

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring CrystalVoice Remote Extension with Avaya Communication Manager and Avaya SIP Enablement Services¹ Server - Issue 1.0

Abstract

These Application Notes describe the steps for configuring CrystalVoice Remote Extension to communicate with Avaya Communication Manager and Avaya SIP Enablement Services Server. The CrystalVoice Remote Extension solution consists of the Remote Extension softphone client and the CrystalVoice ISS/IVX Server. The Remote Extension softphone client communicates with the CrystalVoice ISS/IVX Server via a CrystalVoice propriety protocol; the CrystalVoice ISS/IVX Server then registers the Remote Extension client with Avaya Communication Manager and Avaya SIP Enablement Service Server. Emphasis of the testing was placed on verifying good voice quality from CrystalVoice Remote Extension and its ability to operate with the Avaya SIP Enablement Service Server. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

¹ Beginning with release 3.0 software, the Avaya Converged Communication Server (CCS) has been renamed to the Avaya SIP Enablement Services (SES) Server.

1. Introduction

Avaya Communication Manager and Avaya SIP Enablement Service (SES) Server has the capability to extend advanced telephony features to SIP stations. This feature set can be extended to non-Avaya SIP phones such as CrystalVoice Remote Extension.

These Application Notes describe a solution for configuring CrystalVoice Remote Extension to operate with Avaya Communication Server and Avaya SIP Enablement Service (SES) Server. The CrystalVoice Remote Extension solution consists of two parts, a softphone client that runs on MS Windows and the CrystalVoice ISS/IVX Server that resides in the core network. CrystalVoice Remote Extension communicates with the CrystalVoice ISS/IVX Server via a propriety protocol. In turn, the CrystalVoice ISS/IVX Server registers softphone client(s) to Avaya Communication Manager with the SIP Enablement Services (SES) Server via the standard SIP protocol.

1.1. Configuration

Figure 1 illustrates the configuration used in these Application Notes. The CrystalVoice ISS/IVX Server has two Ethernet connections, one Ethernet is connected into the private core network, and the other Ethernet is connected to the Internet. Two CrystalVoice Remote Extension clients communicate with the CrystalVoice ISS/IVX Server through a Linksys Router via a common Internet IP address through the Internet. The extension numbers used by CrystalVoice Remote Extensions are registered to Avaya Communication Manager via Avaya SIP Enablement Services Server and are also administered as Off-PBX-Telephones stations in Avaya Communication Manager. As a result, each CrystalVoice Remote Extension softphone has access features available from Avaya Communication Manager.



Figure 1: Sample Network Configuration

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2. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided:

Equipment	Software/Firmware
Avaya S8300 Media Server with a G350 Media	Communication Manager 3.0
Gateway	(R0.13x.00.0.340.3) with
	update 00.0.340.3-10458
Avaya SIP Enablement Services Server	3.0 (build 31)
Avaya IA770 INTUITY AUDIX [™]	3.0-1.7
Avaya 4610SW/4620SW IP Telephones	2.2.3
Avaya P333T-PWR Converged Stackable Switch	3.12.1
CrystalVoice Integrated System Services (ISS)	4.2.0.4
CrystalVoice Internet Voice Transcoder for SIP (IVX)	4.2.0.4
Crystal Voice Remote Extension Client w/ MS	4.2.0.3
Windows 2000 Professional	

3. CrystalVoice Remote Extension

There are two parts to the CrystalVoice Remote Extension solution - the CrystalVoice Remote Extension softphone client and the CrystalVoice ISS/IVX Server. The CrystalVoice ISS/IVX Server transcodes traffic between Remote Extension Clients running CrystalVoice's proprietary protocol and the Avaya SIP Enablement Services Server which communicates via SIP. Although this sample configuration shows a dual Ethernet connected ISS/IVX Server, it is possible to configure the ISS/IVX Server with a single Ethernet connection to support both traffic to and from Avaya Communication Manager and the Internet.

From the perspective of the Avaya SIP Enablement Services Server, CrystalVoice Remote Extension clients look like actual SIP telephones all registered from the IP address of the ISS/IVX Server. Since the call-id, from-Tag, and to-tag in the SIP packet header uniquely identify each SIP dialog, the Avaya SIP Enablement Service Server will be able to treat each as separate SIP telephones.

The ISS/IVX Server does not support L2 (802.1Q) or L3 (DiffServ) tagging. Therefore, it is recommended that the Ethernet switch port that the CrystalVoice ISS/IVX Server connects to be configured with the appropriate priority setting, and all intermediate routers to the Avaya SIP Enablement Service Server be configured with the corresponding QoS policy. Both the ISS/IVX Server and the Avaya SIP Enablement Service Server reside on the same network in the sample configuration.

3.1. Configuring the ISS/IVX Server

The following steps describe the configuration for the ISS/IVX Server to communicate with the Avaya Enablement Services Server. This configuration allows for calls to and from Remote Extension clients. A Web Server must be installed on the same machine where the CrystalVoice ISS/IVX server software is running. The sample configuration uses Microsoft IIS.

Step	Description					
1.	Log in to the CrystalVoice ISS/IVX Server via its IP address using the Web browser. The CrystalVoice ISS/IVX installation process automatically changes the Web Server port to 8080.					
	http://50.1.1.200:8080/systemsmanager/home.asp					
	CrystalVoice Systems Manager - Microsoft Internet Explorer					
	Elle Edit View Favorites Tools Help					
	Address Abth://50.1.1.200:8080/systemsmanager/Login.asp					
	Clear Voice Systems Manager Over the					
	Internet Edit Login Logout Home					
	Network System Servers Services Call Routing Accounts Reports					
	The user name or password you entered was invalid. Please try again. If you continue to experience problems contact CrystalVoice technical support for assistance at support@crystalvoice.com Enter Username admin					
	Enter Password Login					
	Done					

Step	Description								
2.	Click Servers on the main menu bar and select Add Server.								
	Note:								
	• For a single Ethernet connection scenario, the IP address must be entered into the Public IP Address field								
	• For the Private IP address field, the CrystalVoice software is designed to								
	recognize private IP addresses as defined in RFC 1918, Address Allocation for								
	Private Internets. The CrystalVoice software is designed to handle calls from								
	both internal/LAN connected and external/Internet connected softphones when the private (LAN ID addresses foll within the recommon detions of DEC 1018. In								
	general this should be the case for most companies. If public IP addresses are								
	used in the private network and there is a public Internet connection with IP								
	addressing in a different public network, for example the LAN is addressed in the								
	50.1.1.x network and the public Internet connection in the 141.150.155.x								
	network, then contact CrystalVoice to discuss the setup.								
	CrystalVoice Systems Manager - Microsoft Internet Evoloper								
	Ejle Edit View Favorites Tools Help								
] ← Back ← → → ⊘ Ø @ @ Search @ Favorites @History ▷ ← @ []								
	Address 🗃 http://50.1.1.200:8080/systemsmanager/ServerEdit.asp?id=1&txtProcessMode=get 💽 🔗 Go 🛛 Links 🎽 🐑 🗸								
	Clear Systems Manager								
	CrystalVoice Over								
	Internet Edit Login Logout Home								
	Network System Servers Services Call Routing Accounts Reports								
	Comune Configuration								
	Server Configuration								
	Edit Server								
	If this server has only one IP address, then put that address in the Public IP Address field and leave the Private IP Address field blank.								
	Server Information								
	Name DEVCON1								
	Public IP Address 141.150.155.69								
	Private IP Address 50.1.1.200								
	Save Delete Back								
	jeg intep://suki/ini/2uu/suku/systemsmanager/nome.asp								

Step	Description							
3.	Select Services \rightarrow ISS from main menu. Click on the server created in Step 2. The							
	sample configuration used DEVCON1 .							
	CrystalVoice Systems Manager - Microsoft Internet Explorer							
	$\begin{array}{c c c c c c c c c c c c c c c c c c c $							
	Address 🖗 http://50.1.1.200:8080/systemsmanager/Iss.asp							
	Clear Systems Manager							
	the Internet Edit Login Logout Home							
	Network System Servers Services Call Routing Accounts Reports							
	ISS Configuration							
	View/Add ISS							
	Click the button on the right to add a new ISS:							
	To view, modify, or delete an existing ISS, click the server name in the display below:							
	ISS							
	Connection Tester							
	Logging Service Presence Manager							
	Service Address Distributor SIP Service							
	Voice Proxy							
	le internet							

Step	Description							
4.	Scroll down to the SIP Service portion. Make sure the red highlighted fields are set as							
	shown. Click Save to complete.							
	🖉 CrystalVoice Systems Manager - Microsoft Internet Explorer							
	Eile Edit View Favorites Tools Help			(E)				
	Gearch → → C C C C C C C C C C C C C C C C C	4 I E						
	Address 2 http://50.1.1.200:8080/systemsmanager/IssEdit.asp?txtProcessMode=ge	et&id=1		🔽 🤗 Go 🛛 Links 🎢 📆 👻				
	Vales Provide Advanced Options)							
	Connection Testers (Chars Advanced Options)			-				
	Connection Tester: (<u>Show Advanced Options)</u>							
	Service Address Distributor: (Show Advanced Options)							
	SIP Service:]				
	SIP Service							
	Local SIP Port:	5060						
	Run SIP Redirect Service:							
	Run SIP Registration Service:							
	Accept SIP Registrations:							
	Send SIP Registrations:							
	SIP Registration Duration (Seconds):	300						
	Send SIP MWI Subscriptions:							
	SIP MWI Subscription Duration (Seconds): 3600							
	Save Delete	Back						
				•				
	Done			🔮 Internet 🥢				
5	After alighing on call Call Douting -> SID Provi	og from the m	oin m	any The following				
5.	SIP Proxy Configuration page will be displayed	Enter all fie	ann m Ids as	shown Make sure				
	the server DEVCON1 entered in step 2 is selected	l. Click Save	to coi	mplete.				
	Public IP Address: Public side IP address of the CrystalVoice ISS/IVX Server							
	Private IP Address: IP address of the Avaya SIP Enablement Services Server.							
	SIP Port : This is the port number used to communicate with the Avaya SIP Enablement							
	Services Server							
	name should be entered in place of <i>devcon.com</i> .	ie configuratio	л. л					

Step		Description						
	In the case of a single Ethernet connection, enter the IP address in the Public IP Address							
	field. Scroll down and click Save to complete.							
	Toucta Waica Suctoms Mapagan - Microsoft Tataupat Euplaner							
	File Edit View Favorites Tools Help	Elie Edit View Favorites Tools Help						
	🚱 Back 🔻 🕤 👻 😰 🏠 🔎 Search	n 📩 Favorites 省 Media 🤣 🍙 - 💺 📄						
	Address 🙋 http://50.1.1.200:8080/systemsn	nanager/SipProxyEdit.asp?id=1&txtProcessMode=get 💽 🎅 Go Links »						
		SIP Proxy Configuration						
		Edit SIP Proxy						
	If this SIP proxy has Public IP Address fiel	only one IP address, then put that address in the d and leave the Private IP Address field blank.						
		SIP Proxy Address						
	Name:	Avaya						
	Public IP Address:	141.150.155.69						
	Private IP Address:	50.1.1.50						
	SIP Port:	5060						
	This is an Outbound SIP Proxy:							
	Request SIP Address:	<sip:#phonenumber#@devcon.com></sip:#phonenumber#@devcon.com>						
	Register Request SIP Address:	<sip: devcon.com=""></sip:>						
	MWI Subscribe Request SIP Address:	<sip:#phonenumber#@devcon.com></sip:#phonenumber#@devcon.com>						
	Use From SIP Address:	<sip:#phonenumber#@devcon.com></sip:#phonenumber#@devcon.com>						
	Use To SIP Address:	<sip:#phonenumber#@devcon.com></sip:#phonenumber#@devcon.com>						
		SIP Proxy Authentication						
	This SI	P Proxy Requires Authentication: 🗹						
	User Name:	#PHONENUMBER#						
	© Use Individual Softphone	e Passwords O Use Password Below For ALL Softphones						
	Password:							
	Confirm Password:							
		SIP Proxy Routing						
	This SIP Proxy will send calls	s to the ISS from its						
	Select one or more I	VXs to handle calls originating from this SIP proxy						
	DEVCON1 - 50 1 1 200-506	Not Selected						
	e	Trusted sites						

6. After clicking on Services → IVX from the main menu and selecting Add a new IVX service, the following screen will appear. The default code set for CrystalVoice ISS/IVX Service is G.711. Click Save to complete. CrystalVoice Systems Hanager - Microsoft Internet Galarer Image: Click Save to complete. CrystalVoice Systems Hanager - Microsoft Internet Galarer Image: Click Save to complete. CrystalVoice Systems Hanager - Microsoft Internet Galarer Image: Click Save to complete. CrystalVoice Systems Hanager - Microsoft Internet Galarer Image: Click Save to complete. Figure 11 1200 0000/systemano april/visit.aspituthrossHude=gridud= Image: Click Save to complete. Figure 11 top: III. 200 0000/systemano april/visit.aspituthrossHude=gridud= Image: Click Save top click S	Step	Description								
service, the following screen will appear. The default code set for CrystalVoice ISS/IVX Server is G.711. Click Save to complete.	6.	After clicking on Services \rightarrow IVX from the main menu and selecting Add a new IVX								
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E GX UND EXPLANT Solution Installed On: Installed On: Edit IVX Installed On: Installed On: Exercise Installed On: Installed Installed On: <t< th=""><th></th><th colspan="9"></th></t<>										
Image: State in the state		CrystalVoice Systems Manager - Microsoft Internet Explorer								
Address Provide Nutripul/90.11.1.000.0000/systemsmanager/tv:dcdt.esp/tudProcessMode=getXdd-3 Edit IVX Idit IVX IVX Information Installed On: IVX Information IVX Type: IVX Type: IVX Self Route: IV Type: IVX Self Route: IP Port Number: 4055 Maximum Channel Capacity: 4 Digit Duration: I20 Reservation Required: Network Type: WAN Output Gain: I100 Service Address Distribution Heartbeat Interval: 3000 Presence Manager Heartbeat Interval: 10000 SIP Configuration Information SIP Configuratio		Image: Search image: Searc								
Edit IVX IVX Information IVX Type: IVX Type: Automatic I IVX Type: Automatic I IP Port Number: 4950 Maximum Channel Capacity: 4 Digit Duration: 120 Reservation Required: Installed On: Network Type: Output Gain: 100 Service Address Distribution Heartbeat Interval: 2000 SIP Configuration Information RIP Port Range Begin: 17000										
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SIP Configuration Information SIP Local SIP Port: 5061 Proxy/Redirect Server: Avaya - 50.1.1.50:5060 ▼ Voice Mail Dialing Patterns: Busy / No Answer: 49000 maximum of 10 characters maximum of 10 characters Login: 49000 maximum of 10 characters maximum of 10 characters RTP RTP Port Range Begin: 17000 RTP RTP Port Range End: 17999 General Audio Source Type: 64K Mulaw (G.711) ▼ Allow SIP Incoming Calls: ▼ Forward On Called IVX Busy: number		Logging and Directory Services Heartbeat Interval:	60000							
Local SIP Port: 5061 Proxy/Redirect Server: Avaya - 50.1.1.50:5060 • Voice Mail Dialing Patterns: Busy / No Answer: 49000 maximum of 10 characters Login: 43000 maximum of 10 characters RTP RTP Port Range Begin: 17000 RTP Port Range End: 17999 General Audio Source Type: 64K Mulaw (G.711) • Allow SIP Incoming Calls: Forward On Called IVX Busy: number		SIP Configuration Inform	nation							
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RTP RTP Port Range Begin: 17000 RTP Port Range End: 17999 General Audio Source Type: 64K Mulaw (G.711) Allow SIP Incoming Calls: Forward On Called IVX Busy: number		Busy / No Answer: 49000	maximum of 10 characters							
RTP Port Range Begin: 17000 RTP Port Range End: 17999 General Audio Source Type: 64K Mulaw (G.711) ▼ Allow SIP Incoming Calls: ✓ Forward On Called IVX Busy: number		Lugin: 143000	maximum of 10 characters							
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Audio Source Type: 64K Mulaw (G.711)		General Audio Source Type: 64K Mulaw (G.711) ▼								
Forward On Called IVX Busy:										
		Forward On Called IVX Busy:	number							
		A Dana								

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Step	Description							
7.	Click on Services \rightarrow PMA from the main menu and Add a new PMA configuration.							
	Select the Server, <i>DEVCON1</i> that was created in Step 2 for the Installed On field. Set							
	PMA state as <i>Active</i> . Leave all other field as default. Click Save to complete.							
	TrustaWaice Systems Manager - Microsoft Internet Evologer							
	File Edit View Favorites Tools Help	File Edit View Favorites Tools Help						
	t → Back → → → 🙆 😰 🚮 🔞 Search 📷 Favorites 👹 History 🔹 → 🎒 💽 📃							
	Address 🙋 http://50.1.1.200:8080/systemsmanager/PmaEdit.asp?txtProcessMode=get&id=2 💽 🄗 Go 🗍 Links 💈	*] 🔁 -						
	Clear Systems Mai							
	Voice Systems Ha							
	the							
	Network System Servers Services Cell Pouting Accounts Perpets	Home						
	Network System Services Can Routing Accounts Reports							
	PMA Configuration							
	Edit PMA							
	PMA Information							
	Installed On:							
	Start Up Type:							
	PMA State:							
	Client Check In Verification Interval: [1000 (Milliseconds)							
	PMA Configuration Check Interval: [15000 (Milliseconds)							
	PM Probation Timeout: 5 (Seconds)							
	PM Failure Timeout: 180 (Seconds)							
	PM Failover Timeout: 360 (Seconds)							
	Save Delete Back							
		-1						
	Internet	11.						

Step	Description						
8.	Click on Accounts → Softphones from the main menu and Add a new softphone						
	account into the system.						
	Phone Name is the Remote Extension client's log in name.						
	Phone Number is the phone number assigned to the softphone client. This phone						
	number should be	the same as phone number assigned for this Softphone in the					
	Avaya SIP Enable	ement Services Server. This sample configuration used 40000					
	as the Phone Nur	nber.					
	Password is the s	ame as the password set in the Avaya SIP Enablement Services					
	Server.	· ·					
	Voice Mail, MW	I, & Transfer are checked.					
	Default IVX is th	e server created under services IVX earlier.					
	Repeat this step for each	Remote Extension Softphone Client that needs to log in to the					
	system. Enter a unique P	hone Name and Phone number for each client.					
	Click Save to complete.						
	CrystalVoice System	s Manager - Microsoft Internet Explorer					
	j <u>Eile Edit View Fa</u>	onites Tools Help					
	Address @ http://50.1.	1.200:8080/systemsmanager/SoftphoneEdit.asp?txtProcessMode=get&id=11 🗸 🔗 Go 🛛 Links » 👘 🗸					
		Softphone Configuration					
		Edit Softphone					
		Softphone Information					
	Phone Name	TestPhone1					
	Phone Number	40000					
	Password	Jackelerer					
	Confirm Passwor						
	Softphone Group	Default Group 💌					
	Phone ID	11					
	Date Created	9/13/2005 9:38:52 AM					
	Date Medified	0/12/2000 11:19:10 44/					
	Date Moullieu						
	First Name						
	Fischame						
	Last Name						
	Email Address	(Optional)					
		Integrated Services					
	Conference						
	Voice Mail						
	Transfer						
		Routing Information					
	Default IVX DEVCONI IVX-SIP (Optional)						
	Priority						
	A Done						

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Step					Descri	ption			
9.	To verify if the C	o verify if the CrystalVoice ISS/IVX is running, click on Network from the main menu.							
	This will display	a list c	of all th	e servic	es and	the current status.	ISS, PM	A. and	IVX-SIP
	should all be liste	ed as R	unning				,	,	
	@ http://50.1.1.200	8080/syste	msmanager.	/NetworkFran	neset.asp - M	icrosoft Internet Explorer			
	<u>Eile E</u> dit <u>V</u> iew F	<u>a</u> vorites <u>T</u> o	ols <u>H</u> elp						1
	🕒 Back 👻 🕤 👻 💌	2 🏠 🔒	🔎 Search 🛛 🚽	😽 Favorites 🛛 🍳	😗 Media 🛛 🥝	🔗 - 🖕 🚍			
	Address 🙆 http://50.	1.1.200:8080	/systemsmana	ger/NetworkFra	meset.asp			• 🔁 Go	Links »
	Cryste	Noice		Clear Voice Over the		Systems	Manager		
				Interne	t	Edit Login Logo	ut Home		
	Network Syste	m Server	s Services	Call Routin	g Account	s Reports			
				Netwo	ork Status	•			
			Last U	lpdated 9/2	21/2005 10):47:09 AM			
				Refresh	every 5 sec	conds.			
				<u>Change</u>	<u>Refresh Ra</u>	<u>te</u>			
				s	erver				
	Server Name:	DEVO	CON1			View Application Event	Log		
	IP Address:	50.1	1.200			View System Event Log Reboot Server			
	Status:	Conr	nected			Reload Configuration Reload Security Keys			
	Service 1	ype	Version	Status	Start Up	Operations			
	ISS Connection Directory Se Logging Ser Presence M Address Dis SIP Service Voice Proxy	Tester rvice vice anager cribution	4.2.0.4	Running	Auto	<u>Stop Pause Restart Moni</u>	tor <u>Edit</u>		
	PMA		4.2.0.4	Running	Auto	Stop Pause Restart Monit	tor <u>Edit</u>		
	IVX-SIP		4.2.0.4	Running	Auto	Stop Pause Restart Monit	tor <u>Edit</u>		
	IVX-H323		4.2.0.4	Running	Auto	Stop Pause Restart Monit	tor <u>Edit</u>		
	ē							usted sites	

3.2. Configuring Remote Extension Client

The following screens highlight the areas that must be set during the installation of CrystalVoice Remote Extension Client. Begin installation of the CrystalVoice Remote Extension Client by running the client installation software "SoftphoneInstall.RE.SIP.exe".

Step	Description
1.	Select the appropriate type of network connection. The sample configuration used <i>Local</i>
	Area Network.
	S Domoto Futencion
	Network
	Please select the type of network connection you have:
	C Cable, xDSL or ISDN
	Colocal Area Network
	C Satellite
	< Back Cancel Cancel
2	Enter the DNS name or IP address of the ISS. In the sample configuration, the
2.	CrystalVoice ISS/IVX Server's IP address was used.
	Remote Extension
	Please enter the DNS name or the IP
	address of your Integrated System Services (ISS):
	141.150.155.69
	<u>A Back</u> <u>Next</u> Cancel

Step		Description
3.	Log in to the (CrystalVoice Remote Extension client.
	Phone name	The Softphone account name configured on the CrystalVoice ISS/IVX Server.
	Password:	The password for the Phone name above, which is the same as
		configured on the Avaya SIP Enablement Services Server for the
		extension that corresponds to the Phone name.
	Below displate testphone2 was	ys the login window of the Remote Extension Client. Phone name as created following step 8 in section 3.1.
		Login to Remote Extension
		CrystalVoice
		Phone name: testphone2
		Password: XXXXX
		Remember my password
		Login Cancel
	CrystalVoice should display	Remote Extension client screen is displayed below. Both L1 and L2 Ready .
		Options Logs Help _ ×
		TestPhone2 x40001 🖂 🏴
		L1 Ready
		L2 Ready
		KEYPAD CONTACTS Clear
		call bksp
		redial
		xfer
		conf
		add edit delete hold
		Acoustic QoS™
		Read No.
		CrystalVoice

4. Configuring the Avaya SIP Enablement Services Server

The following steps describe the configuration of the Avaya SIP Enablement Services server to support Remote Extension Clients.

🍘 denali(Standard Manageme	nt Solutions) - Microsoft In	ternet Explorer		2
<u>File E</u> dit <u>Y</u> iew F <u>a</u> vorites <u>T</u> ools <u>H</u> elp				
G Back 🔹 🕤 👻 😰 🏠	🔎 Search 👷 Favorites 🤞	8 🔗 🕹 🖻		
Address 🛃 https://50.1.1.50/cgi-	bin/unified		🗾 🔁 Go	Links '
AVAYA	Integrat Standard Mana	Integrated Management Jard Management Solutions		
Help Log Off				
· ·	Administration	The administration web interface allows you to administer this Converged Communication Server.	<u>Launch Administrat</u> <u>Web Interface</u>	tion
	Maintenance	The Maintenance Web	<u>Launch Maintenanc</u> Web Interface	<u>e</u>



Check the Add Media Serve	er Extension field at th	ne bottom. Click Add to contin			
🍯 Add User - Microsoft Interne	et Explorer	_ []			
🕒 Back 👻 🕤 👻 😰 🏠	🔇 Back 🔹 🕤 👻 😰 🏠 🔎 Search 👷 Favorites 🔗 🙆 👟 😓				
Address 🙆 https://50.1.1.50/impl	Address 🕘 https://50.1.1.50/impress/do/listusers/add_user				
F(VF(YF)		SIP Server Management			
Help Exit		Server: 50.1.1.50			
Top Lisers	🗧 📩 Add User				
List					
Add	Primary Handle*	40000			
Search	User ID	40000			
Edit	Password*	•••••			
Delete	Confirm Password*	* •••••			
Password	Host*	denali.devcon.com 💌			
Default Profile	First Name*	Remote			
Registered Users	Last Name*	Extension			
Extensions	Address 1	,			
List	Address 2	<u></u>			
Add	Office				
Search	City				
	City				
Media Servers	State				
Services	Country				
Server Configuration	Zip				
IM Logs	Add Media Server Extension				
Evport/Import to BroVis	ion Fields marked * ar	e required			

Step		Description			
4.	Type in the phone number in the	Extension field. Click Add to complete.			
	🖉 Add Media Server Extension - Mi	crosoft Internet Explorer			
	<u>File Edit Vi</u> ew F <u>a</u> vorites <u>T</u> ools <u>H</u> elp				
	Sack ▼ ③ ▼ X 2				
	Address 🙋 https://50.1.1.50/impress/	do/listextension/add_extension			
	AVAYA	Integrated Management			
	Help Exit	Server: 50.1.1.50			
	Top Users List Add Search Edit Delete Password Default Profile Registered Users	Add Media Server Extension Extension* 40000 Media Server S8300 Fields marked * are required. Add			
	é	Trusted sites			

5. Avaya Communication Manager

This section highlights the important commands for defining SIP telephones on Avaya Communication Manager. For complete documentation, see Reference[1][2]. Use the System Access Terminal (SAT) interface to perform these steps. Log in with the appropriate credentials.

5.1. Adding new stations to Avaya Communication Manager

Using the **add station** command, add a station for each SIP phone to be supported. The sample configuration uses 6408D+ for the station type and be sure to include the coverage path for voice mail if it is available. Use the appropriate COS value. Make sure that the station has three (3) "*call-appr*" for **Button Assignment**. Set the IP Softphone to *n*. Repeat the following steps to add additional telephone extensions.

add station 40000	Page 1 of 4
	STATION
Extension: 40000 Type: 6408D+ Port: X Name: SIP40000 STATION OPTIONS	Lock Messages? nBCC: 0Security Code:TN: 1Coverage Path 1: 1COR: 1Coverage Path 2:COS: 1
Loss Group: 2 Data Module? n Speakerphone: 2-way Display Language: english Loss Group: 2 Data Module? n Speakerphone: 2-way	Personalized Ringing Pattern: 1 Message Lamp Ext: 40000 Mute Button Enabled? y Media Complex Ext: IP SoftPhone? n
add station 40000	Page 2 of 4
	STATION
FEATURE OPTIONS	
LWC Reception: audix	Auto Select Any Idle Appearance? n
LWC Activation? y	Coverage Msg Retrieval? y
LWC Log External Calls? n	Auto Answer: none
CDR Privacy? n	Data Restriction? n
Redirect Notification? y	Idle Appearance Preference? n
Per Button Ring Control? n	Bridged Idle Line Preference? n
Bridged Call Alerting? n	Restrict Last Appearance? y
Active Station Ringing: single	Conf/Trans on Primary Appearance? n
H.320 Conversion? n	Per Station CPN - Send Calling Number? y
	Display Client Redirection? n
AUDIX Name: IA770	Select Last Used Appearance? n
	Coverage After Forwarding? s
	Direct IP-IP Audio Connections? y
Emergency Location Ext: 40000	IP Audio Hairpinning? y
- 11 40000	
add station 40000	Page 3 of 4
	STATION
SILE DATA	Upp dept 2 m
	HeadSet? n
	Speaker? II Mounting: d
	Mouncing: a
FICUL:	Cord Length. U
ADDDEVIATED DIALING	Set COIDI.
ADDREVIATED DIALING	2. Lict2.
	2. DISCO.
1: call-annr	5:
2. dall-appr	6.
2. call-appr	7.
3. Carr-appr	1.

AL; Reviewed: SPOC 11/18/2005 Solution & Interoperability Test Lab Application Notes ©2005 Avaya Inc. All Rights Reserved. Use the **change off-pbx-telephone station-mapping** command to map Avaya Communication Manager extension (40000) to the Avaya SIP Enablement Service (SES) Server extension (40000). Select the trunk-group number for the **trunk-group** configured between Avaya Communication Manager and Avaya SIP Enablement Services Server. Select the **Configuration Set** number applicable for this configuration. The sample configuration uses **Configuration Set** 1. For additional information related to Avaya Communication Manager and OFF-PBX-EXTENSION support, please refer to Avaya documentation[2][5][6].

change off-r	pbx-telephone STATION	station-mappi NS WITH OFF-PB	ng 40000 X TELEPHONE	INTEGRATION	Page	1 of	2
Station Extension	Application	Dial Phon Prefix	e Number	Trunk Selection	Config Set	guration	L
40000	OPS	- 4000	0	1	1		
change off-pbx-telephone station-mapping 40000 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION					Page	2 of	2
Station Extension 40000	Call Limit 3 h	Mapping Mode ooth	Calls Allowed all	Bridged Calls both			

Below shows the setting for **configuration-set 1**.

change off-pbx-telephone configuration-set 1		Page	1 of	1
CONFIGURATION SET: 1				
Configuration Set Description:	Remote Extension			
Calling Number Style:	network			
CDR for Origination:	phone-number			
CDR for Calls to EC500 Destination?	У			
Fast Connect on Origination?	n			
Post Connect Dialing Options:	dtmf			
Cellular Voice Mail Detection:	none			
Barge-in Tone?	n			
Identity When Bridging:	principal			

5.2. Verify OPS Capacity

Use the display **system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones** –OPS has been set to a value that will accommodate the number of phones to be supported. Avaya Services has provisioned this during installation according to the system configuration purchased.

```
change system-parameters customer-options
                                                                    Page
                                                                            1 of 10
                                 OPTIONAL FEATURES
    G3 Version: V13
      Location: 1
      Platform: 13
       Location: 1
                                                 RFA System ID (SID): 1
       Platform: 13
                                                 RFA Module ID (MID): 1
                                                                  USED
                                 Platform Maximum Ports: 900
                                                                 48
                                       Maximum Stations: 40
                                                                  20
                               Maximum XMOBILE Stations: 0
                                                                  0
                     Maximum Off-PBX Telephones - EC500: 50
                                                                  0
                     Maximum Off-PBX Telephones - OPS: 50
Maximum Off-PBX Telephones - SCCAN: 0
                                                                  10
                                                                  0
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.3. Configuring Audio Codec

In order for calls to be established successfully, during initial call setup the two end points must agree upon a mutually supported codec. Remote Extension supports both the G.711 (default) and G.729 codecs. G.711 codec was used in the sample configuration (see section 3.1 Step 6 for CrystalVoice ISS/IVX Server setting). To use the G.729 codec, use the **change ip-codec-set** command to set the codec setting to G.729AB, and make a corresponding change to the IVX Server in section 3.1 Step 6.

```
    change ip-codec-set 1
    Page 1 of 2

    IP Codec Set

    Codec Set: 1
    Audio

    Audio
    Silence

    Frames
    Packet

    Codec
    Suppression

    Per Pkt
    Size(ms)

    1:
    G.711MU

    n
    2

    2:
    20
```

5.4. General Test Approach

The general test approach was to attempt calls between Avaya SIP telephones, CrystalVoice Remote Extension softphones and Avaya H.323 IP telephones while exercising features of the telephones such as hold, transfer and conference. Additionally, Remote Extension was tested against many of the Avaya Feature Name Extensions such as Call Park, Call Pick-up, Find-me, and Call Forwarding. Both G.711 and G.729 codecs were exercised as well as shuffling and non-shuffling.

5.5. Test Results

Remote Extension successfully completed test cases for all supported features with the exception of Unattended Transfer. Remote Extension must be the call originator and transferring phone in order to execute an Unattended Transfer among SIP telephones. In addition, the CrystalVoice ISS/IVX Server does not perform any L2 (802.1Q) or L3 (DiffServ) tagging for its traffic. Therefore, it is recommended that appropriate priority setting be configured on the Ethernet switch port where the CrystalVoice ISS/IVX Server is connected to and corresponding policy be set on all intermediate

AL; Reviewed: SPOC 11/18/2005 routers in the case where the CrystalVoice ISS/IVX Server is situated in a different network from that of Avaya SIP Enablement Service (SES) Server to ensure proper Quality of Service.

6. Verification Steps

The following steps may be used to verify the configuration:

- Log in to the CrystalVoice Systems Manager via the web browser and select Network from the main menu. The following three (3) services should be in a running state.
 i) ISS
 - i) ISS
 - ii) PMA
 - iii) IVX-SIP
- Log in to the Avaya SIP Enablement Service (SES) Server via the Web browser. The registered users field under Users will also show all registered SIP users including all the registered Remote Extension users.
- Place call from CrystalVoice Remote Extension Client.

7. Support

For technical support on the CrystalVoice product line, contact CrystalVoice Communications at support@CrystalVoice.com or 1-805-889-4260

8. Conclusion

These Application Notes have described the administration steps required to support CrystalVoice Remote Extension on Avaya Communication Manager with the Avaya Enablement Service (SES) Server. With the exception of Unattended Transfer among SIP telephones, the CrystalVoice Remote Extension supported all basic and extended features that were tested and can interoperate successfully with Avaya Communication Manager SIP solution.

9. Additional References

- [1] Administrator Guide for Avaya Communication Manager, Doc # 03-300509, Issue 1, June 2005
- [2] Avaya Communication Manager Advanced Administration Quick Reference, Doc # 03-300364, Issue 2, June 2005 Release 3.0
- [3] Expanded Meet-me Conference (EMMC) version 1.0 Installation and Troubleshooting Guide for the S8500, Doc # 04-300527, Issue 1, June 2005
- [4] Avaya IA 770 INTUITY AUDIX Messaging Application, Doc # 585-313-159, Issue 4, December 2003
- [5] Converged Communications Server Installation and Administration, Doc # 555-245-705, February, 2004
- [6] Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0, Doc # 210-100-500, Issue 9, June 2005
- [7] CrystalVoice Remote Extension for SIP Install Notes, Doc #5900-1045
- [8] CrystalVoice Systems Manager Reference Guide, Doc #5900-1029

Product documentation for Avaya products may be found at <u>http://support.avaya.com</u>

Product documentation for CrystalVoice products may be found at <u>http://www.crystalvoice.com</u>

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