

Avaya Solution & Interoperability Test Lab

# Application Notes for Configuring the Tango Networks Abrazo Solution with an Avaya Telephony Infrastructure - Issue 1.0

# Abstract

These Application Notes describe a compliance-tested configuration comprised of the Tango Networks Abrazo Solution connected to an Avaya telephony infrastructure. The Abrazo solution extends enterprise PBX functionality to mobile devices, allowing end users to be accessible when out of the office. The Abrazo solution integrates mobile devices with existing Private Branch Exchanges (PBXs) so that the PBX sees the mobile device as another desk phone. This allows the existing PBX feature set to be applied consistently across both devices. Mobile specific functionality is then layered on top.

Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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# 1. Introduction

These Application Notes describe a compliance-tested configuration comprised of the Tango Networks Abrazo Solution connected to an Avaya telephony infrastructure, including Avaya Communication Manager and Avaya SIP Enablement Services.

Tango Networks' Abrazo Solution is a fixed mobile convergence (FMC) solution that employs solution components in both the enterprise network and the mobile operator network in order to seamlessly extend the corporate PBX features to the mobile phone. This unparalleled level of convergence allows mobile phones to offer the same productivity features as a conventional enterprise desk phone.

## 1.1. Background

The Tango Networks Abrazo Solution includes the Abrazo-C and the Abrazo-E components. As shown in **Figure 1**, the Abrazo-C communicates with the mobile operator network using standard protocols and always resides in the mobile operator's network or a hosting center. The Abrazo-E communicates with the enterprise network components including the PBX, voice mail systems, and corporate databases via standard interfaces to extend the enterprise network functionality transparently to the mobile network.

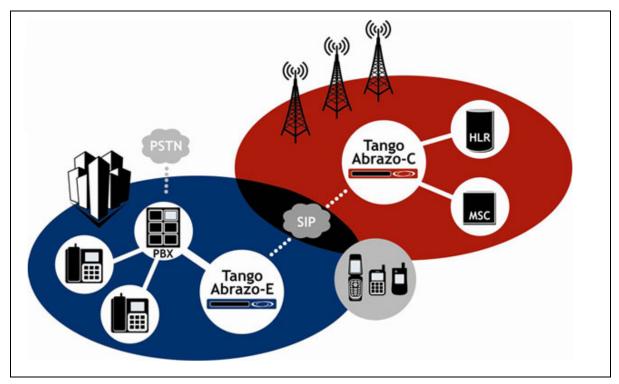


Figure 1: Tango Networks' Architecture Diagram

The Abrazo solution interacts with the Avaya Communication Manager (ACM) via Avaya SIP Enablement Services (SES) as shown in **Figure 2**.

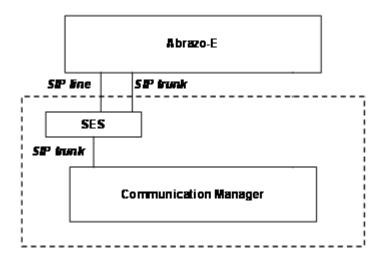


Figure 2: Abrazo-E Interfaces to the Avaya Telephony Infrastructure

The Abrazo solution uses a combination of SIP lines and trunks to integrate with Avaya Communication Manager. SIP lines are used so that Abrazo-controlled mobile devices appear as standard SIP phones and therefore benefit from the common set of PBX services offered to such devices. SIP trunks are used when the Abrazo solution must terminate a call via the *Public Switched Telephone Network* (PSTN).

#### Mobile Originations

The Abrazo solution captures all mobile originations from a user's mobile device and redirects them into the enterprise. This allows calls made from a mobile device to receive the same originating services (e.g., Abbreviated Dialing, Class of Service, Accounting, etc.) as a desk phone. To do this, the Abrazo solution redirects the call in the wireless carrier network to a *Pilot Directory Number* (PDN) (or set of DNs). This Pilot DN is owned by the enterprise (i.e., the PSTN will route calls to it into the enterprise) and must be provisioned to route to Avaya Communication Manager. Within Avaya Communication Manager, telephony translations are created that then route all calls to the Pilot DN to the Abrazo solution.

When the Abrazo solution receives calls to a Pilot DN, it replaces the Pilot DN with the original dialed digits for the call and changes the *Calling Line ID* (CLID) from the user's mobile number to the user's enterprise number. The call is then routed back to Avaya Communication Manager. Setting the CLID in the P-Asserted-Identity header ensures that Avaya SIP Enablement Services recognizes the user and applies originating services (i.e., treats this as a SIP line call).

#### Mobile Terminations

To receive calls made to a subscriber, the Abrazo solution registers a route pointing to an Abrazo Session Conductor by sending a SIP register message. This is done on a per subscriber basis when subscribers are added to the Abrazo system. Once a subscriber is registered with the Avaya, all calls made to the subscriber fork to the Abrazo solution simultaneously whenever the Avaya alerts other client devices, such as the subscriber's desk phone. The Abrazo, upon receipt of this forked leg of the call, retrieves the temporary roaming number of the subscriber's mobile device from the wireless network and re-routes the call back to the Avaya addressed to the retrieved number.

To prevent the Avaya from recognizing the originating subscriber and erroneously applying originating services again, the P-Asserted-ID header is removed from the SIP Invite message. This causes the Avaya to treat this request as a trunk based origination and to route the call to the PSTN without providing any subscriber services. By leaving the user portion of the originating subscriber information intact, CLID information is preserved.

# **1.2. Solution Configuration**

These application notes describe a solution for integrating the Tango Abrazo-E with the Avaya Product Portfolio. **Figure 3** illustrates the configuration used in these application notes. The diagram indicates the logical signaling connections between the Tango Abrazo and Avaya products. The solution described herein is also extensible to other Avaya Servers and Media Gateways.

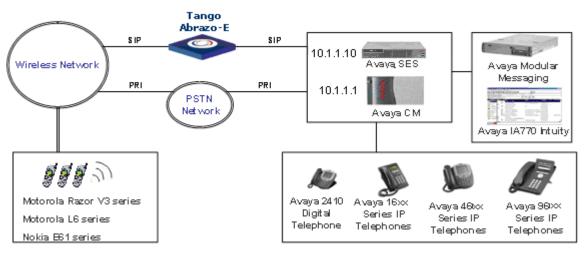


Figure 3: Interoperability Configuration Diagram

# **1.3. Equipment and Software Validated**

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

Equipment	Software/Firmware					
Avaya PBX Products						
Avaya S8300 Server running Avaya	Avaya Communication Manager 5.1 -					
Communication Manager	R015x.01.0.414.3					
Avaya G700 Media Gateway with	26.31.0					
MM712 DCP Media Module 8	FW 008					
Avaya SIP Enabled Services (SES) Server	SES-5.1.0.0-414.3f					
Avaya Messaging (Voice .	Mai)l Products					
Avaya IA 770 INTUITY	5.1					
Avaya Modular Messaging Server	4.0					
Avaya Telephon	y Sets					
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone SIP 2.0.5					
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone Edition 2.0					
Avaya 1600 Series IP Telephones	1.0.3					
Avaya 4600 Series IP Telephones	SIP (2.2) H.323 (2.9)					
Avaya 2410 Digital Telephone	4.0					
Tango Abrazo Pr	oducts					
Tango Networks Abrazo-Enterprise Release	4.0					
Tango Networks Abrazo-Carrier Release	4.0					
Mobile Devic	es					
Nokia	E61 Series					
Motorola	L6 Series					
Motorola Razor	V3 Series					

The configuration tested utilized Avaya Communication Manager running on the Avaya S8300 server; however the solution described in this document is also extensible to other Avaya servers and media gateways.

In addition, the configuration tested interfaced with a GSM wireless network and utilized GSM mobile devices; however, the solution described in this document is also extensible to IS-41 and IMS based networks. Any mobile device may be supported with the Tango Abrazo solution.

# 2. Configure Avaya Communication Manager

Basic configuration of Avaya Communication Manager and Avaya SES are beyond the scope of these Application Notes. See **Section 8** for Avaya documentation references. It is assumed that the reader has a basic understanding of the administration of Avaya Communication Manager and has access to the System Access Terminal (SAT).

This section describes the steps required for Avaya Communication Manager to support the configuration in **Figure 3: Interoperability Configuration Diagram**. The following pages provide step-by-step instructions on how to administer the required configuration parameters. The steps are performed from the SAT interface.

# 2.1. System Parameters Customer Options

The steps in this section verify that there are a sufficient number of SIP trunks and stations between Avaya Communication Manager and Avaya SES.

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

Step	Description				
2.1.1.	Issue the command <b>display system-parameters customer-o</b> active licensed features. Go to <b>Page 1</b> to ensure that the <b>Max</b> - <b>OPS:</b> value is equal to or greater than the number of endpo configuration.	imun	n Off-PBX	Telephon	nes
	display system-parameters customer-options OPTIONAL FEATURES		Page	<b>1</b> of 1	.0
	-		(SID): 1 (MID): 1		
	Platform Maximum Ports Maximum Stations Maximum XMOBILE Stations Maximum Off-PBX Telephones - EC500 Maximum Off-PBX Telephones - OPS Maximum Off-PBX Telephones - SCCAN	: 450 0 : 100 : <b>100</b>	<b>USED</b> 80 29 0 0 <b>21</b> 0		
2.1.2.	On <b>Page 2</b> verify that the <b>Maximum Administered SIP trun</b> is sufficient.	nks si	upported by	the system	m
	display system-parameters customer-options OPTIONAL FEATURES	Pag	ge 2 of 1	.0	
	IP PORT CAPACITIES Maximum Administered H.323 Trunks: 450 Maximum Concurrently Registered IP Stations: 450 Maximum Administered Remote Office Trunks: 0 Maximum Concurrently Registered IP eCons: 0 Maximum Concurrently Registered IP eCons: 0 Max Concur Registered Unauthenticated H.323 Stations: 40 Maximum Video Capable Stations: 40 Maximum Video Capable IP Softphones: 40 Maximum Administered SIP Trunks: 100 Maximum Administered Ad-hoc Video Conferencing Ports: 0 Maximum Number of DS1 Boards with Echo Cancellation: 30 Maximum TN2501 VAL Boards: 0 Maximum TN2602 Boards with 80 VoIP Channels: 0 Maximum TN2602 Boards with 320 VOIP Channels: 0 Maximum Number of Expanded Meet-me Conference Ports: 300	USED 50 4 0 0 0 0 0 20 0 0 0 0 0 0 0 0 0 0 0 0			
	(NOTE: You must logoff & login to effect the permission	on cha	anges.)		

## 2.2. IP Codec Set

This section describes the steps for administering the 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			Des	scription				
2.2.1.	inclusive, and e	nter "G.711MU" etwork Region fo	for Audi	o Codec. Thi	is a number between is IP codec set will idecs may be used v	be se	elected	
	change ip-code	c-set 1			Pag	ge	1 of	2
		IP	Codec Set					
	Codec Set:	1						
	Audio Codec 1: G.711MU 2:	Silence Suppression <b>n</b>		Packet Size(ms) 20				

# 2.3. IP Network Region

This section describes the steps for administering the IP Network Region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

Step	Description
	Enter the <b>change ip-network-region h</b> command, where "h" is a number between 1 and
2.3.1.	250, inclusive. On Page 1 of the ip-network-region form, set Codec Set to the number
	of the IP codec set configured in <b>Step 2.2.1</b> .
	change ip-network-region 1 Page 1 of 19
	IP NETWORK REGION
	Region: 1
	Location: 1 Authoritative Domain: dev4.com
	Name: 1
	MEDIA PARAMETERSIntra-region IP-IP Direct Audio: yesCodec Set: 1Inter-region IP-IP Direct Audio: yes
	UDP Port Min: 2048 IP Audio Hairpinning? n
	UDP Port Max: 3329
	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

### 2.4. IP Node Names

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

Step		Description	
2.4.1.	U	of the <b>change node-names ip</b> IP address of the SES,	
	change node-name	es ip	Page 1 of 2
	-	IP NODE NAMES	_
	Name	IP Address	
	50SES	50.1.1.50	
	AES-DevCon2	192.45.100.153	
	G250-314	10.10.200.10	
	G350	50.1.1.10	
	SES	10.1.1.10	
	default	0.0.0	
	mm	10.1.1.45	
	msgserver	10.1.1.20	
	procr	10.1.1.1	

## 2.5. Trunks and Signaling Groups for Avaya SES

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

This SIP trunk will carry the SIP signaling sent to the Abrazo-E for mobile originated and terminated calls. This SIP trunk will also provide the trunking for calls originated by the Abrazo-E when acting as a SIP endpoint to support intelligent call delivery.

Step			Description		
2.5.1.	number. On Pa • Group Type • Group Name	nge 1 of the – set to "si e – enter a r a Trunk Ac	neaningful name/descripticess Code that is valid und	gure on.	e the following:
	add trunk-grou 21	up 1	TRUNK GROUP		Page 1 of
	Group Number:	1	Group Type:	sip	D CDR
	Reports: y Group Name: TAC: *001	TO SES	COR:	1	TN: 1
	Direction: Dial Access?	n	Outgoing Display?		Night Service:
	Queue Length: Service Type:		Auth Code?	n	
	0				Signaling Group: Number of Members:

Step	Description
2.5.2.	Enter the <b>add signaling group j</b> command, where " <b>j</b> " is an available signaling group number. On <b>Page 1</b> of the <b>signaling-group</b> form, configure the following:
	<ul> <li>Group Type – set to "sip".</li> <li>Transport Method – set to "tcp".</li> <li>Near-end Node Name – enter the node name of a local C-LAN board, or "procr" if the local node is an Avaya S8300 Server.</li> <li>Near-end Listen Port – specify the local listen port, typically "5060".</li> <li>Far-end Node Name – enter the node name of the SES configured in Step 2.4.1.</li> <li>Far-end Listen Port – specify the local listen port, typically "5060".</li> <li>Far-end Listen Port – specify the local listen port, typically "5060".</li> <li>Far-end Listen Port – specify the local listen port, typically "5060".</li> <li>Far-end Domain – "dev4.com"</li> <li>Far-end Network Region – enter the IP network region configured in Step 2.3.1.</li> <li>DTMF over IP – set to "rtp-payload".</li> <li>Direct IP-IP Audio Connections – set to "y".</li> </ul>
	add signaling-group 1 Page 1 of 1 SIGNALING GROUP
	Group Number: 1 Group Type: sip Transport Method: tcp
	IP Video? n
	Near-end Node Name: procr Near-end Listen Port: 5060 Far-end Listen Port: 5060 Far-end Network Region: 1 Far-end Domain: dev4.com
	Bypass If IP Threshold Exceeded? n
	DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y Enable Layer 3 Test? n Session Establishment Timer(min): 120

Step		Description							
2.5.3.	Enter the <b>change trunk-group i</b> command, where "i" is the number of the trunk group configured in <b>Step 2.5.1</b> . On <b>Page 1</b> of the <b>trunk-group</b> form, configure the following:								
	<ul> <li>Signaling Group – enter</li> <li>2.5.2.</li> <li>Number of Members – s</li> </ul>		nber t	hat was used in <b>Step</b>					
	change trunk-group 1 1 of 21	TRUNK GROUP		Page					
	Group Number: 1 Reports: y	Group Type:	sip	CDR					
	Group Name: TO SES TAC: *001	COR :	1	TN: 1					
	Direction: two-way Dial Access? n Oueue Length: 0	Outgoing Display?		light Service:					
	Service Type: tie	Auth Code?	n						
				Signaling					
	Group: 1			2 2					
	Members: 24			Number of					

### 2.6. Dial Plan – AAR and Route Pattern

This section describes the steps for setting the Dialplan, AAR digit analysis and Route Pattern in Avaya Communication Manager for proper routing of calls from Avaya Communication Manager to Avaya SES. These calls are ultimately destined for the Tango Abrazo-E.

If the connectivity between the wireless carrier and the enterprise is via VoIP, then no incremental dial plan modifications are required on Avaya Communication Manager to route inbound calls to the Tango Abrazo.

If the connectivity between the wireless carrier and the enterprise is via the PSTN, then dial plan and route patterns must be configured on Avaya Communication Manager for both pilot directory numbers and service pilot pools as described in the steps below.

Step	Description
2.6.1.	Calls are routed to route patterns based upon the dialed number. A telephony route pattern must be created that matches the pilot directory numbers and routes requests over the SIP trunk to the SES to the Abrazo-E. The SES will also match the pilot DN and route to the Abrazo-E.
	A route pattern should be created for all of the pilot directory numbers. The example below specifies that incoming numbers beginning with <b>732852</b> with a length of ten digits get routed to Tango.
	Use the <b>change inc-call-handling-trmt trunk-group g</b> command, where "g" is the trunk assigned to the PSTN trunk. The example below specifies the trunk ID as <i>56</i> .
	change inc-call-handling-trmt trunk-group 56
	Page1 of3 INCOMING CALL HANDLING TREATMENTService/CalledCalledDel InsertPer Call NightFeatureLenNumberCPN/BNServtie11173285213
	Use the <b>change aar analysis 0</b> command, and enter the dial string need to be matched.
	change aar analysis 0 Page 1 of 2 AAR DIGIT ANALYSIS TABLE Location: all Percent Full: 1
	Dialed Total Route Call Node ANI String Min Max Pattern Type Num Reqd
	732852 10 10 1 aar n

2.6.2.	Calls are routed to route patterns based upon the dialed number. A telephony route pattern must be created that matches the service pilot pool numbers and routes requests over the SIP trunk to the SES to the Abrazo-E. The SES will also match the service pilot pool numbers and route to the Abrazo-E.
	A route pattern should be created for all of the service pilot pool numbers beginning with the dial plan analysis table. In the example below, our service pilot pool number is <b>7328522963</b> . The leading number <b>3</b> is configured as a feature access code in the dial plan analysis table.
	NOTE: The leading digit of the service pilot pool numbers must be defined as a feature access code (fac) in the dial plan analysis table.
	change dialplan analysis       Page 1 of 12         DIAL PLAN ANALYSIS TABLE       DIAL PLAN ANALYSIS TABLE         Location: all       Percent Full: 0
	DialedTotalCallDialedTotalCallDialedTotalCallStringLengthTypeStringLengthTypeStringLengthType03fac13fac25ext31fac45udp55ext65ext

ypically used for pub ons. Either the AAR of access-codes				• 1	•			
access-codes								
viated Dialing List viated Dialing List viated Dialing List al - Prgm Group Lis Announcemen Answer Bac Attendan ernate Routing (AAR Selection (ARS) - Automatic Callba	2 Access 3 Access t Access t Access k Access t Access <b>) Access</b> Access C ck Activ	Code: Code: Code: Code: Code: Code: Code: Code: Code: Code: Code: Code: Code:	101 102 103 106 *550 <b>3</b> 9 *5	AC) Access Code Deactivatic	on:	#5	of	8
	Announcemen Answer Bac Attendan Frnate Routing (AAR Selection (ARS) - Automatic Callba Activation Busy/D	Announcement Access Answer Back Access Attendant Access Frnate Routing (AAR) Access Selection (ARS) - Access C Automatic Callback Activ Activation Busy/DA: *2	Announcement Access Code: Answer Back Access Code: Attendant Access Code: Selection (ARS) - Access Code 1: Automatic Callback Activation: Activation Busy/DA: *2 All:	Announcement Access Code: 106 Answer Back Access Code: *550 Attendant Access Code: rnate Routing (AAR) Access Code: 3 Selection (ARS) - Access Code 1: 9 Automatic Callback Activation: *5 Activation Busy/DA: *2 All: *551	Announcement Access Code: 106 Answer Back Access Code: *550 Attendant Access Code: arnate Routing (AAR) Access Code: 3 Selection (ARS) - Access Code 1: 9 Access Code Automatic Callback Activation: *5 Access Code Activation Busy/DA: *2 All: *551 Deactivation	Announcement Access Code: 106 Answer Back Access Code: *550 Attendant Access Code: 3 Selection (ARS) - Access Code 1: 9 Automatic Callback Activation: *5 Access Code 2: Deactivation: so Deactivation: *5	Announcement Access Code: 106 Answer Back Access Code: *550 Attendant Access Code: 3 Selection (ARS) - Access Code 1: 9 Automatic Callback Activation: *5 Access Code 2: Deactivation: #5 Activation Busy/DA: *2 All: *551 Deactivation: #2	Announcement Access Code: 106 Answer Back Access Code: *550 Attendant Access Code: arnate Routing (AAR) Access Code: 3 Selection (ARS) - Access Code 1: 9 Automatic Callback Activation: *5 Access Code 2: Automatic Callback Activation: *5 Activation Busy/DA: *2 All: *551 Deactivation: #2

Create a route pattern that will send the piolet number to the SES SIP. 2.6.4. Use the **change route-pattern** *n* command, where *n* is the number of an unused route pattern. Enter a descriptive name for the **Pattern Name** field. Set the **Grp No** field to the trunk group number created for the SES SIP trunk. Set the Facility Restriction Level (FRL) field to a level that allows access to this trunk for all users that require it. The value of  $\boldsymbol{0}$  is the least restrictive level. The default values may be retained for all other fields. change route-pattern 1 Page 1 of 3 Pattern Number: 1 Pattern Name: ToTango SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits DCS/ IXC QSIG Dats Tntw 1: **1** 0 n user 2: n user 3: n user 4: n user 5: user n 6: n user BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR

rest

rest

rest

rest

rest

rest

Dgts Format Subaddress

none

none

none

none

none

none

0 1 2 M 4 W Request

1: yyyyyn n

2: ууууул п

3: yyyyyn n 4: yyyyyn n

5: уууууп п

6: уууууп п

none

# 2.7. SIP Line Configuration

This section describes the steps for setting Off-PBX-Telephones in Avaya Communication Manager for proper routing of calls from Avaya Communication Manager to Avaya SES, ultimately destined for the Tango Abrazo-E.

Step	Description
2.7.1.	Using the <b>change system-parameters customer-options</b> command, validate that <b>OPS</b> has available licenses for the Abrazo solution because acts as a SIP telephone.
	change system-parameters customer-options Page 1 of 11 OPTIONAL FEATURES
	G3 Version: V15 Software Package: Standard Location: 1 RFA System ID (SID): 1 Platform: 7 RFA Module ID (MID): 1
	USED Platform Maximum Ports: 900 147 Maximum Stations: 450 29 Maximum XMOBILE Stations: 10 0 Maximum Off-PBX Telephones - EC500: 450 1 Maximum Off-PBX Telephones - OPS: 450 12 Maximum Off-PBX Telephones - PBFMC: 0 0 Maximum Off-PBX Telephones - PVFMC: 0 0 Maximum Off-PBX Telephones - SCCAN: 100 0
2.7.2.	(NOTE: You must logoff & login to effect the permission changes.) Determine the feature name extensions and feature access codes by using the <b>change dial</b> <b>plan analysis</b> and <b>change feature access codes</b> commands. Analyze these to ensure there is no conflict with the Abrazo mid-call services (conference, sacada, transfer).
2.7.3.	Create a Class of Service (COS) and Class of Restriction (COR) set by using the <b>change class-of-service</b> and <b>change class-of-restriction</b> commands. This essentially defines the Avaya feature set that will be available to the SIP phone.

Every Abrazo user must be defined as an off-PBX station in order to enable simultaneous ringing to the Abrazo-E. To do this, go to the **Stations with Off-PBX Telephone Integration** screen and map the Avaya Communication manager extension to the extension defined in the SES. One OPS entry is need for H.323 extensions, two entries are needed for SIP extensions, one for the primary number and one for the mobile endpoint. For more information, refer to **Section 3.3**.

	Example for H.323 extension:
2.7.4.	• Set the <b>Station Extension</b> to the station extension of Abrazo-E as configured above (The example which follows uses 34071.)
	• Set Application to "OPS".
	• Set <b>Phone Number</b> to the number Abrazo will use for registration and call origination and terminations, which is the user portion of the SIP addresses defined for subscribers on Abrazo-E. This field maps the Avaya media server extension defined on the SES (example: 34071) to this station defined on Avaya Communication Manager.
	• Set <b>Trunk Selection</b> to the number of the SIP trunk group connected to the SES server.
	• Set <b>Configuration Set</b> to the set to be used for IP phone call treatments as defined above.
	• Set Mapping Mode to "both".
	• Set <b>Call Limit</b> to "4".
	change off-pbx-telephone station-mapping 34071 Page 1 of 2 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
	Stations with OFF-PBA TELEPHONE INTEGRATIONStation Application Dial Phone NumberTrunkConfigurationExtensionPrefixSelectionSet34071 OPS -3407111
	change off-pbx-telephone station-mapping 34071 Page 2 of 2 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION Station Call Mapping Calls Bridged Extension Limit Mode Allowed Calls 34071 4 both all both

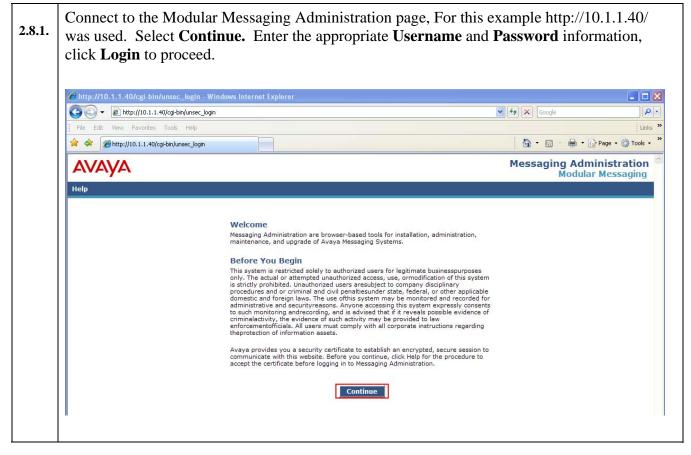
2.7.5.	Example for SIP extension:
2.7.5.	Note: link the second SIP extension (34081) to the first extension (34080) so both station will ring when 34080 is called.
	• Set the <b>Station Extension</b> to the station extension of Abrazo-E as configured above (The example which follows uses 34080 and 34081.)
	• Set Application to "OPS".
	• Set <b>Phone Number</b> to the number Abrazo will use for registration and call origination and terminations, which is the user portion of the SIP addresses defined for subscribers on Abrazo-E. This field maps the Avaya media server extension defined on the SES (example: 34080 and 34081) to this station defined on the Communication Manager.
	• Set <b>Trunk Selection</b> to the number of the SIP trunk group connected to the SES server.
	• Set <b>Configuration Set</b> to the set to be used for IP phone call treatments as defined above.
	• Set Mapping Mode to "both".
	• Set <b>Call Limit</b> to "4".
	change off-pbx-telephone station-mapping 34071 Page 1 of 2 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION Station Application Dial Phone Number Trunk Configuration Extension Prefix Selection Set 34080 OPS - 34080 1 1
	34080 OPS - 34080 1 1
	change off-pbx-telephone station-mapping 34071 Page 2 of 2 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION Station Call Mapping Calls Bridged
	Extension Limit Mode Allowed Calls 34080 4 both all both
	34080 4 both all both

# 2.8. Voice Mail Configuration

This section describes the steps for configuring voicemail for extensions in the Avaya telephony infrastructure. For informational purposes, steps for both Avaya Modular Messaging and Avaya IA770 INTUITY AUDIX are included in this document. Use the setup information appropriate for the environment being configured.

Note: It is recommended that at least four rings be used to route a call to voice mail.

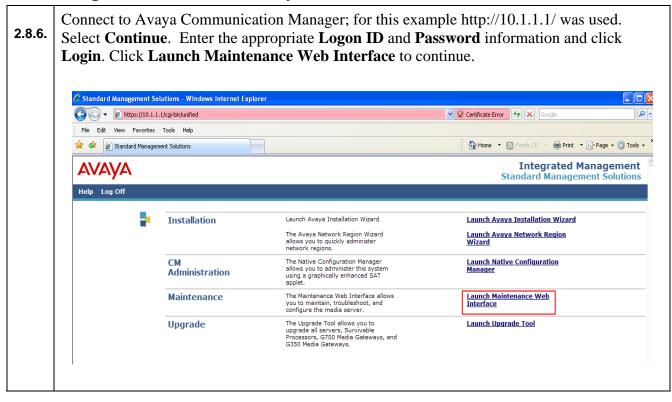
### Configure Subscriber on Avaya Modular Messaging



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Modular Messaging Messaging Administration
This server: 10.1.1.40
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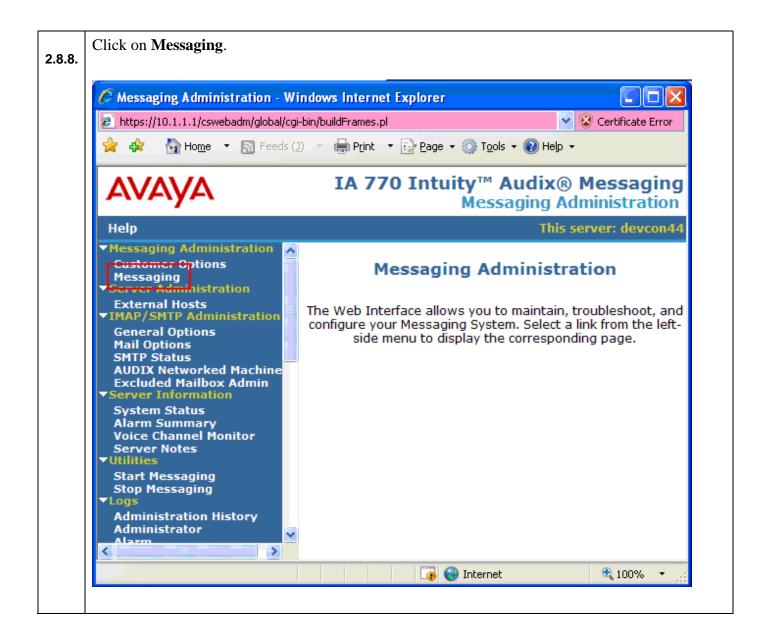
Step	Description						
2.8.4.	Select Add a N	ew Subscribe	r.				
	C Messaging Administration - W	indows Internet Explorer					
		ji-bin/do_login			💌 😵 Certificate Error	fy 🗙 Google	P-9
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	🖌 🏟 🍘 Messaging Administration					🟠 • 🖻 • 🖶 • E	Page 🔹 🎯 Tools 👻 🎽
	AVAYA						ar Messaging Administration
	Help Log Off					Th	is server: 10.1.1.40
	<ul> <li>Messaging Administration A Subscriber Management Activity Log Configuration Messaging Attributes Classes-of-Service Enhanced-Lists Sending Restrictions System Administration Networked Machines Trusted Servers</li> <li>Server Administration TCP/IP Network Configura External Hosts</li> <li>MAS Host Setup MAS Host Setup MAS Host Setup MAS Host Setup Mash Setup Mash Setup Mash Setup Mash Setup Madem/Terminal Configur Modem/Terminal Configur Modem/Terminal Configur Modem/Terminal Configur Modem/Terminal Removal Default Router Ping</li> <li>IMAP/SHTP Administration SMTP Options Mail Options Mail Options</li> <li>Server Information Server Status</li> </ul>	Manage Local S Subscriber Licenses Used: System Mailboxes: Subscriber Name 11, test glad, tom master, postmaster test, dylan test5 test678	6 of 100 1	Item         Total Subscribers: 7           Filtered Subscribers: 7           umber         Numeric Address           \$ 51002           \$ 51003           \$ 0001           99998           \$ 50005           \$ 52000           \$ 40003	COS     CID       0     1       0     1       0     1       0     1       0     1       0     1       0     1       0     1       0     1       0     1		
	Alarm Summary Disk Information Server Notes CMOS Settings RAID Status Rebuild DAID Status	Sort and Filter Subscribe		Launch Sub Delete the Select	scriber Options		
	Rebuild RAID Status Reboot Interval Vilities Rebuild RAID 1 Amov	Add a New Subscriber		Edit the Selec	ted Subscriber		
	Rebuild RAID 1 Array	Back			Help		

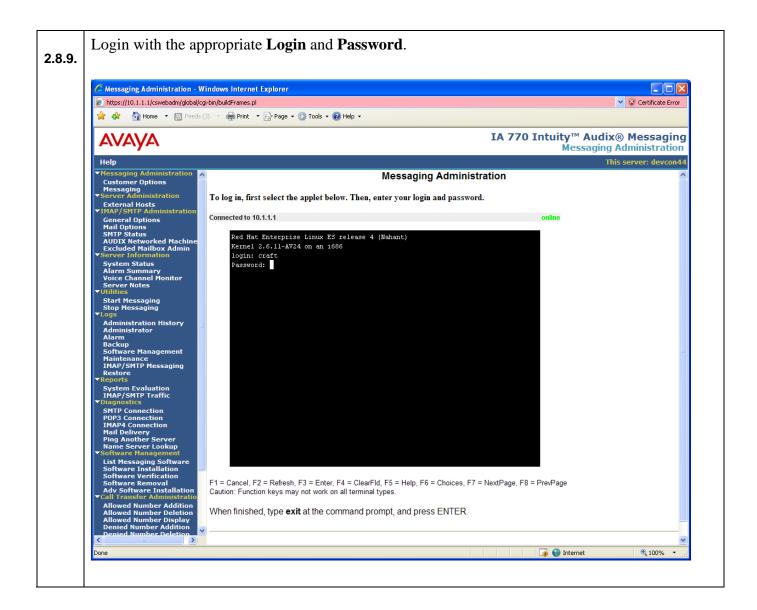
Step	Description	on			
2.8.5.		following user in <b>1e, Password</b> , <b>M</b>		umeric Address. S	Select <b>Save</b> to continue.
	🕒 🗸 🖉 🕹 🕹	(10.1.1.40/cgi-bin/do_login		V 😵 Certifica	te Error 😚 🗙 Google
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	AVAYA				Modular Messaging
	Help Log Off				Messaging Administration This servers 10.1.1.40
	<ul> <li>Messaging Administration Subscriber Management Accurcy Log Computation Messaging Attributes Classes-of-Service</li> </ul>	Add Local Subscriber			<u>^</u>
	Enhanced-Lists Sending Restrictions System Administration	BASIC INFORMATION			
	Request Remote Update Networked Machines Trusted Servers Server Administration	* (Required Fields) <u>*Last Nam</u>	2 last	<u>First Name</u>	
	Configure Using DCT TCP/IP Network Configuration External Hosts MAS Host Setup MAS Host Send	n Passwo		Mailbox Number	50000
	Windows Domain Setup Console Reboot Option	*Numeric Addres		PBX Extension Community ID	
	Date/Time/NTP Server Syslog Server Modem/Terminal Display Modem/Terminal Configuration				
	Modem/Terminal Removal TCP/IP Service Settings TIMAP/SMTP Administration	SUBSCRIBER DIRECTORY			
	SMTP Options Mail Options IMAP/SMTP Status Server Information	Email Hand		ASCII Version of Name	
	Server Status Alarm Summary Disk Information Server Notes		-   L]		
	Server Notes CMOS Settings RAID Status Rebuild RAID Status Reboot Interval	SUBSCRIBER SECURITY			
	- Utilities	Immediately Expire Password	no 💌	Is Mailbox Locked?	no M
	Rebuild AGU 1 Anray CD/0VD Mount CD/0VD Unmount CD/0VD Eject Messaging 0B Audits Start Messaging Stap Messaging Stutdown Server Reboot Server	MAILBOX FEATURES			
	Stop Messaging Shutdown Server Reboot Server	Personal Operator Mailbo		Personal Operator Schedule	Always Active
	Administration History	<u>TUI Message Orde</u>		Intercom Paging	paging is off
	Alarm Backup Command Line History ELA Delivery Failures IMAP/SMTP Maintenance Messaging Start-up	VoiceMail Enable	ves ¥		
	MSS DCT Configuration Log	SECONDARY EXTENSIONS			
	Restore Server Events Software Management Subscriber Activity Web Server	No Secondary Exten	ions	sAdd Seconda	ry Extension
	<ul> <li>Reports</li> <li>IMAP/SMTP Traffic</li> <li>Messaging Measurements</li> <li>System Evaluation</li> </ul>		(	Delete Caller	Application (none) V
	TCP/IP Packet Statistics Diagnostics Alarm Origination				
	LDAP Connection SMTP Connection POP3 Connection IMAP4 Connection	MISCELLANEOUS	1	Miscellaneous2	
	Mail Delivery Ping Another Server Name Server Lookup	Miscellaneou	2	Miscellaneous4	
	Software Management Messaging Software Display Software Display Software Verification Software Removal Software Removal	V Date Heg			×



#### **Configure Subscriber on Avaya IA770 INTUITY AUDIX**

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🚖 🍄 🙋 devcon44		🚹 Home 🔹 🔊 Feeds (J) 🔹 🖶 Print 🔹 📑 Page 🔹 👔
AVAYA		Integrated Manag Maintenance Web Pa
Help Exit		This Server: [1]
Agent Status SNMP Agents	5	Notice
SNMP Traps	© 2001-2007 Avaya Inc. All Rights Reserved.	
Filters SNMP Test Diagnostics	Copyright	
Restarts System Logs	Except where expressly stated otherwise, the Product is protec	ted by copyright and other laws respecting proprietary rights.
Ping Traceroute	Unauthorized reproduction, transfer, and or use can be a crimin	al, as well as a civil, offense under the applicable law.
Netstat Modem Test	Third-party Components	
Network Time Sync erver Status Summary Process Status	Components"), which may contain terms that expand or limit rig	roduct may contain software distributed under third party agreements ("Third Party hts to use certain portions of the Product ("Third Party Terms"). Information identifyin are available on Avaya's web site at: <u>htb://support.avaya.com/ThirdPartyLeense/</u>
Shutdown Server Server Date/Time Software Version	<u>Trademarks</u>	
erver Configuration Configure Server	Avaya is a trademark of Avaya Inc.	
Restore Defaults Eject CD-ROM	MultiVantage is a trademark of Avaya Inc.	
er Upgrades nage Software ke Upgrade Permanent ot Partition nage Updates	All non-Avaya trademarks are the property of their respective o	vners.
OS Upgrade Backup/Restore		
Backup Now Backup History		
Schedule Backup Backup Logs View/Restore Data		
Restore History Format CompactFlash		
curity Administrator Accounts		
Login Account Policy Login Reports Modem		
Server Access Syslog Server		
License File Authentication File		
Firewall Tripwire Tripwire Commands		
install Root Certificate		
Web Access Mask edia Gateways		
Configuration scellaneous File Synchronization		
File Synchronization IP Phones Download Files		
CM Phone Message File Tftpboot Directory		
Serial Numbers SES Software		





O Messaging Administration - W	indows Internet Explorer	
https://10.1.1.1/cswebadm/global/cg	-bin/buildFrames.rl	💌 😵 Certificate Erro
Αναγα		IA 770 Intuity™ Audix® Messagir Messaging Administratio
Help Messaging Administration	Magaaasing Administ	This server: devcon
Customer Options Messaging	Messaging Administ	
▼Server Administration External Hosts ▼IMAP/SMTP Administration	To log in, first select the applet below. Then, enter your login and password	d.
General Options Mail Options	Connected to 10.1.1.1	online
SMTP Status AUDIX Networked Machine	AUDIX Active Alarms: A add subscriber 51020	Logins: 1 Page 1 of 2
Excluded Mailbox Admin <ul> <li>Server Information</li> <li>System Status</li> </ul>	SUBSCRIBER	rage 1 of 8
Alarm Summary Voice Channel Monitor	Name: Enter Name Locked? n	
Server Notes <ul> <li>Utilities</li> </ul>	Extension: <u>51020</u> Password: COS: <mark>glass00</mark> Miscellaneous 1:	
Start Messaging Stop Messaging ▼Logs	Switch Number:	
Administration History Administrator	Secondary Ext: Miscellaneous 4: Account Code: Covering Extension:	
Alarm Backup	Broadcast Mailbox?	
Software Management Maintenance IMAP/SMTP Messaging	Email Address: <u>510200devcon44.</u>	
Restore <b>Reports</b>		
System Evaluation IMAP/SMTP Traffic		
▼Diagnostics SMTP Connection POP3 Connection		
IMAP4 Connection Mail Delivery		
Ping Another Server Name Server Lookup	Press [ENTER] to execute or press [CANCEL] to abort enter command: add subscriber 51020	
▼Software Management List Messaging Software Software Installation		extPage PrevPage
Software Verification Software Removal	F1 = Cancel, F2 = Refresh, F3 = Enter, F4 = ClearFld, F5 = Help, F6 = Choices, F7 =	= NextPage, F8 = PrevPage
Adv Software Installation Call Transfer Administratio	Caution: Function keys may not work on all terminal types.	5, 5

# 2.9. Verification Steps

Use the following steps to verify the configuration on Avaya Communication Manager:

Step	Description
2.9.1.	From the Avaya Communication Manager SAT, use the <b>status trunk n</b> command where <b>n</b> is the trunk group number to verify that the SIP trunk group is in service.
2.9.2.	From the Avaya Communication Manager SAT, use the <b>status signaling-group n</b> command where <b>n</b> is the signaling group number to verify that the SIP signaling group is in service.

# 3. Configure Avaya SIP Enablement Services

This section describes the steps required for Avaya SIP Enablement Services to support the configuration in **Figure 3: Interoperability Configuration Diagram.** The following pages provide step-by-step instructions on how to create the media server entry, define the host address map entry along with contact information for the Tango Abrazo-E.

Note: It is assumed that that the appropriate license and authentication files have been installed on the servers and that login and password credentials are available. It is assumed that the reader has a basic understanding of the administration of Avaya SIP Enablement Services and has access to the SES web browser.

# 3.1. SES Software Configuration

The Abrazo solution needs the ability to trigger originating and terminating services. Avaya has provided a way to do this using the standard SIP P-Asserted-Identity header. If the header is detected in the Invite requests, the SES marks the transaction for origination services, otherwise it assumes termination services. For this functionality to be enabled on Avaya, do the following steps.

Step	Description
3.1.1.	Log into the SES as root.
3.1.2.	Edit the file: /usr/impress/sip-server/etc/ccs.conf
3.1.3.	Under the Proxy section remove the # before the line: EnableThirdPartyOriginatingProcessing=true
3.1.4.	Stop the server by entering: stop –a
3.1.5.	Verify all components are shutdown by entering: statapp
3.1.6.	Once all are shutdown restart the server by entering: start –a

# 3.2. SIP Trunk Configuration

On the SES, the Abrazo solution needs to be configured using both a SIP trunk and a SIP line. The SIP trunk interface(s) are used by the Abrazo solution to terminate a call to the wireless operator's network. A SIP trunk is also used by the PBX to route mobile calls to the Abrazo solution via the enterprise using Pilot Directory Numbers or Service Pool Numbers. Select Launch Administration Web Interface.

Step	Description				
3.2.1.	address/AD	MIN in an Inte	ion web interface by using the rnet browser window, where i appropriate credentials. The f	<b>p-address</b> is th	e IP address of the
	AVAYA				ated Management
	Help Log Off				
	•	SES Administration	The Administration Web Interface allows you to administer this SES server.	Launch SES Administration Interface	
l		Maintenance	The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the server.	Launch Maintenance Web Interface	

	Description			
]	The following screen	is displayed.		
	AVAYA			
ŀ	lelp Exit			
	Top <sup>III</sup> Users	🖡 Тор		
	Address Map Priorities Adjunct Systems	Manage Users	Add and delete Users.	
	<ul> <li>Aggregator</li> </ul>	Manage Address Map Priorities	Adjust Address Map Priorities.	
	<ul> <li>Certificate Management</li> <li>Conferences</li> </ul>	Manage Adjunct Systems	Add and delete Adjunct Systems.	
	Emergency Contacts Export/Import to ProVision	Manage Event Aggregators	Add/Delete Event Aggregators.	
	• Hosts	Certificate Management	Manage Certificates.	
IM logs Communication Manager	Manage Conferencing	Add and delete Conference Extensions.		
	Servers Communication Manager Extensions	Manage Emergency Contacts	Add and delete Emergency Contacts.	
<ul><li>SIP Phone</li><li>Survivable</li></ul>	<ul> <li>Server Configuration</li> <li>SIP Phone Settings</li> </ul>	Export Import to ProVision	Export and import data using ProVision on this host.	
	Survivable Call Processors	Manage Hosts	Add and delete Hosts.	
	System Status Trace Logger Trusted Hosts	IM logs	Download IM Logs.	
		Manage Communication Manager Servers	Add and delete Communication Manager Servers.	
		Manage Communication Manager Extensions	Add and delete Communication Manager Extensions.	
		Server Configuration	View Properties of the system.	
		Manage SIP Phone Settings	Add/Delete Phone Settings	
		Manage Survivable Call Processors	Add and delete Survivable Call Processors.	
l		System Status	View System Status.	
		Trace Logger	Manage SIP Trace Logs.	
I		Manage Trusted Hosts	Add and delete Trusted Hosts.	

Step	Description								
5.2.3.	Outbound calls are first routed by Avaya Communication Manager to the SIP trunk group. These calls are then subject to further routing decisions determined by Host Address Maps in the Avaya SES.								
	Navigate to the <b>Add Host Address Map</b> screen by selecting <b>Hosts &gt; List</b> from the left pane. The <b>List Hosts</b> screen is displayed. Click on <b>Map</b> in the right pane.								
	Help Exit								
	Top Dusers Address May <sup>be</sup> riorities	List Hosts							
	Adjunct Systems		nands		<u>Host</u>	Туре	<u>SES</u> <u>Version</u>		
	<ul> <li>Aggregator</li> <li>Certificate Management</li> </ul>	Edit Map Go-To	Test-Link	Delete	10.1.1.10	SES combined home-edg	e SES-5.1.0.0-414.3f		
	Conferences	Migrate Home/Edge							
	Emergency Contacts Export/Import to ProVision								
	- Hosts								
	List Migrate Home/Edge								
	IM logs								
	<ul> <li>Communication Manager Servers</li> <li>Communication Manager</li> </ul>								
	Extensions  Externation  Extensions								
	■ SIP Phone Settings								
	Survivable Call Processors								
	System Status								
	System Status Trace Logger Trusted Hosts								

Step	Description						
3.2.4.	The List Host Address Map screen is displayed. Scroll to the bottom of the screen if needed. Click on Add Map in New Group.						
		(user) 2.168.82.147:5060;transport=udp t: Delete Group					
	Edit Delete	(user) 2.168.81.144:5060;transport=udp					
	Add Another Map Add Another Contac	Delate					
	Add Map In New Group		-				
3.2.5.	The Add Host Address Map screen is disp	played.					
	Add Ho Name* Pattern* Replace URI Fields marked *						
	<ul> <li>Use the Add Host Address Map screen to specify what calls should be routed to the A</li> <li>For the Name field, enter a descript</li> <li>For the Pattern field, define an app matches the format of the PDN/Ser mobile calls into the Abrazo-E.</li> <li>Retain the check in Replace URI, and c</li> </ul>	Abrazo-E. ive name to denote the routing p propriate syntax for address mapp vice Pilot Pool Numbers that are	attern. bing that				

Step	Description						
	A <b>Continue</b> screen is displayed to confirm the addition.						
3.2.6.							
	- Continue						
	Host address map Tango-E added.						
	Continue						
	Host address map patterns must be defined for Pilot Directory Number (PDNs) and Service						
3.2.7.	Pilot Pool numbers used by the Abrazo solution. Repeat steps above as required.						
	Click the Continue button. The List Host Address screen is redisplayed, showing the						
3.2.8.	newly added item. Define the contact address for the Abrazo-E (Tango-E) by clicking on						
	Add Another Contact on the line below Tango-E.						
	@192120072140.5000;(tan3port=aup						
	Add Another Map Add Another Contact Delete Group						
	Add Another Map Add Another Contact Delete Group						
	Edit Delete Tango-E						
	Add Another Map Add Another Contact Group						
	Edit Delete ftr1_10_rte						
	Edit Delete sip:\$(user)						
	@192.168.81.149:5060;transport=udp						
3.2.9.	The Add Host Contact screen is displayed.						
	Add Host Contact						
	Handle Tango-E						
	Contact*						
	Fields marked * are required.						
	Add						

Step	Description	
3.2.10.	The Contact field specifies the destination for the call. Populate the <b>Contact</b> field with the Abrezo IP address that the SES should substitute into the required URI before sending a message to the Abrazo Tango-E. The Avaya SES replaces <b>\$(user)</b> with the user portion of the request URI before sending the message. Click the <b>Add</b> button.	
	Add Host Contact	
	Handle Tango-E1	
	Contact* sip:\$(user)@10.1.1.65:5060;transport=udp	
	Fields marked * are required.	
	Add	
3.2.11.	A <b>Continue</b> screen is displayed to confirm the addition. Click the <b>Continue</b> button.	
3.2.11.	Host contact ip:\$(user)@10.1.1.65:5060;transport=udp added for map entry Tango-E	
	A Host Contact must be defined for each of the Host Address Maps provisioned for the	
3.2.12.	Abrazo PDNs and Service Pool numbers. Repeat steps above as required.	

Step	Description	
3.2.13.	Administer the Abrazo-E as a trusted host so that the SES will not challenge SIP messages from the Abrazo-E. Select <b>Trusted Hosts</b> $\rightarrow$ <b>Add Trusted Host</b> . The <b>Add Trusted Host</b> screen is displayed. Enter the <b>IP address</b> of the Abrazo-E and, if desired, a descriptive <b>Comment</b> . Click <b>Add</b> to continue.	
	Add Trusted Host	
	IP Address*: 10.1.1.65	
	Host* 10.1.1.10 🛩	
	Comment: TangoE	
	Fields marked * are required.	
	Add	
	A Continue screen is displayed to confirm the addition. Click the Continue button	
3.2.14.	Continue	
	Trusted Host 10.1.1.65 added.	
	Continue	
	To apply the abanges in the above stong plick Undets at the bettern of the left range. This	
3.2.15.	To apply the changes in the above steps, click <b>Update</b> at the bottom of the left pane. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administrative changes performed to that point.	

#### 3.3. SIP Line Configuration

On the SES, the Abrazo solution needs to be configured using both a SIP trunk and a SIP line. The SIP line interface(s) are used to offer originating PBX services to the mobile user. The configuration of the Avaya desk phone dictates how the Abrazo user is added to the SES.

- If the user has an H.323 or digital desk phone, then an SES ID must be created with the same extension that was created for the desk station.
- If the user has a SIP desk phone, he or she will need a new ID created. (Since the desk phone is SIP, the user already has an SES ID created.)

TMA; Reviewed:	
SPOC 11/20/2008	

Step	Description	
	Select Users $\rightarrow$ Add. Fill in the screens as follows depending on the user's type of desk	
3.3.1.	phone.	
	Add User	Add User
	Primary Handle* 34071   User ID 34071   Password* ••••••••   Confirm Password* ••••••••   Host* 192.168.80.99   First Name* Joe   Last Name** Example   Address 1 •••••••   Address 2 ••••••   Office ••••••   City •••••   State •••••   Country ••••	Primary Handle*34074User ID34704Password*
	Zip Add Media Server Extension Fields marked * are required.	Add Media Server Extension Fields marked * are required.
	Adding Joe (H.323)	Adding Mary (SIP)
	<ul> <li>Ensure the following fields are populated as described below:</li> <li>For the <b>Primary Handle</b> field, enter the phone number (OPS station number) of the Abrazo subscriber (example: 34071).</li> <li>For the <b>User ID</b> field of a H.323 or digital user, enter the user's extension/station number. For a SIP user, create a new SIP number. The <b>UserID</b> field must match the Abrazo SIP number provisioned for the subscriber.</li> </ul>	
	registration with the SES. The I are not configured as trusted no password should be entered; ho message.	password to be used by the Abrazo solution during <b>Password</b> field is a required field for SIP users that des. For trusted nodes, like the Tango Abrazo, the wever, it is not included in the SIP registration
	<ul> <li>For the Host field, enter the IP register.</li> <li>Click the Add Media Server E</li> </ul>	address of the Avaya SES with which Abrazo will <b>Extension</b> check box.
	Click Add. A confirmation screen is di	

Step	Description	
	User ID 34071 added.	User ID 34704 added.
	Continue	Continue
	<b>Confirming the Addition of Joe</b>	<b>Confirming the Addition of Mary</b>
	Click Continue. The Add Communication Ma	nager Extension screen is displayed.
	Add Communication Manager Exten	Add Communication Manager Ext
	Add Communication Manager extension for user 34071. Extension 34071 Communication Manager TO-10.1.1.1 Server	Add Communication Manager extension for user 34074 Extension 34074 Communication Manager TO-10.1.1.1 Server
	Fields marked * are required.	Fields marked * are required.
	Adding Media Server Extension for Joe	Adding Media Server Extension for Mary

Step	Description	
3.3.2.	Use the Add Media Server Extension screen to set the corresponding telephone extension. For the Extension field, enter the extension of corresponding OPS station (same one used for primary handle when adding user). For the Media Server field, select the media server on which the desk phone is configured. The SES should automatically populate or default to this field. Click Add. A confirmation screen is displayed.	
	Continue Extension 34071 added for user 34071	
	Click <b>Continue</b> . A list of media server extensions for that user is displayed.           List Media Server Extensions           Media Server extensions for user 34071.           Commands         Extension         Host	
	Free Edit User Delete 34071 34071 procr 192.168.80.99	
3.3.3.	To apply the changes in the above steps, click <b>Update</b> at the bottom of the left pane. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administrative changes performed to that point.	

#### 3.4. Verification Steps

Use the following steps to verify the configuration on the Avaya SES:

Step	Description
3.4.1	From the Avaya SIP Server Management interface, select <b>Trusted Host</b> $\rightarrow$ List to verify that the IP address of the Abrazo Tango-E is listed as a trusted node.
3.4.2	From the Avaya SIP Server Management interface, navigate to User $\rightarrow$ Registered Users to verify that all SIP endpoints are registered with their respective Avaya SES.

# 4. Provision the Tango Networks Abrazo-E

This section describes the processes required to integrate the Abrazo solution with a SIPenabled Avaya Communication Manager. This document assumes that the Abrazo solution has already been provisioned with enterprise information and wireless carrier information. The integration process includes the following steps:

- Provision Avaya Communication Manager in the Abrazo system.
- Create an Avaya outbound SIP trunk to route traffic from the Avaya telephony infrastructure to the Abrazo system.
- Create inbound SIP lines to route traffic from the Abrazo system to the Avaya telephony infrastructure.
- Provision the dial plans for the Avaya telephony infrastructure in the Abrazo system.
- Provision the voicemail system used with the Avaya telephony infrastructure.
- Provision Abrazo subscribers that use the Avaya telephony infrastructure.

Step	Description
4.1.1.	To add the Avaya telephony infrastructure to the Abrazo-E, select Voice Network $\rightarrow$ <b>PBX</b> $\rightarrow$ <b>Add</b> . Enter a <b>PBX Name</b> (e.g., "AvayaPBX"). Select <b>Submit</b> to continue.
	Add New PBX
	* PBX Name:
	* PBX Type: Avaya 5.0 🖌
	Default Ingress PBX for Country:
	PBX Domain:
	Local Area/City Codes:
	Pilot Numbers:
	*-indicates required field
	Submit Clear Cancel
4.1.2.	Define a new trunk group and add trunk group members to communicate with the Avaya telephony infrastructure. To define a new trunk group, select the PBX to which the Trunk Group is to be added from the list displayed by <b>Voice Network</b> $\rightarrow$ <b>PBX</b> $\rightarrow$ <b>List all</b> , as shown below. Select the PBX you want from the list. Click the Add Trunk Group button.

Step	Description
4.1.3.	The Add Trunk Group screen is displayed. Select Next to continue.
	Add Trunk Group
	PBX Name: AvayaPBX  * Trunk Group Name: Dial Plan: URL Parameters:
	*-indicates required field       Next     Cancel
	Enter trunk group information on the Add Trunk Group screen. Click on <b>Next</b> to add trunk group members, clicking on <b>Submit</b> .
	Add Trunk         Trunk Group Name: Trunk Group 1         * Host Address:         * Port:         \$ Port:         \$ Sobo         * Transport Type:         *-Indicates required field         Submit       Back         Cancel

Step	Description
4.1.4.	Define a new line group and add line group members to communicate with the Avaya telephony infrastructure. To define a new line group, select the PBX to which you want to add the Trunk Group (Voice Network $\rightarrow$ PBX $\rightarrow$ List all). Select the PBX you want from the list. Click the Add Line Group button. You'll see the screen shown below.
	Add Line Group
	PBX Name: AvayaPBX  * Line Group Name: URL Parameters:
	*-indicates required field           Next         Cancel
	Enter line group information on the Add Line Group screen. Select <b>Next</b> to add line group members. You'll see the screen shown below.
	Add Line
	Line Group Name: avayalinegrup
	* Host Address: 🕜
	* Port: 5060
	* Transport Type:  UDP OTCP
	*-indicates required field
	Submit Back Cancel
	Figure 4 Add Line Screen
	Enter line information on the Add Line screen. Select Submit to add the line members.

Step	Description	
4.1.5	Update the voice network topic for the newly configured Avaya teleph with the enterprise's voice network layout. At a minimum, add the dia to enable abbreviated dialing support as well as forced on-net services information, select Voice Network $\rightarrow$ Dial Plan $\rightarrow$ Add. Select Subr	l plan information . To add dial plan
	Add Dial Plan	
	* Dial Plan Name:	0
	Country and Area/City Code Settings:	
	> Country: United States (1)	0
	Local Number Length: 10	<b>2</b>
	Domestic Minimum Lengtl 10	<b>@</b>
	Domestic Maximum Length 10	<b>@</b>
	Local Numbers require an area code: 🕞	0
	Default Area/City Code	0
	Note: the Area/City code is used for mobile originated calls.	
	Prefix Settings:	
	On Net Dialing Prefix:	0
	Local Off Net Dialing Prefix:	0
	Domestic LD Off Net Dialing Prefix 1	0
	International Off Net Dialing Prefs 011	0
	*-indicates required field	
	Submit Clear Cancel	

Step	Description	
4.1.6.	The voice mail server used with the Avaya telephony infrastructure must be specified so the Abrazo solution can provide a single voice mail solution. To add a Voice Mail Server, select <b>Voice Network -&gt; Voice mail -&gt; Add</b> . Select PBX as the <b>Voice Mail Server</b>	
	<b>Type</b> . The following screen is displayed.	
	Add Voice Mail Server	
	* Voice Mail Server Name: 2	
	* Voice Mail Server Type: PBX 🔽 🕜	
	* Voice Mail Retrieval Number:	
	% Voice Mail Deposit Number: 🛛 😵	
	*-indicates required field	
	Submit Clear Cancel	
	For <b>Voice Mail Retrieval Number</b> , enter the number that routes callers to their voicemail. For <b>Voice Mail Deposit Number</b> , enter the feature code defined on the PBX to transfer the call to voice mail.	

Description		
Add or modify Abrazo subscribers to associate them with the Avaya telephony infrastructure. To add subscribers, select <b>Subscriber</b> → <b>Add</b> . The following screen is displayed.		
Select AvayaPBX as the user's <b>HomePBX</b> field. The screen is modified to display additional fields ( <b>Line Group</b> , <b>Desk phone is SIP</b> ) as shown below. Select <b>Subm</b> it to continue.		
Add	Abrazo Subscriber	
≉ Last Name:		?
# First Name:		?
Display Name:		<b>@</b>
		<b>?</b>
% Mobile Number:		?
* Mobile Number Country:	United States (1)	?
✤ Home PBX:	AvayaPBX 💌	?
Alias:		<b>?</b>
Direct Inward Dial (DID):		<b>?</b>
Email Address:		?
✤ SIP Address:	@tango.com	?
Subscriber's Wireless Carrier: (Entries found in the Carrier List.)	<no carrier=""></no>	<b>?</b>
* Profile:	defaultNoSvc 💌	?
Mobile Policy Rule Set:	Default 💌	<b>?</b>
Mobile Policy Permission:	Default Permission	0
# Home Time Zone:	· · · · · · · · · ·	<b>?</b>
		0
		0
		0
-		<b>?</b>
_		<b>0</b>
	stant or WLAN:	<b>U</b>
Confirm Password:		
	*-indicates required field	
Submit Clear Cancel		
	Add or modify Abrazo subscribers infrastructure. To add subscribers displayed. Select AvayaPBX as the user's H additional fields (Line Group, D continue. Add * Last Name: * Last Name: * Last Name: * First Name: Display Name * Enterprise Desk Number: * Mobile Number Country: * Mobile Number Country: * Mobile Number Country: * Mobile Number Country: * Home PBX: Alias: Direct Inward Dial (DID): Email Address: * SIP Address: * Mobile Policy Rule Set * Mobile Policy Rule Set * Mobile Policy Permission * Home Time Zone: Daylight Saving Time Observed Dial Plan * Line Group: Desk phone is SIP: Home PBX Provides Orig Svcs: Password to access Mobile-Assis	Add or modify Abrazo subscribers to associate them with the Avaya telephony infrastructure. To add subscribers, select Subscriber→ Add. The following sc displayed. Select AvayaPBX as the user's HomePBX field. The screen is modified to disp additional fields (Line Group, Desk phone is SIP) as shown below. Select Su continue. Add Abrazo Subscriber * Last Name: bisplay Name: bisplay Name: bisplay Name: * Enterprise Desk Number: * Mobile Number: * Mobile Number: * Mobile Number: * Mobile Number: * Mobile Number: * Mobile Number: * SIP Addess: * SIP Addess: * SIP Addess: * SIP Addess: * SIP Addess: * SIP Addess: * Mobile Policy Rule Set: * Mobile Policy Rule Set: befault Nose ~ * Mobile Policy Rule Set: befault Nose ~ * Mobile Policy Rule Set: befault Nose ~ * Mobile Policy Rule Set: befault Policy Central Time (US and Canada) * Line Group: * Line Group: Desk phone is SIP: Home PBX Provides Orig Sves: © Password: Confirm Password: *-indicates required field

Description
The Avaya user's desk phone could be H.323, digital, or SIP. The Abrazo subscriber data should be added based on the following guidelines which depend on the protocol of the user's desk phone.
H.323 or digital User Desk Phone
If the user's desk phone is H.323 or digital, the SIP user added to the SES must match the extension set up for the station. This allows the Abrazo solution to register using the user ID as the station. When these IDs are the same, the Abrazo solution does not need to subscribe for voice mail events. The Avaya will send notifications based on the registration matching the station extension.
The following steps describe the Abrazo configuration required when the desk phone is H.323 or digital.
<ol> <li>Set the Abrazo Enterprise Desk Number to the extension defined for the user's station in the Avaya Communication Manager.</li> <li>Set the Abrazo SIP Address to the user's ID created on the SES. In this case the ID on the SES must match the extension for the user's H.323 station. The Abrazo solution will use this address for registrations.</li> <li>Leave the option Desk phone is SIP unchecked.</li> <li>Ensure the option Home PBX Provides Orig Svs is checked. When checked, Abrazo originates calls for the mobile user through the home PBX.</li> </ol>
SIP User Desk Phone
If the user's desk phone is SIP, the SES will contain two IDs for this user. One ID will match the extension for the station and this will be used by the SIP phone. A second ID is created that is used by the Abrazo. In this case, the Abrazo will register using this second ID. Since this ID does not match the station's extension, the Abrazo will also subscribe for voice mail events.
<ol> <li>The following steps describe the Abrazo configuration required when the desk phone is SIP.</li> <li>Set the Abrazo Enterprise Desk Number to the extension defined for the user's SIP station. The Abrazo solution will use this for voice mail subscriptions.</li> <li>Set the Abrazo SIP Address to the user's ID created on the SES. Because the user's desk phone is SIP, there will be two SES IDs created for this user—one for the SIP desk phone and one for the Abrazo SIP Address. The Abrazo solution will use this address for registrations.</li> <li>Check the option for Desk phone is SIP.</li> <li>Ensure the option Home PBX Provides Orig Svs is checked. By checking this field, the Abrazo always originates the call on behalf of the mobile user for mobile originations into their home PBX within the enterprise network.</li> </ol>

Step	Description
418	Repeat <b>Step 5.1.9</b> for each user to be added to the system.
4.1.0.	Repeat Step 5.1.9 for each user to be added to the system.

## 5. Interoperability Compliance Testing

Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab. Compliance testing verified the integration between Avaya Communication Manager and Tango Networks Abrazo Solution and the ability for an enterprise user to be accessible via one business number whether the user is in the office or mobile.

#### 5.1. General Test Approach

The general test approach was to make mobile originating and mobile terminating calls route through the Avaya telephony infrastructure. All feature functionality test cases were performed manually. In addition, testing entailed verifying different types of Avaya telephones and system features interacting with the Tango Abrazo solution. Tests were performed focusing on the following calling patterns:

- mobile originated calls routed through the Avaya telephony infrastructure terminating to a desk phone, mobile device, or the PSTN
- mobile terminated calls routed through the Avaya telephony infrastructure
- desktop originated calls routed to mobile devices

The following system features were tested to be available on the mobile device using the Abrazo service:

- Abbreviated Dialing Avaya Communication Manager allows extension dialing or internal dialing from the desktop phone. Abrazo allows the user to dial these same abbreviated codes from the mobile phone.
- **Call Hold and Retrieve** lets users temporarily disconnect from a call, use the telephone for another call, and then return to the original call. The Abrazo solution allows for subscribers to use this service.
- **Calling Line Identification (CLID)** provides the user information about the calling party. Abrazo supports calling line identification when it is the called party. Abrazo also supports ensuring that the enterprise identity of the caller is preserved when a call is initiated from the mobile phone. In this case although the call is made from a mobile phone, the calling line ID will be that of the Abrazo user's desktop phone.
- **Calling Name Identification (CNID)** provides the user with calling party name information. When Abrazo subscribers make a call from their mobile phone, Abrazo adds calling name information to the call so that calling name services are supported from the mobile phone.

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- **Call Transfer** lets users transfer the calling party in a currently established call from their mobile phone to another destination. This is implemented by the user entering a mid-call feature code followed by the transfer to number. There are two types of call transfers that are supported by this functionality:
  - **Blind Call Transfer** where the call is transferred without interaction between the user who initiated the transfer and the transfer destination.
  - **Consultative Call Transfer** where the call is transferred allowing interaction between the user who initiated the transfer and the transfer destination.
- The automatic bridged line appearance feature works for subscribers using H.323 desk phones and 9600 Series IP phones running SIP. When a voice call is established on the desk phone, a bridged line appearance remains on the desk phone. With this capability, the subscriber can simply press the bridged line appearance button to reenter the call from their desk phone.
- **Class of Service** allows or denies user access to some system features. The Abrazo-E supports COS for mobile originated calls over SIP lines.
- **Direct Inward Dialing** provides the user a separate number for the desk phone that can be accessed from the PSTN. The Abrazo solution supports enterprise Direct Inward Dialing.
- **Direct Outward Dialing** allows users inside an enterprise to dial directly to an external number. The Abrazo solution supports the mobile device dialing directly to an external number.
- **Flexible Dialing Support** Abrazo has a flexible dialing plan enabling PBX services to be provided to mobile users.
- **Intelligent Call Delivery** ensures that both the desk phone and mobile phone ring when the dialed number is an Abrazo subscriber.
- **Multiple Calls per Line** allows multiple calls to be delivered to a single number and have the incoming call information displayed to the user. Abrazo supports this feature on the mobile phone based on the ability to support call waiting for mobile phone devices.
- **Single Number Services** lets a user share one number with others that he or she wishes to communicate with. When this single number is dialed, the subscriber's enterprise desktop phone as well as mobile phone will ring. This service is provided by Abrazo and available when interworking with Avaya Communication Manager.
- Send All Calls allows the user to temporarily direct all incoming calls for the desk phone and mobile phone to call coverage regardless of the assigned call-coverage redirection criteria. When Send All Calls is activated, the Abrazo service is not invoked.

• Voice Mail Message Waiting Indication - provides a visible indication on the desk phone that there is a message waiting in the voice mail system. Abrazo supports supplying a Message Waiting indication on the mobile phone that indicates that there are voice mail messages in the enterprise voice mail system.

### 5.2. Test Results

The test objectives of section 5.1 were verified. The Tango Networks Abrazo Solution successfully completed all test cases for the features identified in section 5.1. Tango Networks is able to route inbound/outbound calls to/from Avaya Communication Manager with all services tested.

## 6. Support

Use the following contacts for technical support of Tango Networks Abrazo products:

- Web site: <u>http://www.tango-networks.com</u>
- Email: <a href="mailto:sales@tango-networks.com">sales@tango-networks.com</a>
- Telephone: +1 972-301-9316

# 7. Conclusion

These Application Notes describe the configuration steps required for integrating the Tango Networks Abrazo Solution into an Avaya telephony infrastructure. For the configuration described in these Application Notes, the Tango Networks Abrazo Solution was responsible for bridging landline connectivity to Avaya Communication Manager with the wireless connectivity to the GSM network. The functionality of the Avaya/ Tango Networks Abrazo Solution was validated via the DevConnect Program at the Avaya Solution and Interoperability Test Lab. All feature functionality test cases passed.

## 8. Additional References

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

- [1] *Administrator Guide for Avaya Communication Manager*, January 2008, Issue 4.0, Document Number 03-300509
- [2] *Installing and Administering SIP Enablement Services*, January 2008, Issue 5.0, Document Number 03-600768
- [3] Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide
- [4] Messaging Application Server (MAS) Administration Guide Release 3.1, February 2007

Product documentation for Tango Networks products may be found at: <a href="http://www.tango-networks.com">http://www.tango-networks.com</a>

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