

Integration

Title	Avaya ACM/SES SIP
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DuVoice Versions	5.10, 5.00
Switch Versions	

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Features Supported

Message Lights	Yes
Forward to personal greeting RNA	Yes

Forward to personal greeting Busy	Yes
Forward to personal greeting DND	
Different busy greeting	Yes
Auto login to a mailbox	Yes
Trunk ID multiple tenant	
Voicemail transfer	
Record a call	
DID Fax	No
Caller ID with message	Yes
Trunk to trunk transfer (unsupervised)	
Loop start disconnect	No
Tone disconnect	No
SIP disconnect	Yes
Supervised transfers	No
Blind transfers	Yes
Call screening	No
Caller queuing	No

Hospitality Features Supported

Property management integration (PMS)	
Room phone control	
Wakeup calls	Yes
Failed wakeup alerting	
Guest name changes	
Room Clean - Room Dirty Status	

PBX Requirements and Programming

Requirements

- Avaya ACM CM v5 or above.
- Avaya SIP Enablement Services
- Avaya Communications Manager

Known Issues

None.

Programming

This section describes the procedure for setting up a SIP trunk between Avaya Communication manager and Avaya SES.

Capacity Verification

Enter the **display system-parameters customer-option** command. Verify the number of OPS stations for the voicemail system. You will need one for each line on the DuVoice system.

UPIIUMHL FEHIUKES	
G3 Version: V15 Software Pac	kage: Standard
Location: 1 RFA System ID (SID): 1
Platform: 18 RFA Module ID (MID): 1
	USED
Platform Maximum Ports:2850	46
Maximum Stations: 14	10
Maximum XMOBILE Stations: 0	0
Maximum Off-PBX Telephones - EC500: 0	0
Maximum Off-PBX Telephones - OPS: 100	2
Maximum Off-PBX Telephones - PBFMC: 0	0
Maximum Off-PBX Telephones - PVFMC: 0	0
Maximum Off-PBX Telephones - SCCAN: 0	0

On tab 2 verify the number of SIP trunks available.

OPTIONAL FEATURES	
IP PORT CAPACITIES	USED
Maximum Administered H.323 Trunks: 10	3
Maximum Concurrently Registered IP Stations: 12	0
Maximum Administered Remote Office Trunks: 0	0
Maximum Concurrently Registered Remote Office Stations: 0	0
Maximum Concurrently Registered IP eCons: 0	0
Max Concur Registered Unauthenticated H.323 Stations: Ø	0
Maximum Video Capable H.323 Stations: 0	0
Maximum Video Capable IP Softphones: 0	0
Maximum Administered SIP Trunks: 10	10
Maximum Administered Ad-hoc Video Conferencing Ports: 0	0
Maximum Number of DS1 Boards with Echo Cancellation: 0	0
Maximum TN2501 VAL Boards: 10	0
Maximum Media Gateway VAL Sources: 1	1
Maximum TN2602 Boards with 80 VoIP Channels: 0	0
Maximum TN2602 Boards with 320 VoIP Channels: 0	0
Maximum Number of Expanded Meet-me Conference Ports: 0	0

IP Codec

Enter **change ip-codec-set <c>** command, where c is a number between 1 and 7, inclusive.

		IP	Codec Set	
Codec	Set: 7			
Audio		Silence	Frames	Packet
Codec		Suppression	Per Pkt	Size(ms)
1: G.711	MU	n	2	20
2:				
3:				
4:				
5:				
6:				
7:				
Medi	a Encry	ption		
1: none				
2:				
3:				

Configure IP Network Region

Enter **change ip-network-region <n>** command, where **n** is a number between 1 and 250 inclusive, and configure the following.

- Authoritative Domain This domain must match the SIP Domain value configured in Avaya SES.
- Intra-region IP-IP Direct Audio Default value for this field is yes.
- Inter-region IP-IP Direct Audio
 Default value for this field is ve
- Default value for this field is yes.
- Codec Set

Set this to the value of the codec set selected under **IP Codec**.

IP NETWORK REGION						
Region: 1						
Location: Authoritative	Authoritative Domain: duvoice.lan					
Name:						
MEDIA PARAMETERS	Intra-region IP-IP Direct Audio: yes					
Codec Set: 1	Inter-region IP-IP Direct Audio: yes					
UDP Port Min: 2048	IP Audio Hairpinning? n					
UDP Port Max: 3329						
DIFFSERU/TOS PARAMETERS	RTCP Reporting Enabled? 😈					
Call Control PHB Value: 46	RTCP MONITOR SERVER PARAMETERS					
Audio PHB Value: 46	Use Default Server Parameters? 😈					
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): 2	0					
Keep-Alive Interval (sec): 5						
Keep-Alive Count: 5						

Configure IP Node Name

Enter change node-names ip. Add a node name for Avaya SES along with its IP address. IP NODE NAMES

Name	IP Address
SES	192.168.11.70
default	0.0.0
duvoice	192.168.11.26
msqserver	192.168.11.62
procr	192.168.11.70

Configure SIP Signaling

Enter **add signaling-group <s>** command, where **s** is an available signaling group and configure the following.

- Group Type Set to **sip**.
- Near-end Node Name Set to procr.

- Far-end Node Name Set to the Avaya SES name or procr if hosted on the same board.
- Far-end Network Region Set to the region configured under **IP Network Region**.
 Far-end Domain
- Fai-end Domain Set to the value entered in **Authoritative Domain**.

310///11	na anour
Group Number: 101 Group Type Transport Method IMS Enabled? n Co-Resident SES	e: sip d: <u>tls</u> S? y
Now-ood Nodo Namos Succes	Fax-and Node Names Susan
Near-end Node Name: procr	Far-end Node Name: procr
Near-end Listen Port: 6001	Far-end Listen Port: 5061
	Far-end Network Region: 1
Far-end Domain: duvoice.lan	
Incoming Dialog Loophacks, aliginate	Bypass If IP Threshold Exceeded? n
Incoming bialog Loopbacks: eliminate	KFC 3389 CONFORT NUISE? N
DIME over IP: rtp-payload	Direct IP-IP Audio Connections? U
Session Establishment Timer(min): 3	IP Audio Hairpinning? n
Enable Layer 3 Test? n	Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 🚹

Configure SIP Trunk

Enter

Enter **add trunk-group** <**t**> command, where **t** is an unallocated trunk group and configure the following.

- Group Type Set the Group Type field to **sip**.
- Group Name Enter a descriptive name.
- TAC (Trunk Access Code)
 - Set to any available trunk access code.
- Signaling Group Set to the Group Number field value to the signaling group.
- Number of Members
 Allowed value is between 0 and 255. Set this value to twice the number of voice ports
 on the DuVoice.

 Note: Each SIP call between two SIP endpoints requires two SIP trunks for the
 duration of the call.
- Service Type This must be set to public-ntwrk to support proper integration.

		TRUNK GRO	UP		
Group Number:	11	Group	Type:	sip CD	R Reports: U
Group Name:	DuVoice SIP		COR: 1	1 TN: 1	TAC: *211
Direction:	two-way	Outgoing Dis	splay? 🛛	<u>ו</u>	
Dial Access?	n			Night Service	:
Queue Length:	0				
Service Type:	public-ntwrk	Auth	Code?	ח	
				Signalin Number of	ng Group: 101 Hembers: 10

Enable Divesion Header

This is required for proper integration. This is configured on tab 4. **PROTOCOL VARIATIONS**

Mark Users as Phone? n Prepend '+' to Calling Number? n Send Transferring Party Information? n	
Send Diversion Header? y Support Request History? y Telephone Event Payload Type:	

Configure SIP Endpoint

Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. Create a station for each line of the voicemail.

• Type

Set to 9600SIP.

- Name
- Enter a descriptive name for this station.
- On tab 6 set the SIP Trunk to the trunk number created above.

	STATION	
Extension: 2380	Lock Messages? n	BCC: 0
Port: IP	Coverage Path 1:	COR: 1
Name: 2380 Voicemail	line 1 Coverage Path 2: Hunt-to Station:	COS: 1
STATION OPTIONS	Time of Day Lock Table	
Loss Group:	19 Personalized Ringing Pattern:	1
Speakerphone:	Message Lamp Ext: 2-way Mute Button Enabled?	2380
Display Language:	english Expansion Module?	n
Survivable GK Node Name: Survivable COR:	internal Media Complex Ext:	
Survivable Trunk Dest?	U IP SoftPhone?	n

Enter **add off-pbx-telephone station-mapping** command and configure the following. Perform this for each voicemail port configured above.

- Station Extension Set the extension of the OPS station as configured above.
- Application Set to **OPS**.
- Phone Number
 Enter the number the line will use for registration and call termination.
- Trunk Selection
 Set to the trunk group number configured in for your SIP trunk.
 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
2380	OPS	<u> </u>		2380	11	1	

Configure LWC MWI

Enter change feature-access-codes command and enter a feature access code for LWC on/ off. Check your dialplan and other feature codes for an available code.

Note: You cannot use a pound sign for any feature code dialed by DuVoice system.

chan	ge feat	ure-ac	cess-c	odes		send	(return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)	
1	2	3	4	5	6	7								
	Limi	it No Ma	Lea umbe eet- Mess	.eav ive ro Ma me age	e Wo Word f Co lici Conf Seq	rd C Cal ncur ous eren uenc	allin ling rent Call ce Ac ce Tra	FEATURE g Send A Cancel A Calls Act Trace Act cess Code ce (MST)	ACCESS COU Message: [Message:] tivation: [tivation:] c Change: [Disable:]	DE (FAC) 041 042 De De	activation activation	n: n:		

Configure Avaya SIP Enabledment Services

Configuration of the Avaya SES must be performed via Internet Explorer only and will not operate correctly on any other browsers.

Login to your SES via Internet Explorer and set the System Properties

- SIP Domain
 - Enter the domain configured for your IP Network Region.
- Enter the SIP License Host.



Help Exit

Integrated Management

SIP Server Management

This Server: [1]

Top Users	View System Properties			
Address Map Priorities Adjunct Systems Aggregator Conferences	SES Version System Configuration Host Type	SES-5.2.1.0-016.1 Simplex CM combined home-edge		
Emergency Contacts	SIP Domain*	#uvoice.lan		
 Export Import to Provision Hosts IM logs Communication Manager Servers Communication Manager Extensions Server Configuration 	Note that the DNS domain is unknown If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com			
Admin Setup	SIP License Host*	192.168.11.70		
IM Log Settings License	DiffServ/TOS Parameter	5		
System Properties	Call Control PHB Value*	46		
SIP Phone Settings	802.1 Parameters			
Survivable Call Processors	Priority Value*	6		
System Status Trace Logger	Management System Access Login			
Trusted Hosts	Management System Access Password			
	DB Log Level disabled			
	Update			

Add matching user to the SES for the extensions created on the ACM.

- Primary Handle
- Set this to the extension number.
- User ID

Set this to the extension number. This field will be used as the Username for the voicemail port in System Configuration.

- Password Enter a password. This field will be used as the Password for the voicemail port in System Configuration.
- First Name/Second Name Enter a descriptive first and last name.
- Add Communication Manager Extension

Be sure to check this box. This will create a mapping for this extension between the SES and ACM.

When the Add button is pressed enter the extension number and confirm the Communication Manager Server field is correct and click Add.

AVAYA	In	tegrated Management SIP Server Management
Help Exit		This Server: [1]
Top Users	Add User	
Add Dofoult Brofile	Primary Handle*	2380
Delete	User ID	2380
Edit	Password*	•••••
List	Confirm Password*	•••••
Password	Host*	192.168.11.70 -
Search	First Name*	Voicemail
Manage All Registered Users	Last Name*	Port 1
Search Registered	Address 1	
Devices Search Registered	Address 2	
Users	Office	
Address Map Priorities	City	
Adjunct Systems	State	
Conferences	Country	
Emergency Contacts	Zip	
 Export/Import to ProVision Hosts 	Survivable Call Processor	none 🔻
IM logs Communication Manager Servers List	Add Communication Manager Extension Fields marked * are	required.
Communication Manager	Add	

Configure Voicemail

This document assumes the voicemail software has already been installed.

Configure Ports

Each voicemail port must be configured with it's corresponding User in the SES.

Start System Configuration

Double click the first configured port.

- Extension number Enter the extension number for this port.
- Hunt group extension is a member of Enter the hunt group number for the voicemail system

ice Port Number	1	_ ×
Port Configuration	SIP Configuration SIP	
Account Name	2380	
Password	123456	
Domain	duvoice.lan	
Server Port	5060	
User agent	DuVoice	
	ОК	Cancel <u>A</u> pply Help

Click the **SIP Configuration** tab.

- Account Name
 - Enter the user name for this port entered in the SES.
- Password
- Enter the password entered for this user in the SES.
- Domain Enter the domain name the SES is configured for.
- Server Port
 - The default is 5060.
- User agent
 - The default value is DuVoice.

Voice Port Number	1
Port Configuration	SIP Configuration SIP
Account Name	2380
Password	123456
Domain	duvoice.lan
Server Port	5060
User agent	DuVoice
	OK Cancel <u>Apply</u> Help

Click the **SIP** tab.

This tab is a duplicate of the SIP tab located under Features|Connectors. The settings on this tab are not port specific, but rather Location specific.

- Location
 - If this port is located on a different location than the default choose the location.
- Registrar address
 Enter the address of the SES server.
- Extensions address Confirm this is the address the system will use to communicate with the switch on.

Voice Port Number 1		X
Port Configuration SI	IP Configuration SIP	
Location	Default Location	
Registrar address	192.168.11.70	
Extensions address	192.168.11.25	
	OK Cancel <u>A</u> pply H	elp

Click Ok and perform these actions on all the remaining ports.