TechNote

Avaya IP Office 8.0 - SIP

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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 Voiceover-IP environment. For this example we use the SIP protocol, so please select that from the list.

Type of controller	Each voice-over-ip network operates with a specific voice-over-ip protocol like H.323 or STP. The list below contains any voice-over-in protocol that may be used with the
 Network interface 	selected environment. Please select the protocol from the list that is used in your petwork
✓ VoIP environment	
 Signalling protocol 	H.323
Avaya IP Office	
Description and channels	
Confirmation	

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.

Type of controller	Provide the IP address of the one Avava IP Office present	e Avaya IP Office in the network. If there is more that in the network be sure to provide the IP address of
Network interface	Avaya IP Office that you wa	ant to use.
VolP environment	Network Address	172.18.0.46
Signalling protocol		
Avaya IP Office		
Description and channels		
Confirmation		



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.

 Type of controller 	Please enter a meaningfu channels should be availa	I description for the new controller and deci ble for applications. Please consider that th	ide how many e effective
Network interface	number of available chan	nels depend on the installed licence.	
✓ VolP environment			
Signalling protocol	Description	Avaya IP Office 8.0 (SIP)	
✓ Avaya IP Office	Lines	10	
Description and channels			
Confirmation			

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.

Controller Wizard			×
Add new controller Confirm that the provide	d information is correct		
✓ Type of controller	Click Finish to add the new controlle	r with the configuration you have had	made.
 Network Interrace VolP environment 			
 Signalling protocol 			
✓ Avaya IP Office			
 Description and channels 			
Confirmation			
		< Back Einish	<u>C</u> ancel





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**.

f Avaya IP Office R8 Manager	
<u>Eile E</u> dit <u>V</u> iew <u>T</u> ools <u>H</u> elp	
IP Offices	00E007B30B30*
	System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMDR Twinning VCM CCR Codecs LAN Settings Vell Network Topology SIP Registrar Image: Code Code Code Code Code Code Code Code

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

🚏 Avaya IP Office R8 Manager		
Eile Edit View Iools Help IP Offices	☑ 00E007B30B30*	<u>e* - × √ < ></u>
B ♣ BOOTP (1) B © OPECOTS0800 B © System (1) B © System (1) B © System (1) B © Control Link (3) B Exercision (12) B B User (13) B B WarkGroup (1) B B Florid Korapu (1) B B Provise (0) B B Incoming call Route (2) B B Tracening call Route (2) B B Florid Korapu (1) B B Exectory (0) B B B Tracening call Route (1) B Account Code (0) B Execute (1) B Account Code (0) B B User Rights (8) B B B B Warkert (10) B ARS (1) B	System LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR LAN Settings VoiD Network Topology SIP Registrar Network Topology Discovery STUN Server IP Address 0 0 0 STUN Port 3478 = Freewall(NAT Type Open Internet Image Image Ending Refresh Time Image Binding Refresh Time 0 Image Run STUN Cancel Image Public Port 0 Image Run STUN on startup Image Image Image	Twinning VCM CCR Codecs



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.

Kanager File Edit View Tools Help		_ □ ×
IP Offices	☑ 00E007B30B30*	↓ ↓ ↓ ↓ ↓ ↓ ↓
	System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Telephony Tones & Music Call Log Analogue Extensions Defauk Outside Call Sequence Normal ¥ Defauk Dutside Call Sequence Ring Type 1 ¥ Defauk Ring Back Sequence Ring Type 2 ¥ Restrict Analogue Extension Ringer Volkage I Dial Delay Time (secs) 1 1 Dial Delay Count 4 4 Defaulk No Answer Time (secs) 15 4 Hold Timeout (secs) 15 4 Call Priority Promotion Time (secs) 5 4 Defaulk Outrice (secs) 5 5 Defaulk Currency ELR ¥	em Events SMTP SMDR Twinning VCM CCR Codecs

3.3 ARS

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

👫 Avaya IP Office R8 Manager						
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IP Offices	1 E		Main		≓ •	🗙 ✔ < >
B→K BOOTP (1) B→C* Operator (3) B→C* Operator (3) B→C* Operator (3) B→C* Operator (3) B→C* Operator (12) B→C* Operator (13) B→C* Operator	ARS ARS Route Id Route Name Dial Delay Time In Service Time Profile Code ?	S0 Main System Default (1) System Default (1) Chlone> Chlone> Chlone> Chlone> Chlone> Chlone> Chlone> Chlone> Chlone> Chlone	Y Feature Did	Secondary Dial tone SystemTone SystemTone Out of Service Route Out of Service Route Une Group ID 0	<pre> {Vione> {Vione> } </pre>	V Add Remove Edt
	Alternate Route Priority Le	vel 3	T			-
	Alternate Route Wait Time	30		Alternate Route	<none></none>	↓





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.

🖌 Avaya IP Office R8 Manager			_ 🗆 ×
Elle Edit View Loois He	SIP Line - Line	9	Ľ • X • < >
	SIP Line transport SIP UR1 Y0IP T38 Fax SIP Credentials Line Number 9 ITSP Domain Name 172:18:1.61 Prefix	In Service Use Tel URI Check OOS Call Routing Method Originator number for forwarded and twinning calls Name Priority	▼ To Header ▼ System Default ▼





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.

🚏 Avaya IP Office R8 Manager		
File Edit View Tools He	SIP Line - Line 9	☆ • X • < >
BOOTP (1) Gerator (3) Operator (1) Operator (2) Oper	SIP Line [Transport] SIP URI [VoIP] T38 Fax [SIP Credentials] ITSP Proxy Address [72.18.1.61] Network Configuration	सं स

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.

🎦 Avaya IP Office R8 Manager		
File Edit View Tools He	lp	
IP Offices	SIP Line - Line 9	📥 • 🗙 • < >
	SIP Line Transport SIP URI VolP T38 Fax SIP Credentials Channel Groups Via Local URI Contact Display Name PAI Credential Max Calls 1 9000 9000 < *	Add Remove Edit
Leser (13) Leser (13) Leser (13) Leser (13) Leser (14) Leser (15) L	Edit Channel Via Local URI * Contact Use Internal Data Display Name Use Internal Data PA1 Use Internal Data Registration D: Incoming Group 9000 Outgoing Group 9000 Max Calls per Channel 10	OK Cancel



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.

Manager File Edit View Tools He	þ	
IP Offices	E SIP Line - Line 9	☆ • X • < >
	SIP Line Transport SIP URU VoIP T3B Fax SIP Ceclentials Codec Selection System Default Selected Selected G.721 LLAW 64K G.711 LLAW 64K G.711 LLAW 64K G.711 LLAW 64K G.722 64K Selected Selected Selected Fax Transport Support G.711 Selected Selected Call Initiation Timeout (s) G.711 Selected Selected DTMF Support PFC2833 Selected Selected	 □ VoIP Silence Suppression □ Re-invike Supported □ Use Offerer's Preferred Codec □ Codec Lockdown □ PRACK/100rel Supported





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.

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File Edit View Tools Help			
IP Offices		0N;: Dial	📸 • 🗙 🗸 < >
	Short Code		
9X 9N	Code	ON;	
- 10 Service (0) II- 4 RAS (1)	Feature	Dial	
Incoming Call Route (2) WanPort (0)	Telephone Number	ASSN	
- A Directory (0)	Line Group ID	50: Main 💌	
E - W Firewall Profile (1)	Locale		
Account Code (0)	Force Account Code		
Elicense (6)			
User Rights (8) Auto Attendant (0)			
-			

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 SIPto header (for e.g. To: <sip:900@172.18.1.61;user=phone>).

Manager File Edit View Tools Help		
IP Offices	9N: Dial	📸 • 🗙 • < >
X 00; X 00; X 80 X 80	Short Code 9N Code 9N Feature Dial Telephone Number 9Nf@172.18.1.61;user=phone" Line Group ID 9000 Locale Force Account Code Force Account Code	



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.

🌃 Avaya IP Office R8 Manager			
Eile Edit View Iools Help IPOffices	E	0 -454XXX	≝ • X √ < >
⊕ ♣ BOOTP (1) ⊕ ♥ Operator (3) ⊕ ₩ System (1) ⊕ ↑ Line (5) ⊕ ⊂ Control Unit (3) ⊕ € Extension (12) ⊕ ↓ HurkGroup (1) ⊕ ↓ Arabot Code (70) ⊕ ↓ Brown Code (70) ⊕	Standard Voice Recording Bearer Capability Line Group ID Incoming Number Incoming Sub Address Incoming CLI Locale Priority Tag Hold Music Source	Destinations Any Voice 1-454XCX 1-454XCX 1-1-Low 5ystem Source	

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

Y Avaya IP Office R8 Manager Eile Edit View Iools Help			_
IP Offices	0	-454XXX	🛋 • 🗙 ✔ < >
A BOOTP (1) Operator (3) O	Standard Voice Recording Destination	Destination #	Fallback Extension



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.

Manager		_ 🗆 ×
IP Offices	9000	📸 • 🗙 🗸 < >
	Standard Voice Recording Destinations Bearer Capability Arry Voice 1 Line Group ID 9000 1 Incoming Number Incoming Sub Address Incoming CLI Incoming CLI Incoming CLI 1 Locale 1 1 Priority 1 1 Hold Music Source System Source 1	

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.

🗭 XCAPI Configuration	
File View Help	
Configuration	Controller Features
Information Informati	Simulate ECT In cases where the environment does not support call-transfer operations it is possible to simulate call-transfer by call-tromboning (line-interconnect). Simulate ECT by call-tromboning (line-interconnect). Notify destination Tornel signaling information to destination In the dignation of the system in some situations and will be applied to each connection of this controller. Software Codes These features affect the behaviour of the system in some situations and will be applied to each connection of this controller. Always use software modem over audio channels Always use software modem over audio channels



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**.

🕻 XCAPI Configuration	X
File View Help	
Configuration	Options
Information Informat	Payload-type Define the payload-type that should be used to receive telephone-events sent by remote terminals. Payload-Type (0-127) IDI Recommendation





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