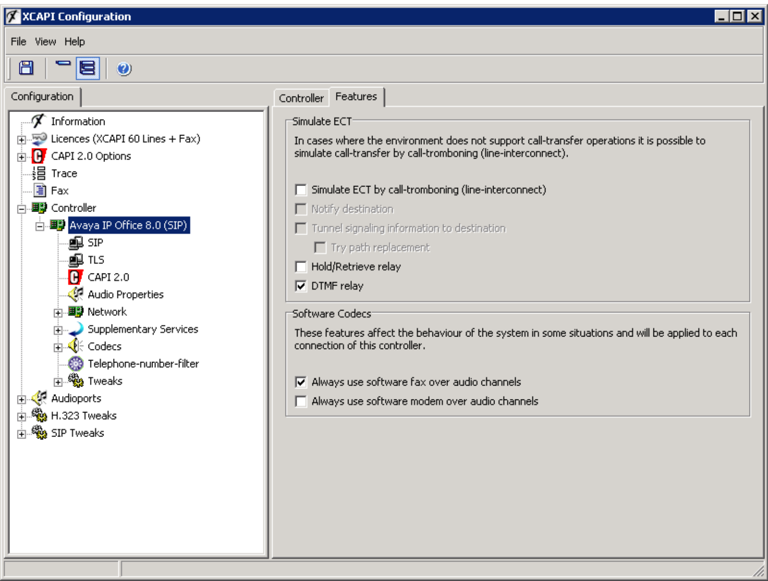
Please review the following chapters for some information on optimal supplementary services and features configuration.

5.1 Call Transfer

Please ensure that the **Simulated ECT by call-tromboning (line-interconnect)** parameter of the XCAPI controller `Features` dialog is disabled for supporting call transfer via the SIP Refer method.

As already mentioned in the **SIP Line** chapter starting on page 10 the **REFER Support** must be allowed for the incoming and/or outgoing direction. Additionally you might have to disable the **Inhibit Off-Switch Forward/Transfer** parameter as mentioned in the **System** chapter starting on page 8.

Depending on the call transfer / forward scenario you might have to set the according parameters for **Restrict Network Interconnect** or **Analog Trunk to Trunk Connection**.

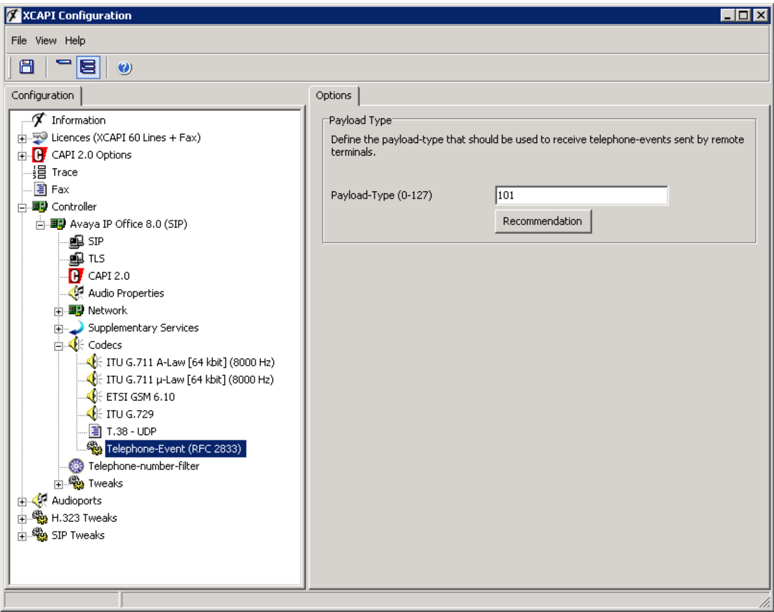


Redirecting Number

The SIP lines **Send Caller ID** must be set to **Diversion Header** for enabling redirection numbers via SIP diversion header. In addition you might have to set some specific short code characters. More details are mentioned in the **SIP Line** and **Short Codes** chapter starting on page 10 and page 13.

DTMF

The payload type for signalling telephone events via **RFC2833** is set by default to value **101**. For DTMF interoperability please ensure that the DTMF Support is set to **RFC2833** as shown in the chapter **SIP Line** starting on page 10.



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