

Avaya Solution & Interoperability Test Lab

Application Notes for the Citel Gateway with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services, and digital telephones connected to a Citel Gateway. Citel Gateway is a SIP-based VoIP appliance. During compliance testing, digital telephones connected to Citel Gateway successfully registered with Avaya SIP Enablement Services, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as three-way conference, transfers, hold, etc. 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. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and digital telephones connected to a Citel Gateway. Citel Gateway is a SIP-based VoIP appliance. During compliance testing, digital telephones connected to the Citel Gateway successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as three-way conference, transfers, hold, etc. During compliance testing, Nortel Meridian, Nortel Norstar, NEC and Panasonic model TDM telephones were connected to the Citel Gateway and these telephones appeared as SIP endpoints to Avaya SES. TDM telephones connected to Citel Gateway will be referred to as Citel Gateway Handsets. These application notes do not cover configuration of Citel Gateway Handsets.

Figure 1 illustrates a sample configuration consisting of a pair of Avaya S8710 Media Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and the Citel Gateways. Avaya Communication Manager is installed on the Avaya S8710 Media Servers. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the SIP-based Citel Gateway and Avaya SIP, H.323, and digital telephones. The analog PSTN telephone is also included to demonstrate calls routed by Avaya Communication Manager between the Citel Gateway and the PSTN.

Citel Gateway originates a call by sending a call request (SIP INVITE message) to the Avaya SES server. The Avaya SES server routes the call over a SIP trunk to Avaya Communication Manager for origination services. If the call is destined for another local SIP telephone, then Avaya Communication Manager routes the call back over the SIP trunk to Avaya SES server for delivery to destination SIP telephone. Otherwise, Avaya Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog telephone, an adjunct, a vector, a hunt group, etc., depending on the destination number.

For a call arriving at Avaya Communication Manager that is destined for the Citel Gateway, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES server for delivery to the Citel Gateway.

These application notes assume that Avaya Communication Manager and Avaya SES are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult [3] and [4].

Notes:

- During compliance testing, only the functionality of the Citel Gateway was verified.
- Citel Gateway only supports one phone range at a time. Following configurations were verified:
 - **Test 1** in **Figure 1** used Meridian on Citel Gateway 1 and Norstar on Citel Gateway 2.
 - Test 2 in Figure 1 used NEC on Citel Gateway 1 and Panasonic on Citel Gateway 2.

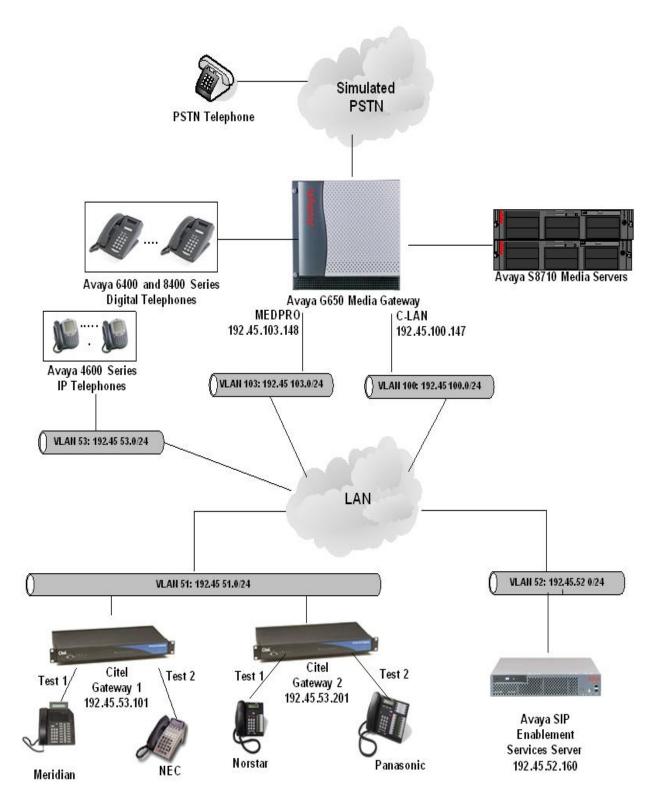


Figure 1: Sample configuration

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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 S8710 Media Server	Avaya Communication Manager 3.1.2
	(R013x.01.2.632.1)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12 FW 31
TN799DP C-LAN Interface	HW01 FW 17
TN2302AP IP Media Processor	HW20 FW 112
Avaya SIP Enablement Services Server	SES 3.1.1(R03.1.1-03.1.114.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323)
	2.5 (4625SW H.323)
	2.2.2 (4610SW SIP)
Avaya 6400 and 8400 Series Digital Telephones	-
Avaya Analog Telephone	-
Citel Gateway	3.61.4d07 HSGW

3. Configure Avaya Communication Manager

This section describes a procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES which includes steps for setting up IP codec sets, an IP network region, a signaling group and its interface. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. Also, a procedure is described here to configure SIP telephones in Avaya Communication Manager. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Citel and other SIP telephones are configured as off-PBX telephones in Avaya Communication Manager. Avaya Communication Manager does not directly control an off-PBX telephone but its features and calling privileges can be applied to it by associating a local, on-PBX telephone with the off-PBX telephone. Similarly, a SIP telephone in Avaya SES is associated with an on-PBX telephone on Avaya Communication Manager. SIP Telephones register with the Avaya SES and use Avaya Communication Manager for call origination and termination services. Throughout the rest of this document, on-PBX telephones associated with SIP telephones in such a manner will be referred to as Outboard Proxy SIP (OPS) stations.

3.1. Capacity Verification

Step	Description									
1.	Enter the display system-parameters customer-options command.	Verify that	t the	e are						
	sufficient Maximum Off-PBX Telephones – OPS licenses. If not,	contact an a	autho	rized						
	Avaya account representative to obtain additional licenses.									
	display system-parameters customer-options OPTIONAL FEATURES	Page 1	of	LO						
	G3 Version: V13 Location: 1 RFA System ID (Platform: 8 RFA Module ID (
	Platform Maximum Ports: 44000 Maximum Stations: 36000 Maximum XMOBILE Stations: 0 Maximum Off-PBX Telephones - EC500: 5 Maximum Off-PBX Telephones - OPS: 200 Maximum Off-PBX Telephones - SCCAN: 0									
2.	Proceed to Page 2 of OPTIONAL FEATURES form. Verify that the trunks supported by the system is sufficient for the number of SIP true contact an authorized Avaya account representative to obtain addition. Note: <i>Each SIP call between two SIP endpoints (whether internal or two SIP trunks for the duration of the call. The license file installed of the maximum permitted.</i>	unks needeo onal licenses • external) r	d. If 1 5. <i>requit</i>	iot, res						
	display system-parameters customer-options OPTIONAL FEATURES	Page	2 0	E 10						
	IP PORT CAPACITIES Maximum Administered H.323 Trunks: 200 Maximum Concurrently Registered IP Stations: 1000 Maximum Administered Remote Office Trunks: 0 Maximum Concurrently Registered IP eCons: 0 Maximum Concurrently Registered IP eCons: 0 Maximum Concurrently Registered IP eCons: 0 Maximum Video Capable H.323 Stations: 0 Maximum Video Capable H.323 Stations: 0 Maximum Video Capable IP Softphones: 0 Maximum Administered SIP Trunks: 200 Maximum Number of DS1 Boards with Echo Cancellation: 0 Maximum TN2501 VAL Boards: 1	USED 148 2 0 0 0 0 0 0 0 153 0 1								
	Maximum G250/G350/G700 VAL Sources: 0 Maximum TN2602 Boards with 80 VoIP Channels: 2 Maximum TN2602 Boards with 320 VoIP Channels: 2 Maximum Number of Expanded Meet-me Conference Ports: 0	0 0 1 0								

3.2. IP Codec Set

This section describes the steps for administering a codec set in Avaya Communication Manager. This codec set is used in the IP network region for communications between Avaya Communication Manager and Avaya SES.

Step	Description	
1.	Enter the change ip-codec-set < c > command, where c is a minclusive. IP codec sets are used in Section 3.3 for configuring specify which codec sets may be used within and between new compliance testing, G.711MU and G.729AB were used and to none as encryption is currently not supported for SIP telep	ng an IP network region to etwork regions. For the Media Encryption was set
	change ip-codec-set 2	Page 1 of 2
	IP Codec Set	
	Codec Set: 2	
	AudioSilenceFramesPacketCodecSuppressionPer PktSize(ms)1:G.711MUn2202:G.729ABn2203:4:5:6:7:	
	Media Encryption 1: none 2: 3:	

3.3. IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description							
1.	Enter the change ip-network-region < n > command, where n is a number between 1 and							
	250 inclusive and configure the following:							
	• Authoritative Domain – Set to the devconnect.com. This should match the SIP							
	Domain value in Section 4, Step 2.							
	 Intra-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio 							
	e ·							
	connectivity between endpoints registered to Avaya Communication Manager or							
	Avaya SES in the same IP network region.							
	• Codec Set – Set the codec set number as provisioned in Section 3.2.							
	• Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio							
	connectivity between endpoints registered to Avaya Communication Manager or							
	Avaya SES in different IP network regions.							
	change ip-network-region 2 Page 1 of 19							
	IP NETWORK REGION							
	Region: 2							
	Location: Authoritative Domain: devconnect.com							
	MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes							
	Codec Set: 2 Inter-region IP-IP Direct Audio: yes							
	UDP Port Min: 2048 IP Audio Hairpinning? y							
	UDP Port Max: 65535							
	DIFFSERV/TOS PARAMETERS RTCP Reporting Enabled? y Call Control PHB Value: 46 RTCP MONITOR SERVER PARAMETERS							
	Audio PHB Value: 46 Use Default Server Parameters? y							
	Video PHB Value: 26							
	802.1P/Q PARAMETERS							
	Call Control 802.1p Priority: 6							
	Audio 802.1p Priority: 6 Video 802.1p Priority: 5 AUDIO RESOURCE RESERVATION PARAMETERS							
	H.323 IP ENDPOINTS RSVP Enabled? n							
	H.323 Link Bounce Recovery? y							
	Idle Traffic Interval (sec): 20							
	Keep-Alive Interval (sec): 5							
	Keep-Alive Count: 5							

Step					D	escription			
2.	betw	een re	egions a	s per be	U	U U	l enable inter-region co g, codec set was set to		•
	Page	3 of	19	Ir	nter Network Re	egion Connectio	n Management		
		dst rgn 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15	codec set 2 2	direct WAN Y		Video WAN-BW-limits	Intervening-regions	Dyn CAC	IGAR n

3.4. IP Node Names

This section describes the steps for setting an IP node name for Avaya SES in Avaya Communication Manager.

Step		Description	
1.	Enter the change with its IP addres	node-names ip command and add a node n s.	ame for Avaya SES along
	change node-name	es ip	Page 1 of 1
		IP NODE NAMES	
	Name	IP Address	
	CLAN-1A06	192.45 .100.147	
	MEDPRO-1A13	192.45 .103.148	
	SES	192.45 .52 .160	

3.5. SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

Step	Description
1.	 Issue the command add signaling-group <s>, where s is an available signaling group and configure the following:</s> Group Type – Set to sip. Transport Method – Set to tls. Near-end Node Name - Set to CLAN name as displayed in Section 3.4. Far-end Node Name - Set to Avaya SES name configured in Section 3.4. Far-end Network Region - Set to the region configured in Section 3.3. Far-end Domain - Set to the devconnect.com. This should match the SIP Domain value in Section 4, Step 2. DTMF over IP – Set to rtp-payload. Direct IP-IP Audio Connections – Set to y for shuffling.
	add signaling-group 10 Page 1 of 5 SIGNALING GROUP Group Number: 10 Group Type: sip Transport Method: tls
	Near-end Node Name: CLAN-1A06 Far-end Node Name: SES Near-end Listen Port: 5061 Far-end Listen Port: 5061 Far-end Domain:devconnect.com Far-end Network Region: 2 Bypass If IP Threshold Exceeded? n
	DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y Session Establishment Timer(min): 120 IP Audio Hairpinning? n

3.6. SIP Trunking

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

p						Desc	ription							
	Issue th	ie comn	nand ad	d trunk	-group	o <t< b="">>, v</t<>	where t	is an ı	ınallocat	ed trun	k gro	oup a	nd	
	configu	ire the f	ollowing	<u>z:</u>	•						U	•		
	•	Group	Type –	Set to the	ne Gro	ουρ Τνι	be field	value	configu	red in S	Sectio	on 3.	5.	
		-	• 1						trunk acc					
					,		•		ld value			in Se	ectio	n
		3.5 .		ap bei		Oroup	, i vuint		iu value	comge	urcu i			11
	•	Numbe	er of Me	mbers	– Allov	wed val	ues are	betw	een 0 and	1 255. 8	Set to	a va	alue	
									telephone					
	1	inge ei	iougn to	acconn	nouale	, the nu		1 DH	cicpiion	- UNIONS	510115	JULI	18 US	
		A	NT.	F (1	• .•								se
	Note: 1	Each SI		tween t	vo SIP	endpo	ints (wh	nether	internal e installe			-		
-	Note: I two SII the max	Each SII P trunks ximum p	P call be for the opermitted	tween t duratior	vo SIP	endpo	ints (wh	nether		d on th	e sys	tem	cont	s ra
-	Note: I two SII the max	Each SII P trunks	P call be for the opermitted	tween t duratior	wo SIP n of the	endpo	ints (wł he licer	nether		d on th		tem		s ro
-	Note: I two SII the max add tru	Each SII P trunks <u>ximum p</u> unk-gro	P call be for the opermitted up 10	tween t duratior	wo SIP n of the	endpo call. T	ints (wh he licer	tether 1se filo	e installe	d on the Pa	e sys	<i>tem</i> 0	of	s rc
-	Note: I two SII the max add tru Group I	Each SII P trunks <u>ximum p</u> unk-gro Number:	P call be for the opermitted up 10	tween t duratior l.	wo SIP a of the	endpo call. T	ints (wh he licer	nether ise filo sip		d on the Par CDR	e sys	<i>tem</i> 0	of y	s rc 2
-	Note: I two SII the max add tru Group I Group I Dire	Each SII P trunks ximum p unk-gro Number: p Name: ection:	P call be for the opermitted up 10	tween to duration l. 5-DevCor	wo SIP a of the T	endpo call. T	ints (wh he licer ROUP Type: COR:	sip	e installe	d on the Pac CDR 1	e sys	1 rts: IAC:	of y	s rc 2
	Note: I two SII the max add tru Group I Group I Dire Dial	Each SII P trunks ximum p unk-gro Number: p Name: ection: Access?	P call be for the opermitted up 10 10 SIP-SES two-way n	tween to duration l. 5-DevCor	wo SIP a of the T	endpo e call. T RUNK GH Grouj	ints (wh he licer ROUP Type: COR:	sip	e installe	d on the Par CDR	e sys	1 rts: IAC:	of y	s rc 2
-	Note: I two SII the max add tru Group I Group I Dire Dial i Queue I	Each SII P trunks ximum p unk-gro Number: p Name: ection: Access? Length:	P call be for the opermitted up 10 10 SIP-SES two-way n 0	tween to duration l. 5-DevCor	wo SIP a of the T	P endpo e call. T RUNK GH Groug oing D:	ints (wh he licer ROUP Type: COR: .splay?	sip n	e installe	d on the Pac CDR 1	e sys	1 rts: IAC:	of y	s ro 2
	Note: I two SII the max add tru Group I Group I Dire Dial i Queue I	Each SII P trunks ximum p unk-gro Number: p Name: ection: Access?	P call be for the opermitted up 10 10 SIP-SES two-way n 0	tween to duration l. 5-DevCor	wo SIP a of the T	P endpo e call. T RUNK GH Groug oing D:	ints (wh he licer ROUP Type: COR:	sip n	e installe	d on the Pac CDR 1	e sys	1 rts: IAC:	of y	s ro 2
-	Note: I two SII the max add tru Group I Group I Dire Dial i Queue I	Each SII P trunks ximum p unk-gro Number: p Name: ection: Access? Length:	P call be for the opermitted up 10 10 SIP-SES two-way n 0	tween to duration l. 5-DevCor	wo SIP a of the T	P endpo e call. T RUNK GH Groug oing D:	ints (wh he licer ROUP Type: COR: .splay?	sip n	e installe TN: Ni	d on the Pac CDR 1	e sys	1 rts: TAC: e:	of y 110	s ro 2

3.7. SIP Stations

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers of the Citel Gateway Handsets.

Step		Description	
1.		d, where s is an available exter Page 1 of the STATION form estration Without Hardware (A to Avaya Communication Mar	, configure the WOH) as SIP stations
	add station 54020	STATION	Page 1 of 3
	Extension: 54020 Type: 6408D+ Port: X Name: CGH54020	Lock Messages? Security Code: Coverage Path 1: Coverage Path 2: Hunt-to Station:	n BCC: 0 TN: 1 COR: 1 COS: 1
	STATION OPTIONS Loss Group: 2 Data Module? n Speakerphone: 2-way Display Language: engli	Mute But	ging Pattern: 1 age Lamp Ext: 54008 ton Enabled? y
			Complex Ext: P SoftPhone?y
2.	Proceed to Page 3 of the STATIO entries in BUTTON ASSIGNMEN match the Call Limit field value in	NT field. The number of call a	
	add station 54020	STATION	Page 3 of 3
	SITE DATA Room: Jack: Cable: Floor: Building:	Spea	-
	ABBREVIATED DIALING LIST1:	List2:	List3:
	BUTTON ASSIGNMENTS 1: call-appr 2: call-appr 3: 4:	5: 6: 7: 8:	

Step			D	escription				
3.	following:			n-mapping comr		U		'n
	1. • Applic	ation – Set to	OPS.			C		-
	registra the san	ation and call ne as the Stat	termination ion Extensi	ber that the Citel In the example on, but is not required nk group number	below, the P uired to be th	'hone N ne same	umber i	
	add off-pbx-t	-		ng 54008 -PBX TELEPHONE I	NTEGRATION	Page	1 of	2
	Station Extension 54008	Application OPS	Prefix	none Number 4008	Trunk Selection 10		iguratio	n
4.				orm and verify that configured in St e		imit fie	ld value	
	add off-pbx-t	-		ng 54008 -PBX TELEPHONE I	NTEGRATION	Page	2 of	2
	Station Extension	Call Limit	Mapping Mode	Calls Allowed Allowed	Bridg Calls			
	54008	2	both	all	both		1	
5.	Repeat Steps 2 for Citel Gates		-	nister additional (OPS stations	and ass	sociation	S

4. Configure Avaya SIP Enablement Services

This section describes the steps for creating SIP trunk between Avaya SES and Avaya Communication Manager. Also, SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. The Citel Gateway will register with Avaya SES using the SIP user accounts. A separate SIP account will be created for each TDM telephone connected to Citel Gateway.

Configuration in the following steps is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

		Description	
Open a web brows	ser, enter http://-	<ip address="" avaya="" of="" s<="" ses="" th=""><th>erver>/admin for the URL,</th></ip>	erver>/admin for the URL,
		edentials. Click on the Lau	
Interface link upc			
1			
On the SIP Server	0	10	
• Click the +	 sign to expand 	the options under Server C	Configuration.
• Click Syste	em Properties.		
• Verify the	SIP Domain m	atches the Far-end Domain	field value configured for
•		aya Communication Manag	6
the signam	ng group on Av	aya Communication Manage	
Edit System Properties - Microso	oft Internet Explorer		
File Edit View Favorites Tools			
😓 Back 🔹 🤿 🙆 🙆 🚳	Search 🙀 Favorites 🎯 Media	3 B- 3 I B	
Address 🙋 https://192.45.52.160/imp	ress/do/thishost/this_host		💌 🔗 🌀 SnagIt 🛃
AVAVA			Integrated Management 🌥
F(VF(YF)			SIP Server Management
Help Exit			Server: 192.45.52.160
Тор			
• Users	Tedit System I	Properties	
Conferences	SES Version	SES-3.1.1.0-114.0	
Media Server Extensions	System Configuration	simplex	
Emergency Contacts	Host Type	home/edge	
 Hosts Media Servers 			
Adjunct Systems	SIP Domain*	devconnect.com	
Services	Note that the DNS doma If you are unsure about	an is: devconnect.com this field, most often the SIP	
Server Configuration	domain should be the ro	ot level DNS domain. For example,	
System Properties		tcoast.example.com, the SIP onfigured to example.com. This	
Admin Accounts	allows SIP calls and inst	ant messages to users with handles	
License	of the format handle@e>	(ample.com	
IM Log Settings SNMP Configuration	License Host*	localhost	
Certificate Management			
IM Logs	Network Properties		
• Trace Logger	Local IP	192.45.52.160	
Export/Import to ProVision	Local Name	SES-DevCon1.devconnect.com	
Update	Logical IP	192.45.52.160	
	Logical Name Gateway IP Address	SES-DevCon1.devconnect.com 192.45.52.1	
	Gateway IP Address	192.45.52.1	
	Redundant Properties		
		CAMP	
	Management Device	SAMP	-

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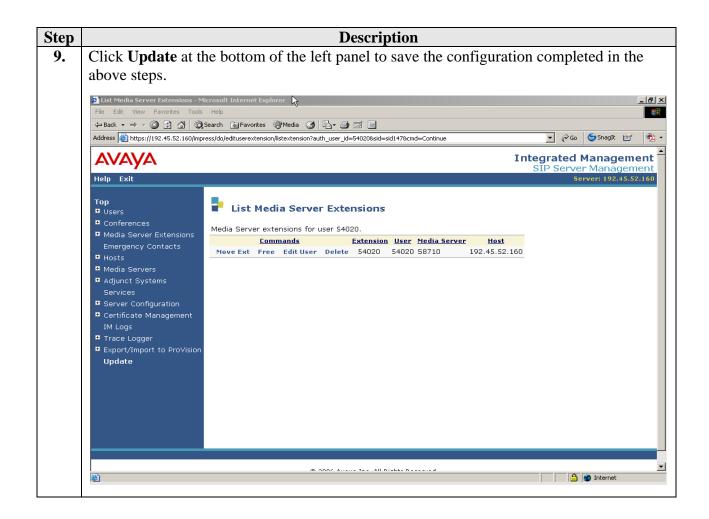
			Ι	Description		
٢	To enabl	e secure	SIP trunking between	Avaya SES and Ava	ya Communication Ma	mager,
			U	•	Manager from the SIP	0
			ent page:	aya communication		
k		0	10		~	
	• (lick the -	 sign to expand the op 	otions under Media S	Servers.	
	• (lick Add	in left pane under Me	dia Servers.		
			1			
			s - Microsoft Internet Explorer			_101 ×
		- Eavorites Tool	Help Search 📷 Favorites 🛞 Media 🎯 🖂 🕶 🖉			199
		s://192.45.52.160/im	ress/do/acp/top		- Poo S	122
	Help Exit	A			Integrated Manage SIP Server Manage Server: 192.4	gement
	тор		-		361Vet. 192.5	43.32.100
	 Users Conference 	ac .	Manage Media Server	Interfaces		
				a server interfaces . server interface .		
	# Hosts		Conservations of the other of particular throad and the State Case			
	Media Serv List	rers				
	Add Adjunct Sy					
	Services Server Cor					
	IM Logs	Management				
	 Trace Logo Export/Imp 	ger port to ProVisior				
	Update					
	http://www.av					
	for conn • S • S	dd Medi ectivity to SIP Trun SIP Trun	a Server Interface page Avaya Communication K Link Type - Set to the K IP Address - Set to the	ons Manager: he Transport Meth e the CLAN IP address	runk parameters as foll od field value in Sections s as displayed in Sections	on 3.5. on 3.4
	for conn • S • S • C	dd Medi ectivity to SIP Trun SIP Trun Click Add	a Server Interface page Avaya Communication K Link Type - Set to the K IP Address - Set to the	ge, provision SIP Tr ons Manager: he Transport Meth e the CLAN IP address	runk parameters as foll od field value in Section	on 3.5. on 3.4
	for conn • S • S • C	dd Medi ectivity to SIP Trun SIP Trun	a Server Interface page Avaya Communication K Link Type - Set to the K IP Address - Set to the	ge, provision SIP Tr ons Manager: he Transport Meth e the CLAN IP address	runk parameters as foll od field value in Sectio s as displayed in Sectio	on 3.5. on 3.4.
	for conn S S C S Add Media Sc	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication K Link Type - Set to the K IP Address - Set to the when finished and the	ge, provision SIP Tr ons Manager: he Transport Meth e the CLAN IP address	runk parameters as foll od field value in Sectio s as displayed in Sectio	on 3.5. on 3.4. e [not
t	for connu S S C S S C S S S S S S S S S S S S S	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication k Link Type - Set to the k IP Address - Set to the when finished and the crosoft Internet Explorer	ge, provision SIP Tr ons Manager: he Transport Meth e the CLAN IP address en click Continue or	runk parameters as foll od field value in Sectio s as displayed in Sectio	on 3.5. on 3.4. e [not
1	for conne S S C S Add Media Sc File Edt View Galactic S S S S S S S S S S S S S S	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication K Link Type - Set to the K IP Address - Set to the when finished and the	ge, provision SIP Tr ons Manager: he Transport Meth e the CLAN IP address en click Continue or	runk parameters as foll od field value in Sectio s as displayed in Sectio	on 3.5. on 3.4. e [not
1	for conne S S C S Add Media Sc File Edt View January Back - P Address @ http:	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication k Link Type - Set to the k IP Address - Set to the when finished and the crosoft Internet Explorer Help (Search Fevorites @Media @ Ex- c	ge, provision SIP Tr ons Manager: he Transport Meth e the CLAN IP address en click Continue or	runk parameters as foll od field value in Section s as displayed in Section the confirmation page	on 3.5. on 3.4. e [not
t	for conne S S C S Add Media Sc File Edt View Galactic S S S S S S S S S S S S S S	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication k Link Type - Set to the k IP Address - Set to the when finished and the crosoft Internet Explorer Help (Search Fevorites @Media @ Ex- c	ge, provision SIP Tr ons Manager: he Transport Meth e the CLAN IP address en click Continue or	runk parameters as foll od field value in Section s as displayed in Section the confirmation page	on 3.5. on 3.4. e [not
1	for conne S S S Add Media 50 File Edit View P Back - P Address There Help Exit	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication k Link Type - Set to the k IP Address - Set to the when finished and the crosoft Internet Explorer Help Search Fravorites Media (2) (2) - 6 ress/do/listacp/add_acp	ge, provision SIP Tr ons Manager: he Transport Methe the CLAN IP address en click Continue or	runk parameters as foll od field value in Section s as displayed in Section the confirmation page the confirmation page	on 3.5. on 3.4. e [not
1	for conne S S Add Media S File Edit View Jadress Thtp: Help Exit Top Users	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication k Link Type - Set to the k IP Address - Set to the when finished and the crosoft Internet Explorer Help (Search Fevorites @Media @ Ex- c	ge, provision SIP Tr ons Manager: he Transport Methe the CLAN IP address en click Continue or	runk parameters as foll od field value in Section s as displayed in Section the confirmation page the confirmation page	on 3.5. on 3.4. e [not
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1	for conne S S Add Media S File Edit Viev Back +	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication Link Type - Set to the IP Address - Set to the when finished and the crosoft Internet Explorer Help Search Favorites Media 2 - C ress/do/listacp/add_acp	ge, provision SIP Tr ons Manager: he Transport Methe the CLAN IP address en click Continue or	runk parameters as foll od field value in Section s as displayed in Section the confirmation page the confirmation page	on 3.5. on 3.4. e [not
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t	for conne S S Add Media Se File Edit View → Back → → Address Thtp: Nelp Exit Top ■ Users ■ Conference ■ Media Serv Emergency ■ Hosts ■ Media Serv List	dd Medi ectivity to SIP Trun SIP Trun Click Add hown].	A Server Interface page Avaya Communication K Link Type - Set to the K IP Address - Set to the When finished and the when finished and the server Interface Name* Host [192: SIP Trunk SIP Trunk Link Type T	ge, provision SIP Tr ons Manager: the Transport Methe the CLAN IP address en click Continue or click Continue or serface	runk parameters as foll od field value in Section s as displayed in Section the confirmation page the confirmation page	on 3.5. on 3.4. e [not
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t	for conne S S Add Media Sc File Edt Ver Dadress There Help Exit Top Help Exit Top Help Exit Conference Media Services Add Addunct Sy Services	dd Medi ectivity to SIP Trun SIP Trun Click Add hown]. • • • • • • • • • • • • • • • • • • •	A Server Interface page Avaya Communication Link Type - Set to the IP Address - Set to the when finished and the crosoft Internet Explorer Help Saarch Favorites Media IP - C ress/do/listacp/add_acp Add Media Server Interface Name* Host 1922 SIP Trunk Link Type C T SIP Trunk Link Type C T SIP Trunk Link Type C T SIP Trunk Link Address (see Help) Media Server Admin Password Media Server Admin Password	ge, provision SIP Tr ons Manager: the Transport Methe the CLAN IP address en click Continue or click Continue or serface	runk parameters as foll od field value in Section s as displayed in Section the confirmation page the confirmation page	on 3.5. on 3.4. e [not
t	for conne S S Add Media Sc File Edt Ver Dadress There Help Exit Top Help Exit Top Help Exit Conference Media Services Add Addunct Sy Services	dd Medi ectivity to SIP Trun SIP Trun Click Add hown]. • • • • • • • • • • • • • • • • • • •	A Server Interface page Avaya Communication Link Type - Set to the IP Address - Set to the when finished and the crosoft Internet Explorer Help Saarch Favorites Media IP - C ress/do/listacp/add_acp Add Media Server Interface Name* Host 1922 SIP Trunk Link Type C T SIP Trunk Link Type C T SIP Trunk Link Type C T SIP Trunk Link Address (see Help) Media Server Admin Password Media Server Admin Password	ge, provision SIP Tr ons Manager: the Transport Methe the CLAN IP address en click Continue or click Continue or serface	runk parameters as foll od field value in Section s as displayed in Section the confirmation page the confirmation page	on 3.5. on 3.4. e [not
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t	for conne S S Add Media Sc File Edt Ver Dadress There Help Exit Top Help Exit Top Help Exit Conference Media Services Add Addunct Sy Services	dd Medi ectivity to SIP Trun SIP Trun Click Add hown]. • • • • • • • • • • • • • • • • • • •	A Server Interface page A Avaya Communication A Link Type - Set to the A IP Address - Set to the when finished and the crosoft Internet Explorer Help Search Favorites Media @ Favorites Media Server Interface Name* Host 192: SIP Trunk Link Type T SIP Trunk IP Address* SIP Trunk IP Address* Media Server Admin Address (see Help) Media Server Admin Password Media Server	ge, provision SIP Tr ons Manager: the Transport Methe the CLAN IP address en click Continue or click Continue or serface	runk parameters as foll od field value in Section s as displayed in Section the confirmation page the confirmation page	on 3.5 on 3.4 e [not

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tep			Description					
5.	In the left pane of the SIP Server Management page, expand Users and click Add .							
	🚰 Top - Microsoft Internet Explore	r						
	<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites <u>T</u> ools	Help						
	$\leftarrow \operatorname{Back} \to \to \bigotimes \boxtimes \boxtimes \boxtimes \boxtimes$		lia 🧭 🖏 🎒 🖬 🗐					
	Address) https://192.45.52.160/impi	ress/do/top/top		 ✓ ở 				
	AVAYA			Integrated Management SIP Server Management				
	Help Exit			Server: 192.45.52.160				
	Top ■ Users	🗗 Тор						
	List Add	Manage Users	Add and delete Users.					
	Search	Manage Conferencing	Add and delete Conference Extensions.					
	Edit Delete	Manage Media Server Extensions	Add and delete Media Server Extensions.					
	Password Default Profile	Manage Emergency Contacts	Add and delete Emergency Contacts.					
	Registered Users Conferences	Manage Hosts	Add and delete Hosts.					
	Media Server Extensions	Manage Media Servers	Add and delete Media Servers.					
	Emergency Contacts Hosts	Manage Adjunct Systems	Add and delete Adjunct Systems.					
	 Media Servers Adjunct Systems 	Manage Services	Start and stop server processes on this host.					
	Services	Server Configuration	Edit Properties of the system.					
	 Server Configuration Web Certificate 	Certificate Management	Manage Web Certificate.					
	Management	IM Logs	Download IM Logs.					
	IM Logs Trace Logger	Trace Logger	Manage SIP Trace Logs.					
	 Frace Logger Export/Import to ProVision 	Export Import to ProVision	Export and import data using ProVision on this host.					

Step			Description	
Step 6.	 number was User ID – Se Password an Handset will Host – Selec SES server. First Name a Check the Additional set of the set	ndle – Enter configured ir et to any desc ad Confirm I use to registe t the IP addre and Last Nar dd Media Se		that the Citel Gateway ame (FQDN) of the Avaya
	🗿 Add User - Microsoft Internet Exp	plorer		X
	File Edit View Favorites Tools			
	← Back		1601a 🕼 🗗 🗃 🖾 🗐	🔻 🧬 Go 🌀 Snagit 📑 🔍 🔻
	AVAYA			Integrated Management
	Help Exit		4	Server: 192.45.52.160
	Top Users List Add Search Edit Delete Password Default Profile Registered Users Conferences Media Server Extensions Emergency Contacts Hosts Media Servers Adjunct Systems Services Services Services Services Services Services Trace Logger Export/Import to ProVision	User ID Password* Confirm Password* Host* First Name*	54020 \$44020 ****** 192.45.52.160 ▼ Meridian Citel Image: Citel	

ep		Description	
	At the Add Media	a Server Extension page, configure the follo	owing:
		– Set to Phone Number field value configu	
		-	
	Media Sei	rver – Set to the media server where this OP	S station is configured.
	Click Add	and then click Continue on the next page [not shown].
			_
	Note Media Serv	er was previously configured on SES.	
	Add Media Server Extension - M		6
	File Edit View Favorites Tools		
	4= Back 🔹 🤿 🛪 🖄	Search 📠 Favorites 🐠 Media 🧭 🛃 🚽 🎒 🗹 📄	
	Address 🕘 https://192.45.52.160/imp	ress/do/edituserextension/add_extension?auth_user_id=540208sid=sid1478.cmd=Continue	💌 🔗 Go 🌀 SnagIt 🛃 🦿
	۸\/۸\/۸	ß	Integrated Management
	AVAYA		SIP Server Management
	Help Exit		Server: 192.45.52.16
	-	-	
	Top Users	Add Media Server Extension	
	List		
	Add	Add Media Server extension for user 54020.	
	Search	Extension* 54020	
		Media Server S8710 💌	
	Edit	Fields marked * are required.	
	Delete	rielus markeu are requireu.	
	Password	Add	
	Default Profile		
	Registered Users		
	Conferences		
	Media Server Extensions		
	Emergency Contacts		
	Hosts		
	Media Servers		
	Adjunct Systems		
	Services		
	Server Configuration		
	Certificate Management		
	IM Logs		
	Trace Logger		
	Export/Import to ProVision		
	Update		
	🕘 Done		📄 🕒 Internet
,	Donast Stong E	The narrow to configure additional Cital	Cotomor Handasta on Arra
	1 1	7 as necessary to configure additional Citel	Galeway Handsets on Ava
	SES.		
	1		



tep		Description	
10.	Click Continue a	t the bottom of the right panel.	
		Search 🗃 Favorites 🎯 Media 🧭 🛃 🖌 🎒 🗃 🗐	
	Address 🙆 https://192.45.52.160/imp	ess/do/listhost/updateall?mainmenu=t	💌 🖓 Go 🌘 Snaglt 🖭 👘 🔹
	AVAYA		Integrated Management SIP Server Management
	Help Exit		Server: 192.45.52.160
	Top ■ Users	Continue	
	 Conferences Media Server Extensions 	Wrote 1 domain access record.	
	Emergency Contacts Hosts Media Servers	Wrote 1 proxy configuration record. Wrote 1 proxy configuration record.	
	 Media Servers Adjunct Systems Services 	Wrote 1 proxy configuration record. Wrote 8 system parameters records.	
	 Server Configuration Certificate Management 	Update 54020 on home node 192.45.52.160.	
	IM Logs Trace Logger Export/Import to ProVision	Wrote 2 contact set records. Wrote 2 public address records. Wrote 1 contact record. Wrote 1 identity record. Wrote 1 inesence list record.	
		Wrote 1 access control list record.	
		Wrote 1 extension record.	
		Continue	1
	Done		🛛 🔓 🙋 Internet

5. Configure Citel Gateway

This section describes the steps for configuring the Citel Gateway. Citel Gateway supports a variety of TDM telephones including Nortel Meridian, Nortel Norstar, NEC and Panasonic. The configuration in this section is only for the Citel Gateway. This section assumes that the Citel Gateway's IP address is already configured. Configuration steps described in this section apply only to the fields where a value needs to be entered or modified. Default values are used for all other fields.

tep	Description
1.	Access the serial interface on the Citel Gateway using following settings for a terminal
	emulation program:
	• Speed 9600
	Data bits 8
	• Parity N
	• Stop bits 1
	Flow control Xon/Xoff
	The Setup Wizard runs automatically the first time the unit is started; or the user can select Setup Wizard on the Citel SIP Handset Gateway->Utilities screen and hit the Enter key to setup the Citel Gateway.
	SIII - HyperTerminal
	Ele Edit View Call Iransfer Help
	System File Diagnostics Upgrade ≛Setup Wizard
	Run the Setup Wizard to set a typical system configuration <f1> or <ctrl-a> for help</ctrl-a></f1>
	Mar 30 09:46:34: HSGW INFO: Got rx connection on port 2000 Mar 30 09:47:28: HSGW INFO: Starting FLASH file system optimize Mar 30 09:52:34: HSGW INFO: Finished FLASH file system optimize
	Mar 30 10:09:14: MGMT INFO: Successful login on the console
	Mar 30 10:09:14: MGMT INF0: Successful login on the console POWER PORT: 1-6 7-12 13-18 19-24 ON DDDDDD DDDDDD DDDDDD

Step	Description
2.	Only relevant setup steps are shown here. PBX type can be changed here depending upon
	what type of telephones will be connected to the Citel Gateway. In this example, Nortel
	Meridian telephone sets will be used.

	********** We need to specify the PBX type ***********

	Currently configured PBX type is 'Meridian'
	Do you want to select a new PBX type? [y/n] n
	Enable all 24 ports? [y/n] y

	* Finally, let's configure this unit's IP configuration **

	Set System IP Configuration? [y/n] y
	Use DHCP to acquire unit's IP address? [y/n] n
	Enter the IP Address: 192.45.53.101
	Enter the Subnet Mask: 255.255.255.0
	Enter the Default Router: 192.45.53.1

Step	Description
3.	At the SIP Handset Gateway screen, select Configuration \rightarrow VLAN \rightarrow LAN to
	configure the following:
	• Priority – This is layer-2 priority setting and can assume any value between 1 and
	7.
	• Id – Set to 53.
	Save the configuration by selecting Save in the left pane of the screen.
	🗞 GIII - HyperTerminal
	<u>Elle Edit View Call Iransfer Help</u>
	IP Handset Gateway->Configuration->VLAN->LAN HSGW Meridian 3.61.4d07 HSGW
	Port 1
	Log <u>*</u> State Enabled
	IP Priority [6] System Id [53]
	VLAN Save
	Enables use of VLAN
	<pre>Chables use of VLHW <f1> or <ctrl-a> for help</ctrl-a></f1></pre>
	Mar 30 09:52:34: HSGW INFO: Finished FLASH file system optimize
	Mar 30 10:09:14: MGMT INFO: Successful login on the console Mar 30 10:10:59: SYS INFO: DHCP client enabled.
	Mar 30 10:11:00: MGMT INFO: The system configuration has changed
	POWER PORT: 1-6 7-12 13-18 19-24
	ON DDDDDD DDDDDD DDDDDD DDDDDD
	Connected 0:52:45 Auto detect 9600 8-N-1 SCROLL CAPS NUM Capture Print echo
4.	Open a web browser and enter <u>http://a.b.c.d</u> for the URL, where a.b.c.d is the IP address
	of the Citel Gateway. Enter the User Name and Password. Click OK to proceed to the
	Inext screen.
	This secure Web Site (at 192.45.53.101) requires you to log on.
	Please type the User Name and Password that you use for
	Citel-Handset-Gateway.
	User Name citel
	Password ******
	Save this password in your password list
	OK Cancel

Step	Description	
5.	This is the main screen on the Citel Gateway to perform v	various configuration steps. Click
	on any of the following buttons to configure the Citel Gat	teway:
	• Dialing Rule – To configure the dial plan for the	Citel Gateway.
	• Configure Handsets – Configure the telephones	
	• Configure QoS – Configure the QoS values and t	-
	Citel SIP Handset Gateway - Microsoft Internet Explorer File Edit View Favorites Tools Help	
	→ Back + → → · ② ② ③ ④ ④ ③ Search ⓐ Favorites ⑧ Media ③ ⑤ □ → ③ ☑ 目	
	Address 🛃 http://192.45.53.101	🗾 🖓 Go 🛛 Links 🎽 🌀 SnagIt 🛃 🔹
	k	<u>A</u>
	Citel. The VolP Migration Company	Citel SIP Handset Gateway
	System Information: Dialing Rules: Read and configure system-wide information Configure Dialing Rules	Configure Handsets: Configure handset parameters
	Compare paining in a compare process and in a compare paining in a compare	compore nanose parameters
	System information	Configure Handlets
	Configure QuoS: Configure SNMP: Configure Quality of Service parameters Configure SNMP parameters	Maintenance: System maintenance pages
	Configure Qo S Configure SNIIP	Maintenance
	Userbile Loopott Page gete rated at 104801	
	Cópyrig 110 2006-2007 Cite l'Technologies Ltd. All Rights Resented.	
	Done	
6.	Select the Dialing Rule button from the screen at Step 5	and click on Add a Rule to
	generate a default rule for the dial plan.	
	Circ I	
	Citel. The VolP Migration Company	Citel SIP Handset Gateway
	Configure Dialing Rules	
	There are a second s	Control Million of
	Home Back	Setup Wizard
	Order Match Translation Rule Excluded	d Ports Actions
	No rules found	
		Add a Rule

					scription					
At the (Configu	ure Dia	ling Rules	s screen, cl	ick Edit	to edit th	ne def	ault rule	•	
Cit	el.	ne VolP Migr	ration Company					Citel SIP I	Hand	set Gatew
Config Home		aling Ru Back	lles							Setup Wiz
Order	Match	Transla	tion Rule	Excluded	d Ports	Actions				
1	(x.)	8		None		Edit		Remove	•	
										Add a Ru
configu creen i	s used a ration a n Step	for diali and then 5.	ng a five-o a click Ho i	t Match cr digit exten me on the	sion. Cli	ck Subm	it to a n [not	complete shown]	e this to g	o back to
configu creen i	s used : ration a n Step	for diali and then 5.	ng a five-o	digit exten me on the	sion. Cli	ck Subm	it to a n [not	complete shown]	e this to g	o back to
configu creen i	s used a ration a n Step el. Th aling R	for diali and then 5.	ng a five-o a click Ho i	digit exten me on the	sion. Cli	ck Subm	it to a n [not	complete shown]	e this to g	5
configu creen i Cit Edit Di	s used a ration a n Step el. Th aling R	for diali and then 5. We VolP Migr	ng a five-o a click Ho i	digit exten me on the	sion. Cli	ck Subm	it to a n [not	complete shown]	e this to g	o back to
Citto Edit Dia	s used : ration a n Step el. Th aling R	for diali and then 5. We VolP Migr	ng a five-o a click Ho ation Company (xxxxx) {} [] Port 1	digit exten me on the Port 2	sion. Cli subseque	ck Subm ent screen	it to a n [not	complete shown]	e this to g	o back to
Cito Edit Dia Rule 1 Match c	s used : ration a n Step el. The aling R riterion tion rule	for diali and then 5. We VolP Migr	ng a five-o a click Ho ation Company (xxxx) {} Port 1 Port 5 Port 9 Port 13 Port 17	digit exten me on the Port 2	sion. Cli subseque Port 3 Port 3 Port 7 Port 11 Port 15 Port 19	ck Subment screen	it to a n [not	complete shown]	e this to g	o back to

Cite	• The VolP Migration Comp	any			Citel SIP Handse	et Gatev
Configure Hand Home	isets			Swap Ports	Add Lines	Refres
Port	Name	Type Configured	Port Status	SIP Status	Configuration	Res
1	Port1	Unknown	Not Fitted	ldle	Handset Lines	
2	Port2	Unknown	Not Fitted	Idle	Handset Lines	
3	Port3	Unknown	Not Fitted	Idle	Handset Lines	
4	Port 4	Unknown	Not Fitted	ldle	<u>Handset Lines</u>	
5	Port 5	Unknown	Not Fitted	ldle	Handset Lines	
6	Port 6	Unknown	Not Fitted	Idle	Handset Lines	
7	Port 7	Unknown	Not Fitted	Idle	Handset Lines	
8	Port 8	Unknown	Not Fitted	Idle	Handset Lines	
9	Port 9	Unknown	Not Fitted	ldle	Handset Lines	
10	Port 10	Unknown	Not Fitted	Idle	Handset Lines	
11	Port 11	Unknown	Not Fitted	Idle	Handset Lines	
12	Port 12	Unknown	Not Fitted	Idle	Handset Lines	
13	Port 13	Unknown	Not Fitted	Idle	Handset Lines	
14	Port 14	Unknown	Not Fitted	Idle	Handset Lines	
15	Port 15	Unknown	Not Fitted	Idle	Handset Lines	
16	Port 16	Unknown	Not Fitted	Idle	Handset Lines	
17	Port 17	Unknown	Not Fitted	ldle	Handset Lines	
18	Port 18	Unknown	Not Fitted	Idle	Handset Lines	
19	Port 19	Unknown	Not Fitted	ldle	Handset Lines	
20	Port 20	Unknown	Not Fitted	Idle	<u>Handset Lines</u>	
21	Port 21	Unknown	Not Fitted	ldle	Handset Lines	
22	Port 22	Unknown	Not Fitted	ldle	Handset Lines	
23	Port 23	Unknown	Not Fitted	ldle	Handset Lines	
24	Port 24	Unknown	Not Fitted	Idle	<u>Handset Lines</u>	

Step	Descr	iption
10.	At the Port 1 Lines screen, click Edit Line 1	l.
	Citel. The VolP Migration Company	Citel SIP Handset Gateway
	Home Back	
	Line AoR	Delete Edit
	1 (Empty)	Edit Line 1
	Show Hidden	Reset Port Delete Checked

		Description			
At Port 1 Line 1 Configuration screen, configure the following:					
•	SIP Address-of-Record (A	AOR) – Set to SIP URI. In this example, it is set to			
		om where 54020 is User ID configured in Section 4,			
	-	\mathbf{n} is the domain name of the Avaya SES server			
	configured in Section 4, St	ep 2.			
•	Display-Name – Enter any	descriptive name.			
•	Domain – Set to the SIP D	omain field value configured in Section 4, Step 2.			
•	Server Address – Set to th Section 4, Step 2.	e IP address of the Avaya SES server configured in			
•		isation box			
	-	mary Handle field value configured in Section 4, Ste			
•	6.	mary manufe field value configured in Section 4, Step			
-		name configured in Section 4 Stop 2			
•		name configured in Section 4, Step 2.			
•		sword – Set to the Password field value configured in			
	Section 4, Step 6.				
•	•	P address of the Avaya SES server configured in			
	Section 4, Step 2.				
•	Click Submit.				
Ci	tel. The VolP Migration Company	Citel SIP Handset Gateway			
Port +	1 Line 1 Configuration				
Port Add SIP	1 Line 1 Configuration	sip:54020@devconnect.com 54020			
Port Add SIP Disp Reg	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) olay-Name istrar Server	54020			
Port Add SIP Disp Reg Don	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) olay-Name istrar Server nain	64020 devconnect.com			
Add SIP Disp Reg Don Expi 3rd	1 Line 1 Configuration I Line 1 Configuration Image: strain server istrar Server nain ration (seconds) party Registrant	54020			
Add SIP Disp Reg Don Expi 3rd (Whe Serv	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) Address-of-Record (AOR) Day-Name istrar Server Dain ration (seconds) Darty Registrant party Registrant In required) rer Address Darty Registrant	64020 devconnect.com			
Port Add SIP Disp Reg Don Expl 3rd (Whe Serv (Whe	1 Line 1 Configuration Back ressing Address-of-Record (AOR) blay-Name istrar Server nain ration (seconds) barty Registrant n required)	64020 devconnect.com 3600			
Add SIP Disp Don Expi 3rd ((Whe Sen (Whe Sen	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) Address-of-Record (AOR) Day-Name istrar Server Dain ration (seconds) Doarty Registrant on required) Pression norisation Data Server Data Server	54020 devconnect.com 3600 192.45.52.160			
Add SIP Disp Don Expi 3rd ((Whe Sen (Whe Sen	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) Address-of-Record (AOR) Day-Name istrar Server Initial Server nain ration (seconds) party Registrant Initial Server nor required) Initial Server name Server	54020 devconnect.com 3600 192.45.52.160			
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Port Add SIP Disp Reg Don Exp 3rd (<i>Whe</i> Sen (<i>Whe</i>) Sen (<i></i>	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) Address-of-Record (AOR) Day-Name istrar Server Dain nation (seconds) Doarty Registrant prer Address Dained/Directory prorisation Date Authorisation ate Authorisation Date Authorisation massword Dep Password ky Server Date Authorisation	64020 devconnect.com 3600 192.45.52.160 192.45.52.160 devconnect.com 64020 devconnect.com 000000000000000000000000000000000000			
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Port Add SIP Disp Reg Don Expl 3rd (Whe Serv (Whe Serv (Whe Serv Corr Con Con Con Con Con Con Con Con Con Con	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) olay-Name istrar Server nain ration (seconds) party Registrant in required) rer Address in required) rer Address in required) rer Address in required) pe Password xy Server rer Address ference URI Park Call URI ieve Call URI rices sage Waiting Indicator URI ree Mail Retrieval URI	64020 devconnect.com 3600 192.45.52.160 192.45.52.160 devconnect.com 64020 devconnect.com 000000000000000000000000000000000000			
Port Add SIP Disp Reg Don Expl 3rd (Whe Serv (Whe Serv (Whe Serv Corr Con Con Con Con Con Con Con Con Con Con	1 Line 1 Configuration ome Back ressing Address-of-Record (AOR) Address-of-Record (AOR) Day-Name istrar Server istrar Server nain ration (seconds) party Registrant nequired) horisation ate Authorisation ate Authorisation mame masword pe Password ky Server rer Address referencing ference URI Park Call URI cices sage Waiting Indicator URI sage Waiting Indicator URI te Mail Retrieval URI red Line URI matic Call Distribution URI	E4020 devconnect.com 3600 192.45.52.160 V 54020 devconnect.com 000000000000000000000000000000000000			

ep			De	scription					
2.	At the Update Succeeded screen, click Reset and click Home on the subsequent screen								
	[not shown] to go back to the screen on Step 5 .								
	Cite	• The VolP Migration Co	ompany			Citel SIP Hands	et Gateway		
	Update Succeeded								
	Home	Back							
	The configuration was undated successfully								
	The configuration was updated successfully. For these changes to take effect, please reset the port.								
			T of these changes to t	ane elleci, please n	eser the port.		Reset		
							Reset		
3.	Repeat	Steps 6 - 12 to	add additional lines	on the same	Citel Gates	way port or to	add		
						way port or to	auu		
	another	nandset on a di	ifferent Cifel Gatew						
4.	Select the	he Configure I	ifferent Citel Gatew Handsets button fro ire the appropriate t	m the screen	-	•	ре		
4.	Select the Configu	he Configure I	Handsets button frour the appropriate t	m the screen	-	•	ре		
4.	Select the Configure of Configu	he Configure I ured to configu ick Submit [no	Handsets button froure the appropriate to the shown].	m the screen	-	•	ре		
4.	Select the Configure of Configu	he Configure H ured to configu	Handsets button froure the appropriate to the shown].	m the screen	del on Citel	•	-		
4.	Select the Configuration of Configuratio	he Configure H ured to configu ick Submit [no el. The VolP Migrat	Handsets button froure the appropriate to the shown].	m the screen	del on Citel	l Gateway.	-		
4.	Select the Configuration of Configuratio	he Configure I ured to configu ick Submit [no	Handsets button froure the appropriate to the shown].	m the screen	del on Citel	l Gateway.	-		
4.	Select the Configuration of Configuratio	he Configure H ared to configu ick Submit [no el. The VolP Migrat ure Handsets	Handsets button froure the appropriate to the shown].	m the screen	del on Citel	l Gateway.	-		
4.	Select the Configuration of Configuratio	he Configure H ared to configu ick Submit [no el. The VolP Migrat ure Handsets	Handsets button froure the appropriate to the shown].	m the screen	del on Citel	l Gateway. Citel SIP Handset	Gateway		
4.	Select the Configuration of Configuration Configuration of Configuration o	he Configure H ured to configu ick Submit [no el. The VolP Migrat ure Handsets	Handsets button fro tre the appropriate t tot shown]. tion Company Type Configured Meridian M2616	om the screen elephone mo	del on Citel	Citel SIP Handset	Gateway		
4.	Select the Configuration of the characteristic of the characterist	he Configure H ared to configu ick Submit [no el. The VolP Migrat ure Handsets	Handsets button fro tre the appropriate t ot shown]. tion Company	om the screen elephone mo Port Status	del on Citel (Swap Ports SIP Status	Citel SIP Handset Add Lines Configuration	Gateway Refresh Reset		
4.	Select the Configuration of the classical configuration of the	he Configure H ared to configu ick Submit [no el. The VolP Migrat ure Handsets Name Port1	Handsets button fro tre the appropriate t ot shown]. tion Company Type Configured Meridian M2616 V Unknown	om the screen elephone mo Port Status Not Fitted	del on Citel Swap Ports SIP Status Idle	I Gateway.	Gateway Refresh Reset		
4.	Select the Configuration of the characteristic of the characterist	he Configure H ared to configu ick Submit [no el. The VolP Migrat ure Handsets Name Port1 Port2	Handsets button fro tre the appropriate t tot shown]. tion Company Type Configured Meridian M2616 ♥ Unknown Meridian A2008 Meridian M2006	Port Status Not Fitted Not Fitted	del on Citel Swap Ports SIP Status Idle Idle	Citel SIP Handset Add Lines Configuration Handset Lines Handset Lines	Gateway Refresh Reset		
4.	Select the Configuration of the characteristic of the characterist	he Configure H ared to configure ick Submit [no el. The VolP Migrat ure Handsets Name Port 1 Port 2 Port 3	Handsets button fro tre the appropriate to tot shown]. tion Company Type Configured Meridian M2616 ♥ Unknown Meridian M2008 Meridian M2008 Meridian M2008 Meridian M2216	Port Status Not Fitted Not Fitted Not Fitted	del on Citel Swap Ports SIP Status Idle Idle Idle Idle	Citel SIP Handset Add Lines Configuration Handset Lines Handset Lines Handset Lines	Gateway Refresh Reset		
4.	Select th Configu Then cl Cit(Configu Home Port 1 2 3 4	he Configure H ared to configu ick Submit [no el. The VolP Migrat ure Handsets Name Port 1 Port 2 Port 3 Port 4	Handsets button fro irre the appropriate to tot shown]. tion Company Type Configured Meridian M2616 ♥ Unknown Meridian M2008 Meridian M2008 Meridian M2008 Meridian M2008 Meridian M2016 Meridian M216 Meridian M3110 Meridian M3310 Meridian M3820	Port Status Not Fitted Not Fitted Not Fitted Not Fitted	del on Citel Swap Ports SIP Status Idle Idle Idle Idle Idle Idle	Citel SIP Handset	Gateway Refresh Reset		
4.	Select th Configu Then ch Cit Configu Home Port 1 2 3 4 5	he Configure H ared to configure ick Submit [no el. The VolP Migrat ure Handsets Name Port 1 Port 2 Port 3 Port 4 Port 5	Handsets button fro tre the appropriate to tot shown]. tion Company Type Configured Meridian M2616 ♥ Unknown Meridian M2008 Meridian M2008 Meridian M2008 Meridian M2016 Meridian M310 Meridian M310 Meridian M310 Meridian M3901 Meridian M3902	Port Status Not Fitted Not Fitted Not Fitted Not Fitted Not Fitted Not Fitted	del on Citel Swap Ports SIP Status Idle Idle Idle Idle Idle Idle Idle	Citel SIP Handset Add Lines Configuration Handset Lines Handset Lines Handset Lines Handset Lines Handset Lines Handset Lines	Gateway Refresh Reset		
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additional features required and click Submit .					
Citel. The VolP Migration Company	Unassigned	Citel SIP Handset			
Port 1 Handset Configuration	ACD Login/Logout ACD (Un-)/Available ACD Login ACD Logout	_			
Home Back	ACD Available ACD Unavailable	Сору			
Name	Auto Answer BLF				
Port Name	Conference Conference/Transfer				
Default Display	Do Not Disturb Flash				
Enter default display (%T=time/date, %P=port, %N=new line, %[n]U=user [n=optional precision	n], %D=domain Headset				
Features	Hold Message Waiting				
Audible Indication Of Message Waiting	Mic On/Off Hook Park/Retrieve				
Key Mappings M2616	Pickup Redial				
	Release				
Key+Indicator 14	Transfer				
Unassigned	Analog Pool Analog FXO 1				
Key+Indicator 13	Analog FXO 2				
Unassigned	Unassigned V				
Key+Indicator 12 Unassigned	Key+Indicator 5				
Key+Indicator 11	Key+Indicator 4				
Unassigned 💌	Line-2				
Key+Indicator 10 Unassigned	Key+Indicator 3 Line-1: 60032				
Key+Indicator 9 Unassigned	Key+Indicator 2 Line-1: 60032				
Key+Indicator 8 Unassigned	Key+Indicator 1 Handsfree/Mute				

)		Description					
•	Select the Configure QoS button from the screen at Step 5 and configure the following						
	fields:						
	• RTP IP TOS Byte –	Set this field for the Layer-3 Diffserv value to be used. In					
	this example, a Hex value of b8 was used, which calculates to ASCII 46 after low order bit shifting.						
	e	- Set the priority for each Codec value to be used.					
	 Click Submit. 						
	Citel. The VolP Migration Cor						
	CICE • The VolP Migration Cor	npany Citel SIP Handset Gateway					
	IP Type of Service						
	RTP IP TOS Byte	b8 (hex)					
	Silence Suppression						
	Voice Activity Detection	Off					
	Codec Preferences						
	G.711u	1 (Highest priority) 🖌					
	G.711a	Off 🖌					
	G.729	2					
		Reset Form Submit					

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment using the Citel Gateway and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, transfer and conference. Verified Citel Gateway interactions with SES, Avaya Communication Manager, and Avaya SIP, H.323, and digital telephones.

6.1. General Test Approach

The general test approach was to place calls to and from the Citel Gateway Handsets and exercise basic telephone operations. The main objectives were as follows:

- The Citel Gateway successfully registers with Avaya SES.
- Successfully establish calls between the Citel Gateway and Avaya SIP, H.323, and digital telephones attached to Avaya SES or Avaya Communication Manager.
- Successfully establish calls between the Citel Gateway and PSTN telephone through Avaya Communication Manager.
- The Citel Gateway successfully handles concurrent calls on its two lines.
- The Citel Gateway successfully negotiates the right codec.
- The Citel Gateway successfully shuffles for VoIP calls.
- The Citel Gateway successfully transmits DTMF during a call.
- The Citel Gateway successfully holds a call, transfers a call, establishes a three party conference call, and displays calling party number.
- The Citel Gateway successfully handles layer-3 (DiffServ) QoS for Audio.

For serviceability testing, failures such as cable pulls and hardware resets were applied. For performance testing, a conference call involving two Citel Gateways and two Avaya telephones was formed as follows. A call was established between an Avaya telephone and a Citel Gateway. The Citel Gateway then established a call with another Citel Gateway, and bridged the two calls together, forming a 3-party conference. The second Citel Gateway then established a call with another Avaya telephone, and bridged the two calls together, effectively forming a 4-party conference.

6.2. Test Results

The test objectives of **Section 6.1** were verified. For serviceability testing, the Citel Gateway operated properly after recovering from failures such as cable disconnects, and resets of the Citel Gateway, the Avaya SES server, and Avaya Communication Manager. For performance testing, the conference call was successfully maintained for approximately two hours. Citel Gateway successfully shuffles to communicate directly with the other endpoints.

The following observations were made during testing:

- Citel Gateway cannot negotiate with Avaya Communication Manager for the correct codec when establishing a conference call if the codec used during negotiation is not the top priority codec for the Citel Gateway. After the conference setup, the other two parties cannot hear audio from the last party added.
- Citel Gateway cannot negotiate with Avaya Communication Manager for the correct codec when putting a held call off hold. This happens if the codec used during negotiation is not the top priority codec for the Citel Gateway and hence no audio for the call when the call is put off hold.

Citel will address and resolve all the above observations with future firmware releases. Contact Citel (<u>www.citel.com</u>) for further updates.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the Citel Gateway Handsets successfully register with the Avaya SES server by following the **Users -> Registered Users** links on the SES Administration Web Interface.
- Place calls to and from the Citel Gateway Handset and verify that the calls are successfully established with two-way talk path.
- From the Avaya Communication Manager System Access Terminal (SAT) interface, perform the following steps to verify:
 - Audio codec used between two telephones
 - Shuffling between two telephones

Step	Description						
1.	Enter status trunk <t> command, where t is the SIP trunk configured in Section 3.6. Note the Member with Service State set to in-service/active. In this example, 0010/002 and 0010/006 are active and either member can be used to verify whether calls shuffled and which codec was used.</t>						
	Status trunk 10						
			TRUNK GRC	UP ST	ATUS		
	Member	Port	Service State	Mtce Busy	Connected Ports		
	0010/002 0010/003 0010/004 0010/005 0010/006 0010/007 0010/008	T00047 T00048 T00050 T00051 T00052 T00053 T00054	<pre>in-service/idle in-service/active in-service/idle in-service/idle in-service/active in-service/idle in-service/idle in-service/idle in-service/idle in-service/idle</pre>	no no no no no no no no no	T0051 T0047		

Step	Description				
2.	 Description Enter status trunk <m>, where m is the member in active state as noted in the previous step for verification of codec used and shuffling status:</m> Codec – The codec used for Audio is G.711MU in this example. Shuffling - If the Near-end IP Addr and Far-end IP Addr for Audio are using the same port and the Audio Connection Type is ip-direct, it signifies that shuffling was successful. In this example, shuffling was successful. 				
	status trunk 10/2 Page 1 of 2				
	TRUNK STATUS				
	Trunk Group/Member: 0010/002 Port: T00047 Signalling Group ID: Signalling Group ID: Signalling Group ID: Service State: in-service/active Maintenance Busy? No				
	Connected Ports: T0051				
	Port Near-end IP Addr Port Far-end IP Addr Port Signaling: 01A0617 192. 45.100.147 : 5061 192. 45.52.160 : 5061				
	G.711MU Audio: 192. 45. 53.101 : 34008 192. 45. 53.102 : 34008 Video: Video Codec: Authentication Type: None				
	Audio Connection Type: ip-direct				

8. Support

For technical support on Citel Gateway and how to configure TDM telephones connected to it, consult the support pages at <u>http://www.citel.com/Support/</u> or contact Citel technical support at:

- Telephone: 1 (877)248-3587
- E-mail: support@citel.com

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services 3.1.1, and digital telephones connected to Citel Gateway. Citel Gateway is a SIP-based VoIP appliance. During compliance testing, digital telephones connected to Citel Gateway successfully registered with Avaya SIP Enablement Services, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as three-way conference, transfers, hold, etc.

10. Additional References

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

[1] Administrator Guide for Avaya Communication Manager, Issue 2.1, May 2006, Document Number 03-300509

[2] Administration for Network Connectivity for Avaya Communication Manager, Issue 11, February 2006, Document Number 555-233-504

[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 6, February 2006, Document Number 555-245-206

[4] *Installing and Administering SIP Enablement Services R3.1.1*, Issue 2.0, August 2006, Document Number 03-600768

Product documentation for Citel products may be found at <u>http://www.citel.com</u>. [5] Citel SIP Gateway install manual G-SIP3D-RUC

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