



Avaya Solution & Interoperability Test Lab

Avaya Modular Messaging Release 3.0 SIP Integration with Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe the steps required to configure Avaya Modular Messaging SIP integration with an Avaya S8710 Media Server running Avaya Communication Manager. Avaya Modular Messaging is a unified messaging solution that addresses the many different messaging needs of customers. SIP connectivity between Avaya Communication Manager and Avaya Modular Messaging is accomplished through a SIP trunk provided by an Avaya SIP Enablement Services (SES) server over an IP network.

1. Introduction

These Application Notes describe the steps required to configure Avaya Modular Messaging SIP integration with an Avaya S8710 Media Server running Avaya Communication Manager. Avaya Modular Messaging is a unified messaging solution that addresses the many different messaging needs of customers. SIP connectivity between Avaya Communication Manager and Avaya Modular Messaging is accomplished through a SIP trunk provided by an Avaya SIP Enablement Services (SES) server over an IP network.

2. Configuration

The Avaya IP Telephony reference configuration used to verify these Application Notes is shown in **Figure 1**. The Avaya Modular Messaging servers, namely the Message Application Server (named DPMAS) and the Message Storage Server (named DPMSS), physically reside in the Main Location along with the Avaya S8710 Media Servers running Avaya Communication Manager and the Avaya SIP Enablement Services (SES) Server. The Avaya G650 Media Gateway in Branch Location 1 and the Avaya G350 Media Gateway in Branch Location 2 provide additional VoIP resources for Avaya Modular Messaging. A Windows XP PC is used for access to Avaya Communication Manager via the System Access Terminal and SES Administration web pages.

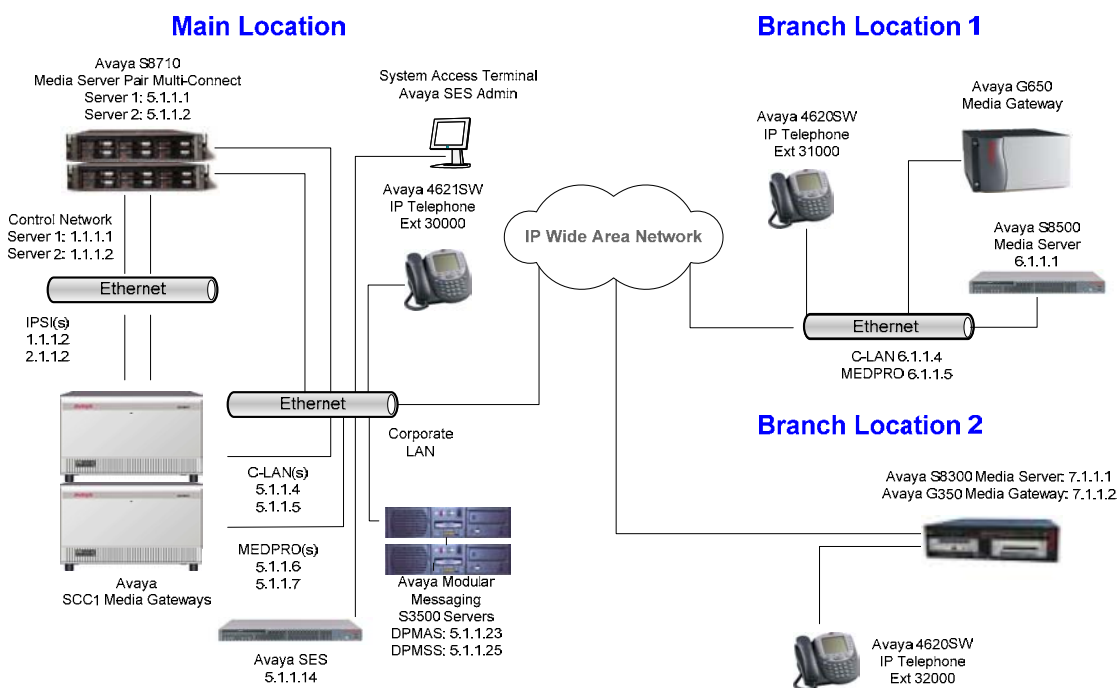


Figure 1: Network Configuration

3. Equipment and Software Validated

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

Equipment & Software	Version
Avaya Communication Manager Avaya S8710 Media Server Avaya S8500 Media Server Avaya S8300 Media Server	3.1.2 (R013x.01.0.632.1)
Avaya G650 Media Gateway IPSI (TN2312BP) C-LAN (TN799DP) MEDPRO (TN2302AP)	HW 12 FW 030 HW 01 FW 017 HW 20 FW 110
Avaya G350 Media Gateway	25.28.0
Avaya SIP Enablement Services	3.1 (3.1.0.0-018.0)
Avaya 4620SW IP Phone	2.3
Avaya 4621SW IP Phone	2.3
Avaya Modular Messaging Message Application Server (DPMAS) Message Storage Server (DPMSS)	3.0.495.0 3.0.495.0

Table 1: Equipment List

4. Configuring Avaya Communication Manager

There are a number of steps that must be performed on Avaya Communication Manager to enable integration with Avaya Modular Messaging via SIP. All the commands discussed in this section are executed on Avaya Communication Manager using the System Access Terminal (SAT). This section assumes that basic configuration on Avaya Communication Manager has been done already.

4.1. Verify Customer Options

The options for SIP trunking must be enabled in order to continue with the configuration. To check these options, run the **display system-parameters customer-options** command and ensure the fields highlighted in bold in the following screens are set.

Note: One SIP trunk is used for each call to Modular Messaging.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	100	0
Maximum Concurrently Registered IP Stations:	100	5
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:	10	0
Maximum Video Capable H.323 Stations:	0	0
Maximum Video Capable IP Softphones:	0	0
Maximum Administered SIP Trunks:	100	40
Maximum Number of DS1 Boards with Echo Cancellation:	0	0
Maximum TN2501 VAL Boards:	1	0
Maximum G250/G350/G700 VAL Sources:	5	0
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
(NOTE: You must logoff & login to effect the permission changes.)		

Figure 2: System Parameters Customer Options Form – Page 2

display system-parameters customer-options	Page 3 of 11
OPTIONAL FEATURES	
Abbreviated Dialing Enhanced List? n	Audible Message Waiting? n
Access Security Gateway (ASG)? n	Authorization Codes? n
Analog Trunk Incoming Call ID? n	Backup Cluster Automatic Takeover? n
A/D Grp/Sys List Dialing Start at 01? n	CAS Branch? n
Answer Supervision by Call Classifier? n	CAS Main? n
ARS? y	Change COR by FAC? n
ARS/AAR Partitioning? y	Computer Telephony Adjunct Links? n
ARS/AAR Dialing without FAC? n	Cvg Of Calls Redirected Off-net? n
ASAI Link Core Capabilities? n	DCS (Basic)? n
ASAI Link Plus Capabilities? n	DCS Call Coverage? n
Async. Transfer Mode (ATM) PNC? n	DCS with Rerouting? n
Async. Transfer Mode (ATM) Trunking? n	Digital Loss Plan Modification? n
ATM WAN Spare Processor? n	DS1 MSP? y
ATMS? n	DS1 Echo Cancellation? n
Attendant Vectoring? n	
(NOTE: You must logoff & login to effect the permission changes.)	

Figure 3: System Parameters Customer Options Form – Page 3

display system-parameters customer-options	Page 4 of 11
OPTIONAL FEATURES	
Emergency Access to Attendant? y	IP Stations? y
Enable 'dadmin' Login? y	Internet Protocol (IP) PNC? n
Enhanced Conferencing? y	ISDN Feature Plus? y
Enhanced EC500? y	ISDN Network Call Redirection? y
Enterprise Survivable Server? n	ISDN-BRI Trunks? y
Enterprise Wide Licensing? n	ISDN-PRI? y
ESS Administration? n	Local Survivable Processor? n
Extended Cvg/Fwd Admin? y	Malicious Call Trace? n
External Device Alarm Admin? n	Media Encryption Over IP? y
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n
Flexible Billing? n	
Forced Entry of Account Codes? n	Multifrequency Signaling? y
Global Call Classification? n	Multimedia Appl. Server Interface (MASI)? n
Hospitality (Basic)? y	Multimedia Call Handling (Basic)? n
Hospitality (G3V3 Enhancements)? n	Multimedia Call Handling (Enhanced)? n
IP Trunks? y	
IP Attendant Consoles? n	
(NOTE: You must logoff & login to effect the permission changes.)	

Figure 4: System Parameters Customer Options Form – Page 4

display system-parameters customer-options		Page 5 of 11
OPTIONAL FEATURES		
Multinational Locations? y	Station and Trunk MSP? y	
Multiple Level Precedence & Preemption? n	Station as Virtual Extension? n	
Multiple Locations? y		
Personal Station Access (PSA)? n	System Management Data Transfer? n	
Posted Messages? y	Tenant Partitioning? n	
PNC Duplication? n	Terminal Trans. Init. (TTI)? y	
Port Network Support? y	Time of Day Routing? y	
	Uniform Dialing Plan? y	
Processor and System MSP? y	Usage Allocation Enhancements? y	
Private Networking? y	TN2501 VAL Maximum Capacity? y	
Processor Ethernet? y	Wideband Switching? n	
	Wireless? n	
Remote Office? n		
Restrict Call Forward Off Net? y		
Secondary Data Module? y		
(NOTE: You must logoff & login to effect the permission changes.)		

Figure 5: System Parameters Customer Options Form – Page 5

4.2. Verify Features

Run the **change cos** command and ensure that the **Trk-to-Trk Transfer Override** field is set to 'y' for the Class of Service (COS) used by the call coverage path hunt group. By default all hunt groups use COS '1'. Some Avaya Modular Messaging features such as 'find-me' require the Avaya Communication Manager Trunk-to-Trunk transfer feature to be enabled.

With **Trk-to-Trk Transfer Override** set to 'y', this will override any system and/or COR-to-COR calling party restrictions that would otherwise prohibit the trunk-to-trunk transfer operation for users with this COS.

With Trunk-to-Trunk transfer (or Trunk-to-Trunk transfer override) enabled, the Avaya Modular Messaging attendant (or any other user) can connect an incoming trunk call to an outgoing trunk. Use this COS with caution, as the ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

change cos	Page 1 of 2															
	CLASS OF SERVICE															
	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Auto Callback	n	y	y	n	y	n	y	n	y	n	y	n	y	n	y	n
Call Fwd-All Calls	n	y	n	y	y	n	n	y	y	n	n	y	y	n	n	y
Data Privacy	n	y	n	n	n	y	y	y	y	n	n	n	n	y	y	y
Priority Calling	n	y	n	n	n	n	n	n	n	y	y	y	y	y	y	y
Console Permissions	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Off-hook Alert	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Client Room	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Restrict Call Fwd-Off Net	y	y	y	y	y	y	y	y	y	y	y	y	y	y	y	y
Call Forwarding Busy/DA	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Personal Station Access (PSA)	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding All	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding B/DA	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Trk-to-Trk Transfer Override	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n
QSIG Call Offer Originations	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Contact Closure Activation	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n

Figure 6: Class of Service Form – Page 1

4.3. Configure AAR Feature Access Code

The code for private network access must be defined. Use the command **change feature-access-codes** to assign this value. In this example configuration, the code was set to 8.

change feature-access-codes	Page 1 of 7														
	FEATURE ACCESS CODE (FAC)														
Abbreviated Dialing List1 Access Code:	*01														
Abbreviated Dialing List2 Access Code:	*02														
Abbreviated Dialing List3 Access Code:	*03														
Abbreviated Dial - Prgm Group List Access Code:	*04														
Announcement Access Code:	*05														
Answer Back Access Code:	#06														
Auto Alternate Routing (AAR) Access Code: 8															
Auto Route Selection (ARS) - Access Code 1:	9														
Automatic Callback Activation:	*09														
Call Forwarding Activation Busy/DA: #11	All: *10														
Call Park Access Code:	*06														
Call Pickup Access Code:	*07														
CAS Remote Hold/Answer Hold-Unhold Access Code:															
CDR Account Code Access Code:	*11														
Change COR Access Code:															
Change Coverage Access Code:															
Contact Closure Open Code:															
Contact Closure Pulse Code:															

Figure 7: Feature Access Codes Form – Page 1

4.4. Assign Local Node Number

In the dial plan, assign the local node number if not already assigned. This is accomplished by using the command **change dialplan parameters**.

```
change dialplan parameters
                                DIAL PLAN PARAMETERS

                                Local Node Number: 1
                                ETA Node Number:
                                ETA Routing Pattern:
                                UDP Extension Search Order: local-extensions-first
                                6-Digit Extension Display Format: xx.xx.xx
                                7-Digit Extension Display Format: xxx-xxxx
                                AAR/ARS Internal Call Prefix:
                                AAR/ARS Internal Call Total Length:
```

Figure 8: Dial Plan Parameters Form

4.5. Administer IP Network Region and Codec Set

Define the IP Network Region to be used for Avaya Modular Messaging and other devices in the network. This is accomplished using the command **change ip-network-region X**, where **X** is an available number for the new region. The UDP Port range must include 7000-7900 for Avaya Modular Messaging. Make sure the local domain for the SIP network is set to the SES domain name, i.e., “trade.com”. This domain name will be used later to configure the signaling group between Avaya Communication Manager and Avaya SIP Enablement Services server in section 4.6.

```
change ip-network-region 1
                                IP NETWORK REGION
                                Page 1 of 19

                                Region: 1
                                Location: 1      Authoritative Domain: trade.com
                                Name: Location 1
                                MEDIA PARAMETERS
                                Codec Set: 1      Intra-region IP-IP Direct Audio: yes
                                UDP Port Min: 2048 Inter-region IP-IP Direct Audio: yes
                                UDP Port Max: 65531 IP Audio Hairpinning? y
                                DIFFSERV/TOS PARAMETERS
                                Call Control PHB Value: 46 RTCP Reporting Enabled? y
                                Audio PHB Value: 46 RTCP MONITOR SERVER PARAMETERS
                                Video PHB Value: 26 Use Default Server Parameters? y
                                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
                                H.323 Link Bounce Recovery? y
                                Idle Traffic Interval (sec): 20
                                Keep-Alive Interval (sec): 5
                                Keep-Alive Count: 5
                                RSVP Enabled? n
```

Figure 9: IP Network Region Form – Page 1

On page 3, define the IP Network Regions that can communicate with each other and which audio codec set the regions will use. In the sample configuration described in these Application Notes, all devices in the Main Location were configured with IP Network Region 1 (including Avaya Modular Messaging) and all devices in the customer premises were configured with IP Network Region 2. IP codec set 1 was used for intra-region traffic and IP codec set 2 was used for inter-region traffic. Repeat the IP Network Region configuration shown here for IP Network Region 2 and 3.

```
change ip-network-region 1                                     Page 3 of 19
```

Inter Network Region Connection Management

src rgn	dst rgn	codec set	direct WAN	WAN-BW-limits	Intervening-regions	Dynamic CAC Gateway	IGAR
1	1	1					
1	2	2	y	:NoLimit			n
1	3	2	y	:NoLimit			n
1	4						

Figure 10: IP Network Region Form – Page 3

Administer the IP codec sets for intra-region and inter-region traffic using the command **change ip-codec-set X**, where **X** is an available number for the new codec set. In these Application Notes, codec set 1 was configured with the G.711MU audio codec. Avaya Modular Messaging currently supports the G.711MU and G.711A audio codecs; however only one audio codec can be active at any time.

Note: Frames per packet should be set to 2 and packet size (ms) should be set to 20.

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

IP Codec Set

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.711MU	n	2	20
2:			
3:			
4:			
5:			
6:			
7:			

Media Encryption

1: none

2:

3:

Figure 11: IP Codec Set “1” Form – Page 1

IP codec set **2** was configured with audio codec G.729AB for all inter-region traffic over the Wide Area Network (WAN).

```

change ip-codec-set 2                                     Page 1 of 2

                                IP Codec Set

Codec Set: 2

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt   Size (ms)
1: G.729AB      n          2        20
2:
3:
4:
5:
6:
7:

Media Encryption
1: none
2:
3:

```

Figure 12: IP Codec Set “2” Form – Page 1

Add the Avaya SES server IP address to the IP Node Names. Use the command **change node-names ip**.

```

change node-names ip                                     Page 1 of 1

                                IP NODE NAMES

Name      IP Address      Name      IP Address
A1_G350_LSP      7 .1 .1 .1      . . .
A1_LSP_Native    7 .1 .1 .2      . . .
A2_G650_LSP      6 .1 .1 .1      . . .
A2_LSP_Native    6 .1 .1 .2      . . .
CLAN-A           5 .1 .1 .4      . . .
CLAN-B           5 .1 .1 .5      . . .
MEDPRO           5 .1 .1 .6      . . .
TF-SES-SIP      5 .1 .1 .14    . . .

```

Figure 13: IP Node Names Form

4.6. Create the Signaling Group

Create the SIP signaling group between Avaya Communication Manager and Avaya SIP Enablement Services using the **add signaling-group X** command, where **X** is an available group number.

- Select *sip* as the **Group Type**.
- The **Transport Method** should be *tls*, which is auto-populated when the **Group Type** is *sip*.
- The name for one of the C-LAN circuit packs (*CLAN-A*) as it is shown in the **node-names ip** form should be entered for **Near-End Node Name**.
- The **Far-End Node Name** should be the name of the Avaya SES server as shown in the **node-names ip** form. For the **Far-end Network Region**, IP network region **1** was used.
- As stated in Section 4.5, all devices in the Main Location were configured in IP Network Region **1**.
- Enter the domain name of Avaya SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *trade.com*. This domain is specified in the Uniform Resource Identifier (URI) of the “SIP To Address” in the INVITE message. Configuring this field incorrectly may prevent calls from being successfully established to other SIP endpoints or to the PSTN. The domain name will also be used later to configure the Avaya SES server in section 5.3. The domain names in both Avaya Communication Manager and Avaya SES servers must match.
- The **Direct IP-IP Audio Connections** field was set to ‘*n*’ since calls between the endpoints and Avaya Modular Messaging were not shuffled.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833.
- The default values for the other fields may be used.

add signaling-group 1		Page 1 of 1	
SIGNALING GROUP			
Group Number: 1		Group Type: sip	
		Transport Method: tls	
Near-end Node Name: CLAN-A		Far-end Node Name: TF-SES-SIP	
Near-end Listen Port: 5061		Far-end Listen Port: 5061	
		Far-end Network Region: 1	
Far-end Domain: trade.com			
Bypass If IP Threshold Exceeded? n			
DTMF over IP: rtp-payload		Direct IP-IP Audio Connections? n	
		IP Audio Hairpinning? n	
Session Establishment Timer(min): 120			

Figure 14: Signaling Group Form

4.7. Create the Trunk Group

Create a new trunk group using the **add trunk-group X** command, where **X** is an available trunk group number.

- Enter a **Group Type** of *sip*. The **Group Type** field represents the call distribution method of this trunk.
- Enter a descriptive name for **Group Name**. In this configuration, the Avaya SES server name of *TF-SES-SIP* was used as the **Group Name**.
- Enter a trunk access code (**TAC**) that conforms to the configured dial plan. In this configuration, the **TAC** was set to *101*.
- Enter a **Service Type** of *tie*. The **Service Type** field defines the type of service provided. This trunk ties together Avaya Communication Manager and Avaya SES server.
- For **Signaling Group**, enter the number of the signaling group created in section 4.6. The **Signaling Group** number created in that section was *1*, and should be entered here.
- In this configuration, the MAS server was configured using an Avaya S3500 server. Avaya S3500 servers support up to 48 ports. This number should be used as the **Number of Members**, meaning *48* concurrent calls to Modular Messaging can be supported over this trunk group.
- On page 3, keep the default value for **Numbering Format**, which is *public*. The **Numbering Format** is used for identification purposes in the calling number and/or connected number information elements.

add trunk-group 1		Page 1 of 21	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: TF-SES-SIP	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? y		
Dial Access? n		Night Service:	
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 1	
		Number of Members: 48	

Figure 15: Trunk Group Form – Page 1

add trunk-group 1		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: internal		
		Maintenance Tests? y	
Numbering Format: public			
	Prepend '+' to Calling Number? n		
	Replace Unavailable Numbers? n		

Figure 16: Trunk Group Form – Page 3

4.8. Add Hunt Group

Configure a hunt group to be used for the Call Coverage Path assigned to Modular Messaging subscribers. This hunt group's extension number will be used as the Avaya Modular Messaging Access Number. This hunt group is configured with no members assigned to it.

add hunt-group 250		Page 1 of 60	
HUNT GROUP			
Group Number: 1	ACD? n		
Group Name: MM	Queue? n		
Group Extension: 30333	Vector? n		
Group Type: ucd-mia	Coverage Path:		
TN: 1	Night Service Destination:		
COR: 1	MM Early Answer? n		
Security Code:	Local Agent Preference? n		
ISDN/SIP Caller Display:			

Figure 17: Hunt Group Form – Page 1

On page 2, enter the **Voice Mail Number** and the **Voice Mail Handle** which will be used by Avaya SES in a later step. Also, in the **Routing Digit (e.g. AAR/ARS Access Code)** field of this form, enter the AAR Access Code as defined in Section 4.3. Enter *sip-adjunct* as the **Message Center**, indicating the type of messaging adjunct used for this hunt group. This value will also be used in section 4.14 when administering the message-waiting indicator on each phone.

add hunt-group 250		Page 2 of 60	
HUNT GROUP			
Message Center: sip-adjunct			
Voice Mail Number	Voice Mail Handle	Routing Digits	
		(e.g., AAR/ARS Access Code)	
30333	mm	8	

Figure 18: Hunt Group Form – Page 2

4.9. Create Coverage Path

Create a coverage path to be used for subscriber extensions. Assign the hunt group created to be the Coverage Point. The **Point1** value of *h250* is used to represent the hunt group number 250 created in the previous section. Use the command **add coverage path X** (in this case *X* is **1**). The default values for the other fields may be used.

add coverage path 1		Page 1 of 1	
COVERAGE PATH			
Coverage Path Number: 1		Hunt after Coverage? n	
Next Path Number:		Linkage	
COVERAGE CRITERIA			
Station/Group Status	Inside Call	Outside Call	
Active?	n	n	
Busy?	y	y	
Don't Answer?	y	y	Number of Rings: 2
All?	n	n	
DND/SAC/Goto Cover?	y	y	
Holiday Coverage?	n	n	
COVERAGE POINTS			
Terminate to Coverage Pts. with Bridged Appearances? n			
Point1: h250	Rng:	Point2:	Point3:
Point4:		Point5:	Point6:

Figure 19: Coverage Path Form

4.10. Create Route Pattern

For the trunk group created, define the route pattern. The route pattern consists of a list of trunk groups that can be used to route a call. The following screen shows route-pattern 1 that was configured. Use the command **change route-pattern 1**.

For this configuration, **Grp No 1** represents the trunk group number created back in section 4.7. The **FRL** field defines the facility restriction level for this route pattern. The value of 0 is the least restrictive. The default values for the other fields may be used.

change route-pattern 1															Page 1 of 3	
Pattern Number: 1 Pattern Name: sip-route																
SCCAN? n Secure SIP? n																
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted								DCS/	IXC
No			Mrk	Lmt	List	Del	Digits								QSIG	
															Intw	
1:	1	0													n	user
2:															n	user
3:															n	user
4:															n	user
5:															n	user
6:															n	user
BCC		VALUE		TSC	CA-TSC	ITC		BCIE	Service/Feature		PARM	No. Numbering		LAR		
0	1	2	3	4	W	Request						Dgts Format				
															Subaddress	
1:	y	y	y	y	y	n	n			rest				none		
2:	y	y	y	y	y	n	n			rest				none		
3:	y	y	y	y	y	n	n			rest				none		
4:	y	y	y	y	y	n	n			rest				none		
5:	y	y	y	y	y	n	n			rest				none		
6:	y	y	y	y	y	n	n			rest				none		

Figure 20: Route Pattern Form

4.11. Update AAR Digit Analysis Table

For the AAR Digit Analysis Table, create the dial string that will map calls to Avaya Modular Messaging via the route-pattern created in Section 4.10. Use the command **change aar analysis X** (in this case **X** is 30). The dialed string created in the AAR Digit Analysis table should contain a map to the pilot number for the Avaya Modular Messaging system. The value 30333 is the voice mail pilot number that's dialed to access Avaya Modular Messaging. Refer Section 5.7 for the voice mail pilot number.

change aar analysis 30										Page 1 of 2	
AAR DIGIT ANALYSIS TABLE											
										Percent Full: 1	
Dialed		Total		Route		Call		Node		ANI	
String		Min Max		Pattern		Type		Num		Reqd	
30333		5 5		1		aar				n	

Figure 21: AAR Digit Analysis Table Form

4.12. Set Route Pattern Location

Set the Route Pattern for the customer premise locations that leads to the Avaya SES server. This allows for the correct date and time information and trunk routing based on the IP network region. The **Proxy Sol. Rte. Pat.** value of 1 maps to the route pattern created in section 4.10.

change locations									
LOCATIONS									
ARS Prefix 1 Required For 10-Digit NANP Calls? y									
Loc. No.	Name	Timezone Offset	Rule	NPA	ARS FAC	Attd FAC	Loc. Pams.	Pre-fix	Proxy Sel. Rte. Pat.
1:	Main Location	+ 00:00	0				10		1
2:	Location 1	+ 00:00	0				11		1
3:	Location 2	+ 00:00	0				12		1

Figure 22: Locations Form

4.13. Define Numbering

This step is required for the Call Sender feature on Avaya Modular Messaging. This allows Avaya Communication Manager to send the calling party number along with the call information. This can be accomplished using the command **change public-unknown-numbering**. The dial plan used in this configuration included all 5-digit extensions that had pattern 3xxxx. Use the Trunk Group 1 created in Section 4.7. The **Ext Len** and **CPN Len** values should not be more than 7 digits.

change public-unknown-numbering 3									
NUMBERING - PUBLIC/UNKNOWN FORMAT									
Total									
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	CPN Len	Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len
5	3	1		5					
5	3	2		5					

Figure 23: Numbering – Public/Unknown Format Form

4.14. Subscriber Station Administration

For each subscriber extension, the **Coverage Path** and **Message Lamp Ext:** must be specified. Use the command **change station X** (in this case *X* is extension *30000*). An example extension, for an Avaya 4621SW IP Phone, is shown.

change station 30000		Page 1 of 4
STATION		
Extension: 30000	Lock Messages? n	BCC: 0
Type: 4621	Security Code: 1234	TN: 1
Port: S00001	Coverage Path 1: 1	COR: 1
Name: John Doe	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 30000	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Expansion Module? n	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? y	
	IP Video Softphone? n	
	Customizable Labels? Y	

Figure 24: Station Form – Page 1

Set the **MWI Served User Type** field to *sip-adjunct* to allow the station MWI lamp to work via the Avaya SES server.

change station 30000		Page 2 of 4
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? n	Restrict Last Appearance? y	
Active Station Ringing: single	Conf/Trans on Primary Appearance? n	
	EMU Login Allowed? n	
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed		
Multimedia Mode: enhanced		
MWI Served User Type: sip-adjunct	Display Client Redirection? n	
	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 30000	Always Use? n	IP Audio Hairpinning? Y

Figure 25: Station Form – Page 2

5. Configure the Avaya SES Server

This section covers the administration of Avaya SIP Enablement Services (SES) for integration with Avaya Modular Messaging. Avaya SES is configured via an internet browser using the administration web interface. During the software installation, the initial installation script is run on the Linux shell of the server to specify the IP network properties of the server along with other properties. For additional information on these installation tasks, refer to [3].

5.1. Log in to Avaya SES

Access the SES Administration web interface, by entering <http://<ip-addr>/admin> as the URL in an Internet browser, where *<ip-addr>* is the IP address of Avaya SIP Enablement Services server.

Log in with the appropriate credentials and then select the *Launch Administration Web Interface* link from the main screen as shown in **Figure 26**.

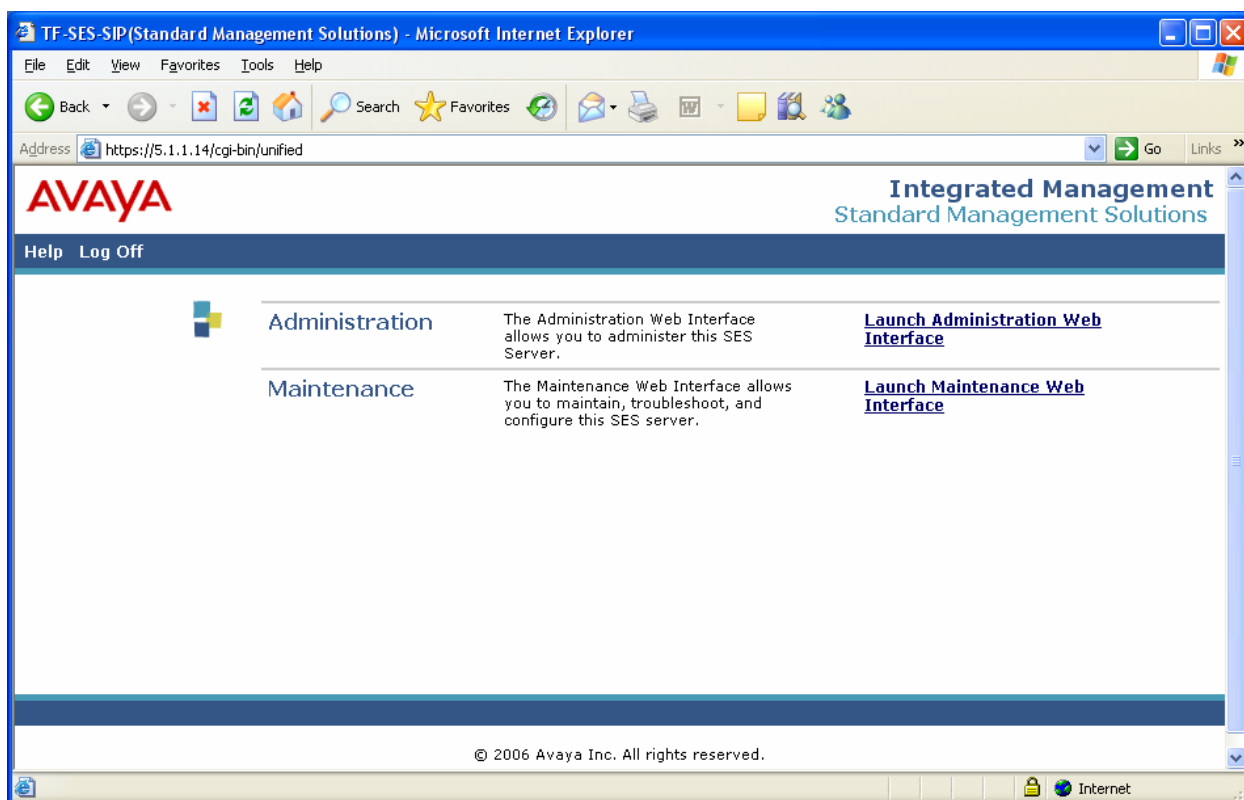


Figure 26: Avaya SES Main Screen

The SES administration home screen shown in **Figure 27** should be displayed.

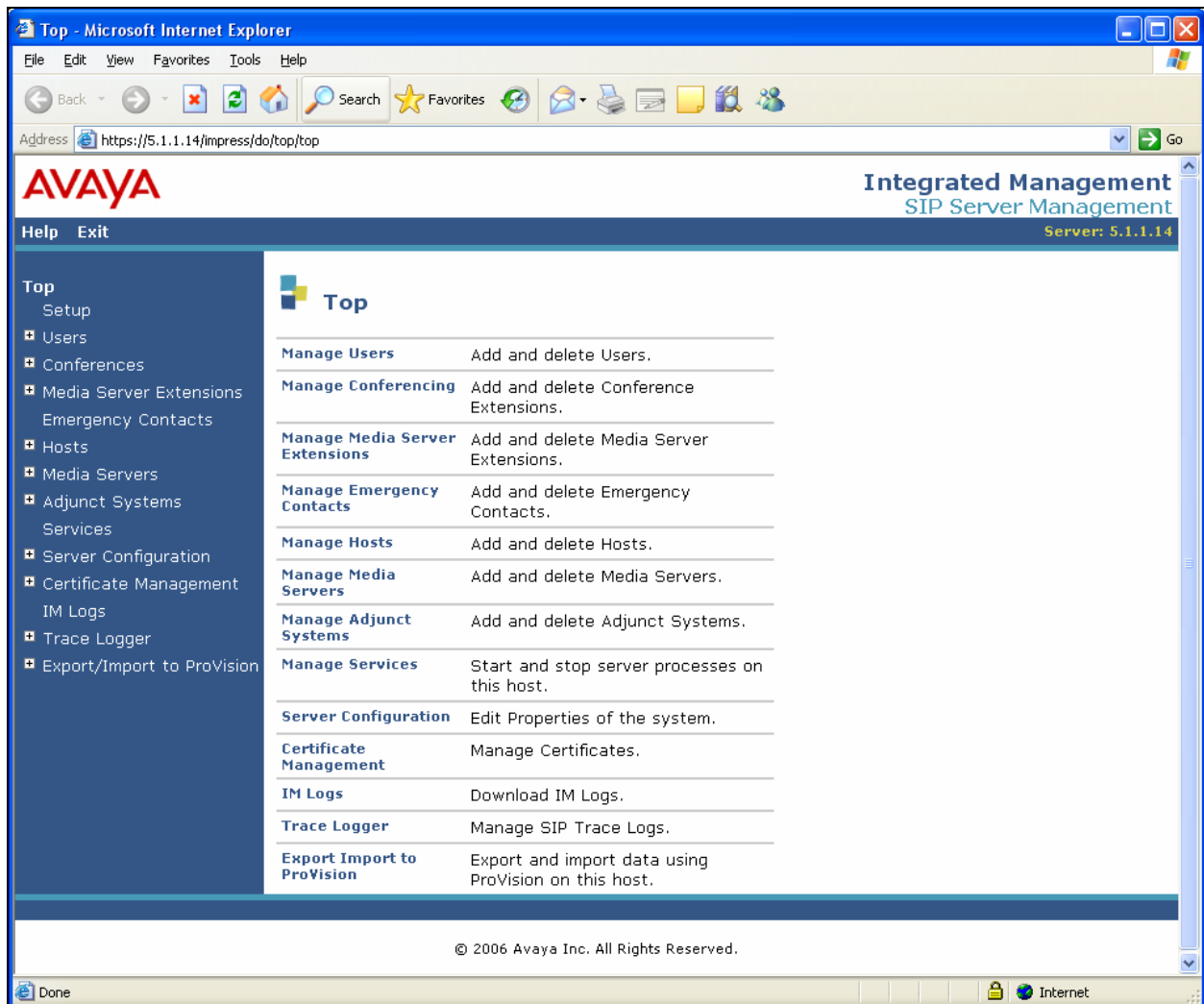


Figure 27: Avaya SES Administration Home Page

5.2. Saving Changes

After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left of any of the Avaya SES administration pages as shown in **Figure 28**. It is recommended that this be done after making each set of changes described in the following sub-sections.



Figure 28: Avaya SES Update Screen

5.3. Define System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the Avaya SES version and the network properties entered during the installation process.

In the **System Properties** screen,

- Enter the **SIP Domain** name assigned to Avaya SIP Enablement Services.
- Enter the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the SIP proxy server that is running the WebLM application and has the associated license file installed. This entry should always be **localhost** unless the WebLM server is not co-resident with this server.
- After configuring the **System Properties** screen, click the **Update** button.

Figure 29: System Properties

5.4. Configure Avaya SES Host Information

Create a host server entry for Avaya SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been added to the system. The **Edit Host** screen shown in **Figure 30** is accessible by clicking on the **Hosts** link in the left pane and then clicking on the **edit** option under the **Commands** section of the subsequent page that is displayed.

- Enter the **Logical IP** or **Logical Name** (shown in **Figure 29**) of this server in the **Host IP Address** field.
- Enter the **DB Password** that was specified while running the system installation.
- Configure the **Host Type** field. In this example, the host server was configured as a *home/edge* since no additional SES proxy servers exist in the enterprise network.
- The default values for the other fields may be used as shown in **Figure 30**.
- Click the **Update** button.

Figure 30: Edit Host

5.5. Add Avaya Communication Manager as a Media Server

Under the **Media Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site. This will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager.

In the **Add Media Server** screen, enter the following information:

- A descriptive name in the **Media Server Interface** field (e.g., s8720clan1).
- Select **TLS** (Transport Layer Security) for the **Link Type**. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for SIP trunking with Avaya Communication Manager.
- Enter the IP address of the C-LAN board administered for the signaling group for Avaya Communication Manager in Section 4.6 in the **SIP Trunk IP Address** field.
- (Optional) Enter the login/password information for the switch along with the switch name or IP (in the case of S87xx duplicated servers this should be the “active” shared IP-address). If any Avaya SES users have associated extensions on Avaya Communication Manager, Avaya SES obtains certain configuration information from Avaya Communication Manager over this interface.
- After completing the **Add Media Server** screen, click on the **Add** button.

Add Media Server Interface

Media Server Interface Name* 8710clanA

Host 5.1.1.14

SIP Trunk

SIP Trunk Link Type ☐ TCP ☒ TLS

SIP Trunk IP Address* 5.1.1.4

Media Server

Media Server Admin Address (see Help)

Media Server Admin Login

Media Server Admin Password

Media Server Admin Password Confirm

Fields marked * are required.

Add

© 2006 Avaya Inc. All Rights Reserved.

Figure 31: Add Media Server

5.6. Specify Address Maps to Media Servers

Incoming calls arriving at Avaya SES are routed to the appropriate Avaya Communication Manager for termination services. This routing is specified in a Media Server Address Map configured on Avaya SES.

This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map, and if there is a match, the call is routed to the designated Avaya Communication Manager. The URI usually takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Avaya Communication Managers supported by the Avaya SES server.

In these Application Notes, only incoming calls from Avaya Modular Messaging require a media server address map entry.

For Avaya Modular Messaging, the *user* portion of the SIP URI will contain the extensions on the Avaya S8710 Media Server. In this configuration, the pattern specification (without the double quotes) for extensions beginning with 30 is: `^sip:30 [0-9]*`. This matches URIs beginning with “sip:30” followed by any combination of digits.

To configure a **Media Server Address Map**:

- Select **Media Servers** in the left pane of the Administration web interface. This will display the **Manage Media Server Interfaces** screen.
- Click on **List Media Servers** to display all configured media servers.
- Click on the **Map** link associated with the appropriate media server to display the **List Media Server Address Map** screen.
- Click on the **Add Map In New Group** link. The screen shown in **Figure 32** is displayed. The **Host** field displays the name of the media server to which this map applies.
- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field.
- In case the contact information in this map is that of an endpoint (e.g., a SIP phone or a user on a media server running Communication Manager), then the **Replace URI** box should be checked for "yes." The box is checked by default, because the SIP proxy on an SES Server will overwrite the URI of the SIP request for these cases. If, however, you wish to configure this proxy to forward requests to another entity (i.e., another SIP proxy server) so that the other entity can resolve the contact and route the request, then uncheck the **Replace URI** box.

Click the **Add** button once the form is completed.

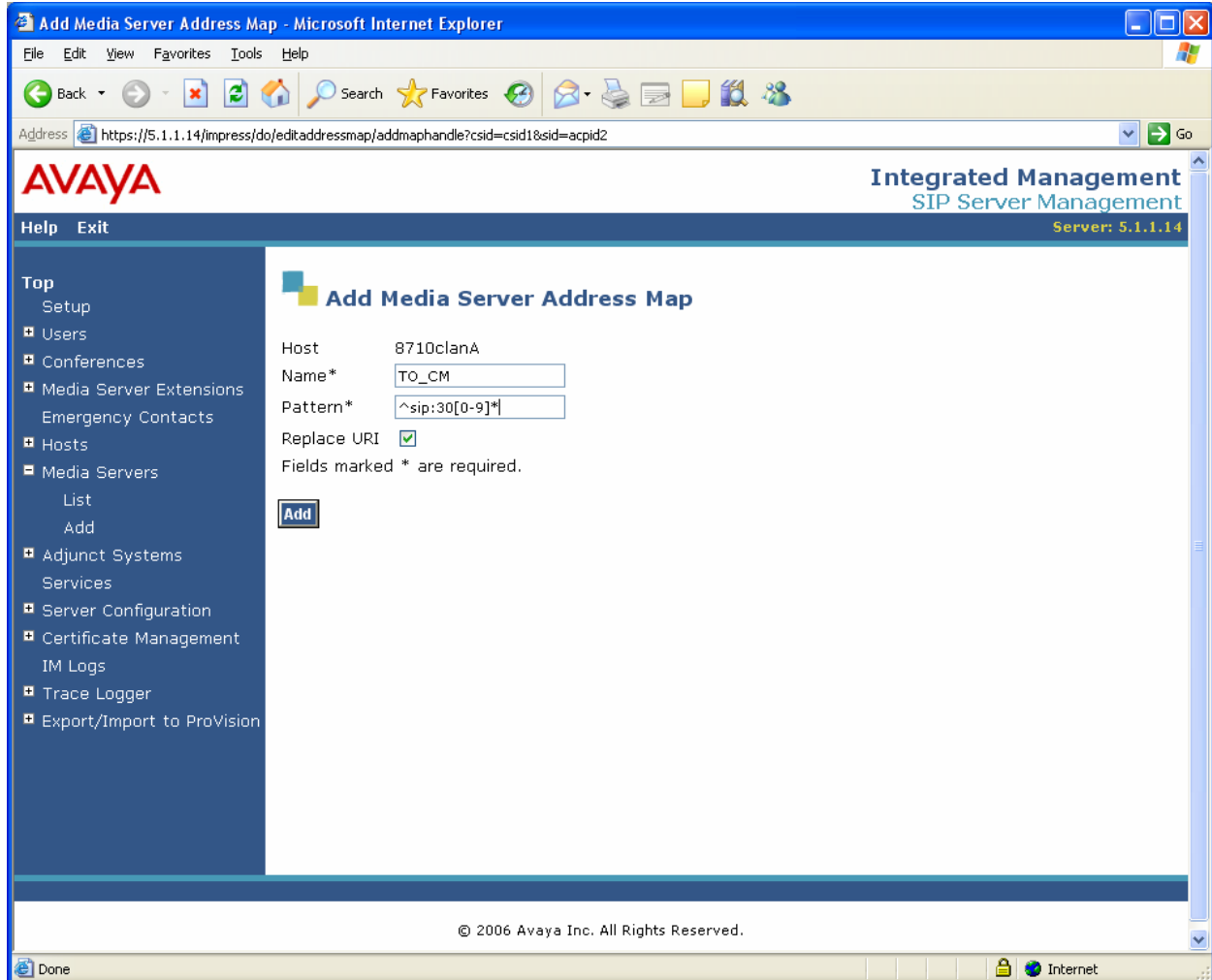


Figure 32: Media Server Address Map

After configuring the media server address map, the **List Media Server Address Map** screen appears as shown in **Figure 33**.

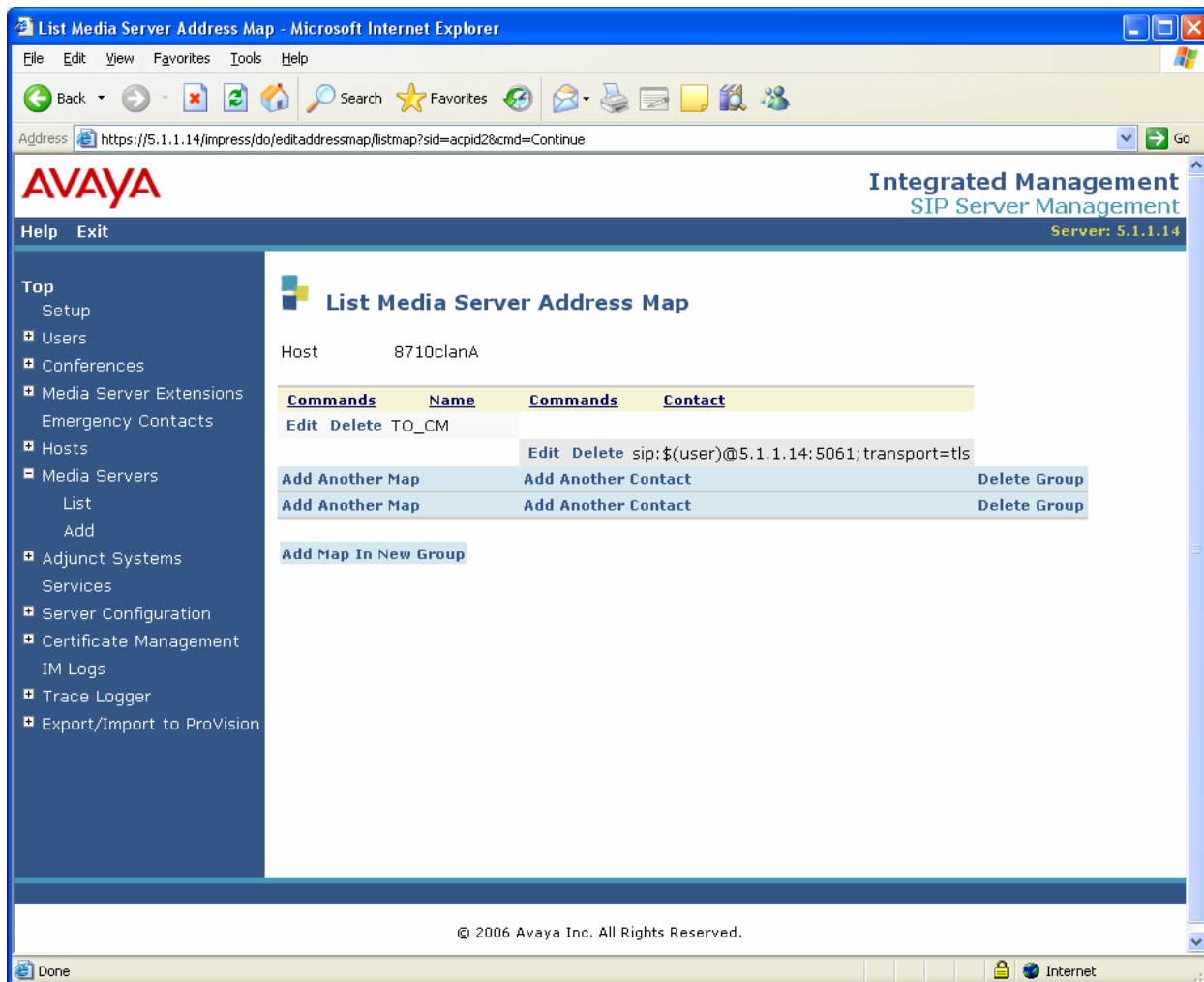


Figure 33: List Media Server Address Map

Note that after the first **Media Server Address Map** is added, the **Media Server Contact** is created automatically. For the **Media Server Address Map** added in **Figure 33**, the following contact was created: *sip:\${user}@5.1.1.14:5061;transport=tls*

The contact specifies the IP address of the C-LAN board on the Avaya SCC1 Media Gateway registered to the Avaya S8710 Media Server and the transport protocol used to send SIP signaling messages. The user in the original request URI is substituted for *\$(user)*.

5.7. Add Avaya Modular Messaging as an Adjunct System

Starting from the SIP Server Management page, perform the following actions:

- Expand **Adjunct Systems**.
- Click **Add**.
- In the **System Name** field, enter the desired name for the system. This **MUST** match the entry for the **Voice Mail Handle** defined on the Avaya Communication Manager pilot hunt group in Section 4.8.
- In the **Pilot Number** field, enter the Voice Mail Number specified in the Hunt Group of Avaya Communication Manager in Section 4.8.
- From the Host Name drop-down, select the name of the home proxy (Avaya SES) the MM system is using.
- Click the **Add** button and then the **Continue** button.

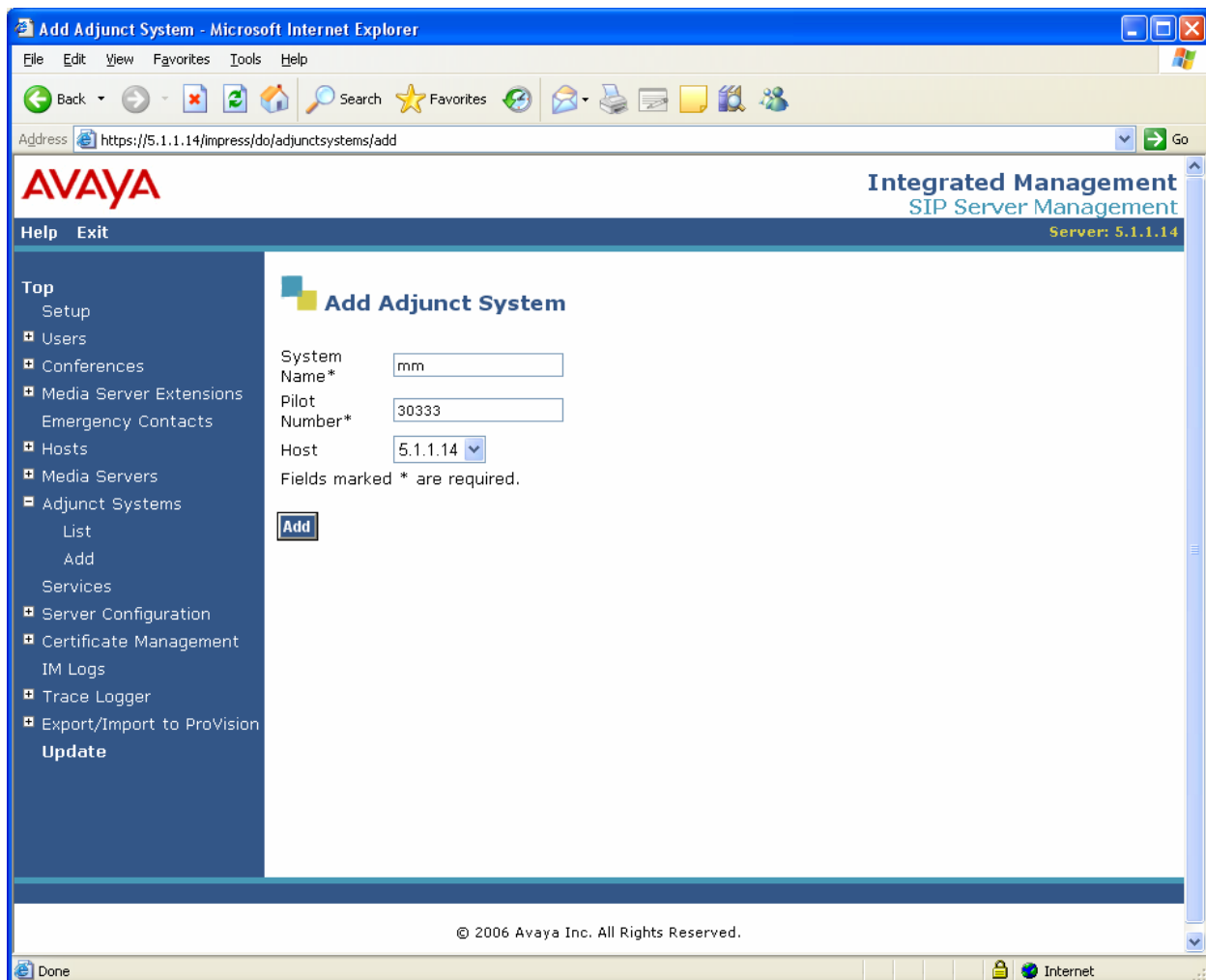


Figure 34: Adjunct Systems

The list of available Adjunct Systems is now displayed.

- Click on **List Adjunct Systems** for the system (just defined above)
- Click on **Add Another Adjunct System**.

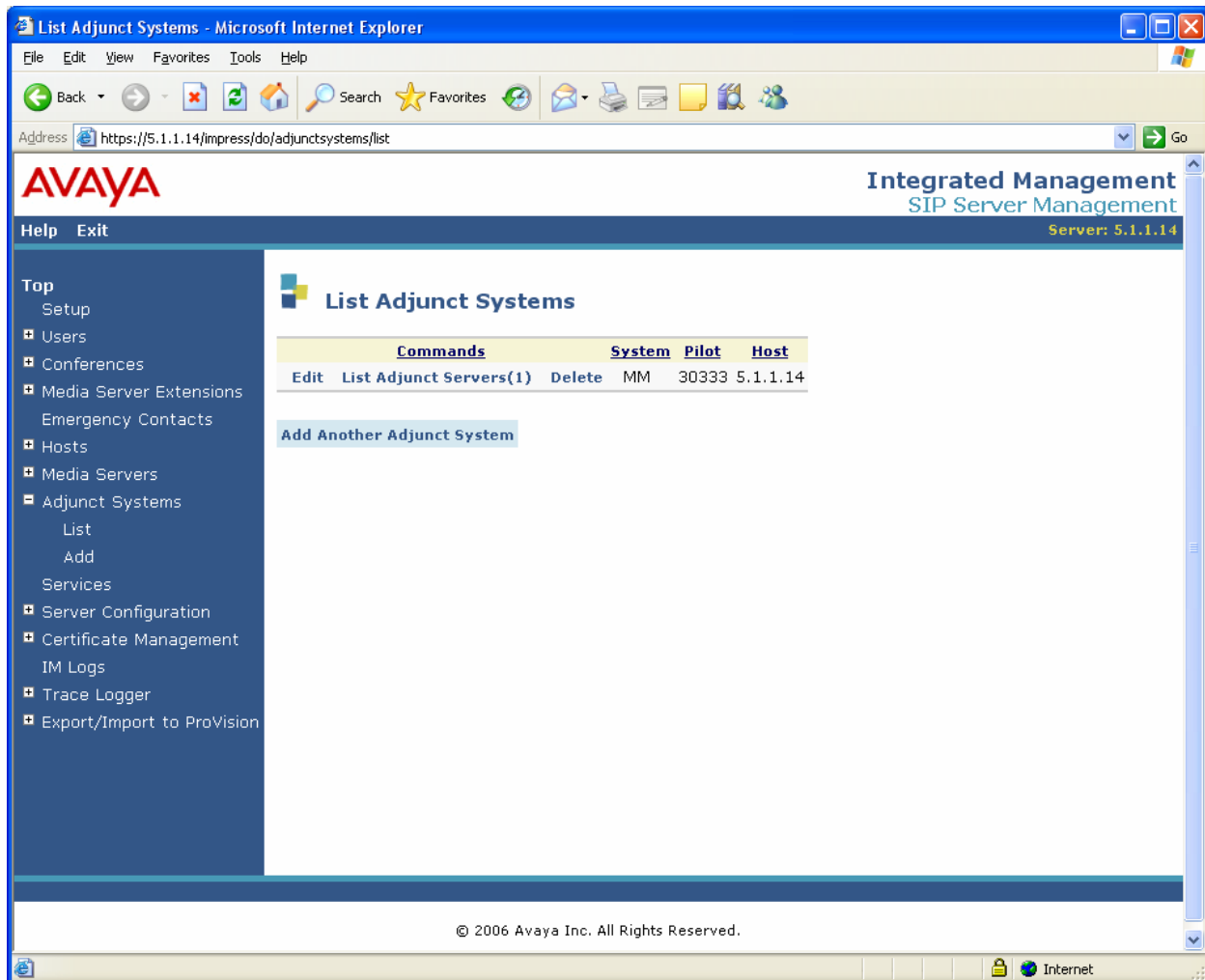


Figure 35: List Adjunct Servers

- In the **Server Name** field (below), enter the name of the MAS
- In the **Extension #** field, enter an unassigned extension number. The extension entered must be unique. Do not use the extension of any Avaya Communication Manager station or OPTIM extension.
- Select the **TLS** setting
- Enter the IP or Fully Qualified Domain Name (FQDN) of the MAS in the **Server IP Address** field.
- Click **Add** and **Continue**.
- Click **Update** when complete.

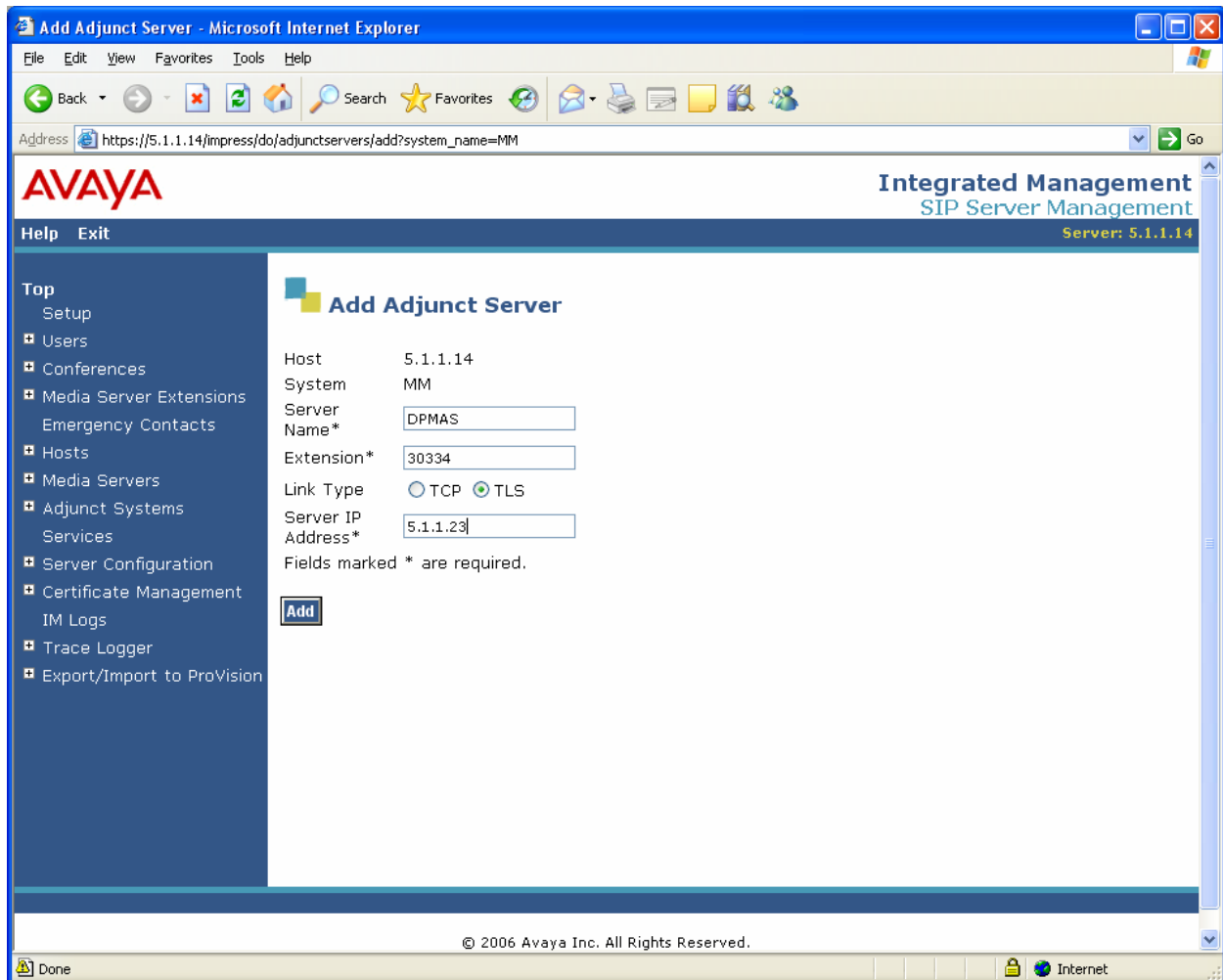
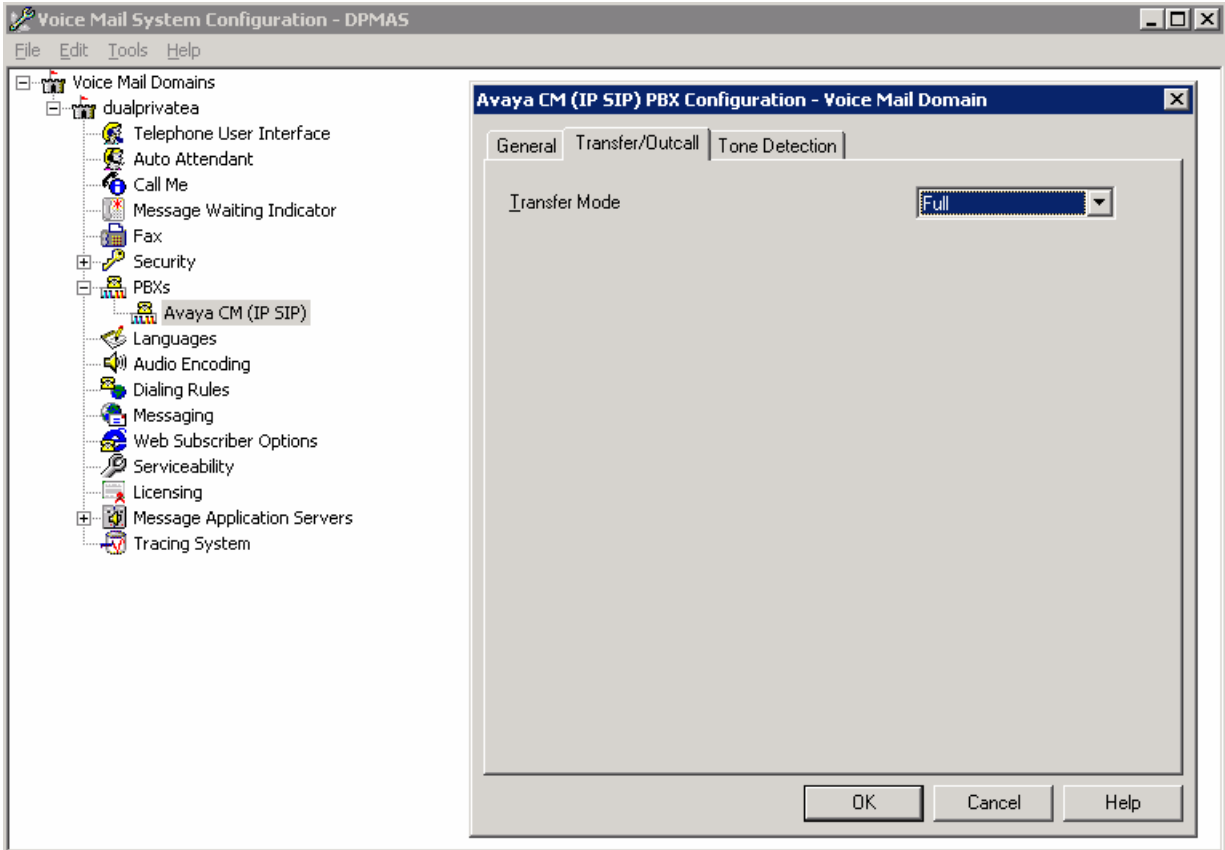


Figure 36: Adjunct Server

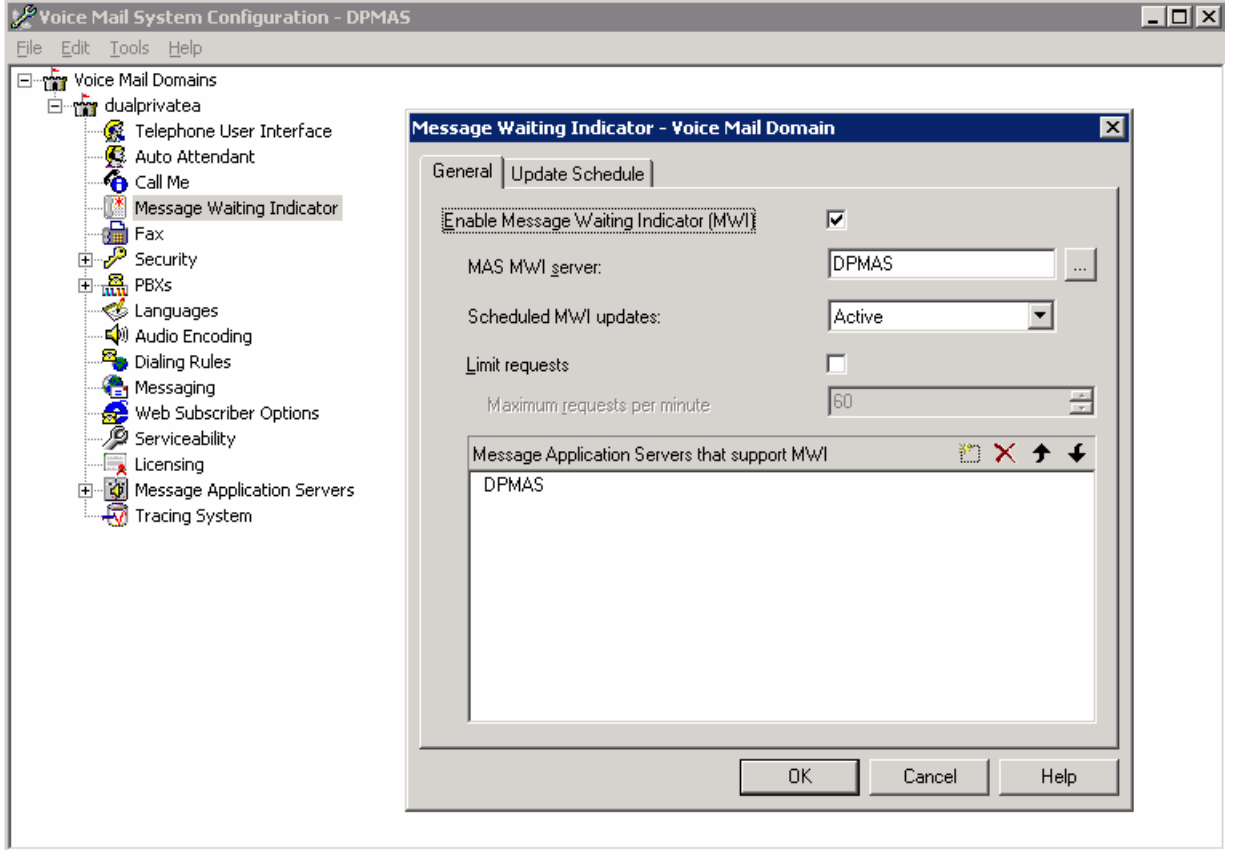
6. Configure the Message Application Server

There are a number of parameters that must be set in the Voice Mail Domain for Avaya Modular Messaging. These changes are made through the Voice Mail System Configuration application on the MAS server. This section assumes that the Message Application Server is installed and configured on Avaya S3500 servers. Refer to the Modular Messaging Software Message Application Server Guide [4] for more information.

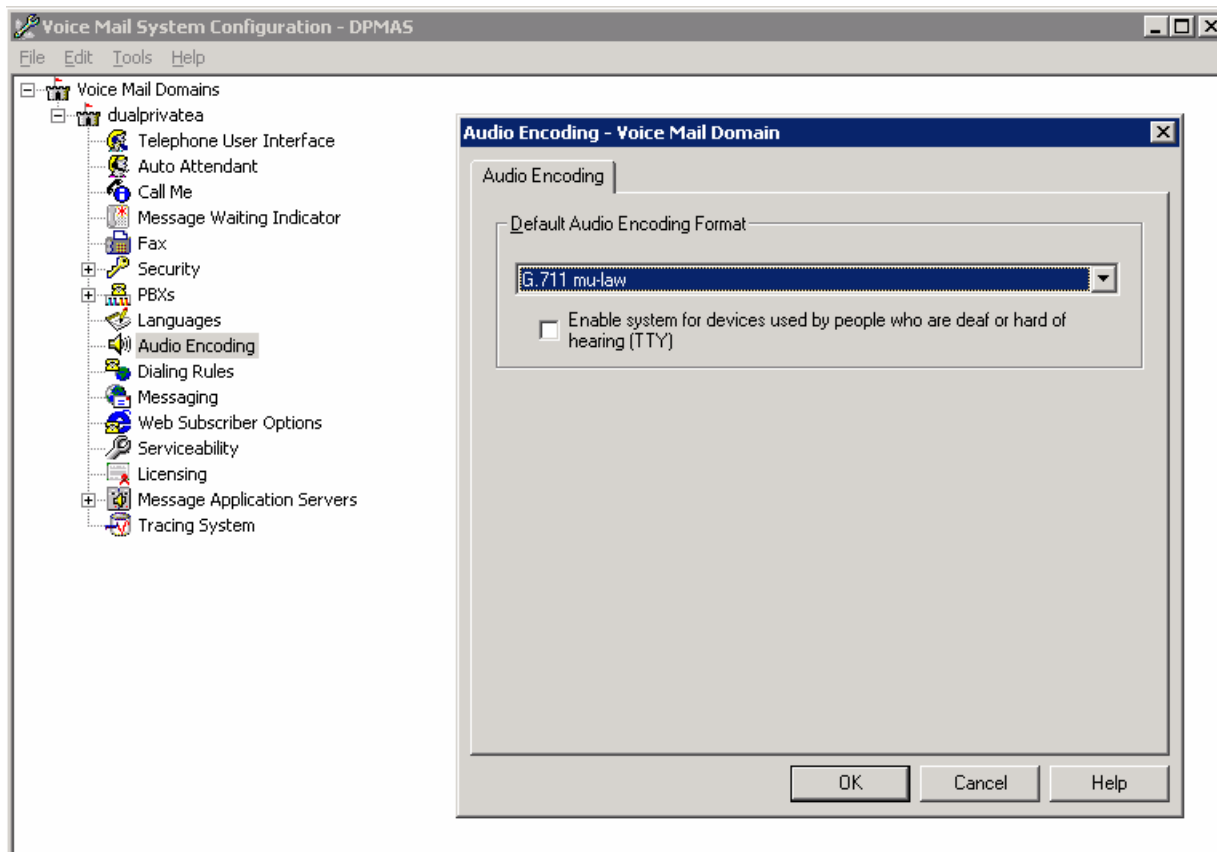
Step	Description
1.	Start the application by selecting Start → Programs → Avaya Modular Messaging → Voice Mail System Configuration .
2.	Select Voice Mail Domains and expand the Voice Mail Domain defined during the Installation of the MAS server. Select PBXs and double-click the Avaya CM (IP SIP) listed to modify properties. Select the Transfer/Outcall tab and set the Transfer Mode to Full . Administering transfers as 'Full' (Supervised transfer) prevents callers from being disconnected when calls are re-routed back to Avaya Modular Messaging. Transfers should only be administered as blind or partial when the transferred to numbers will not be re-routed to the message server.

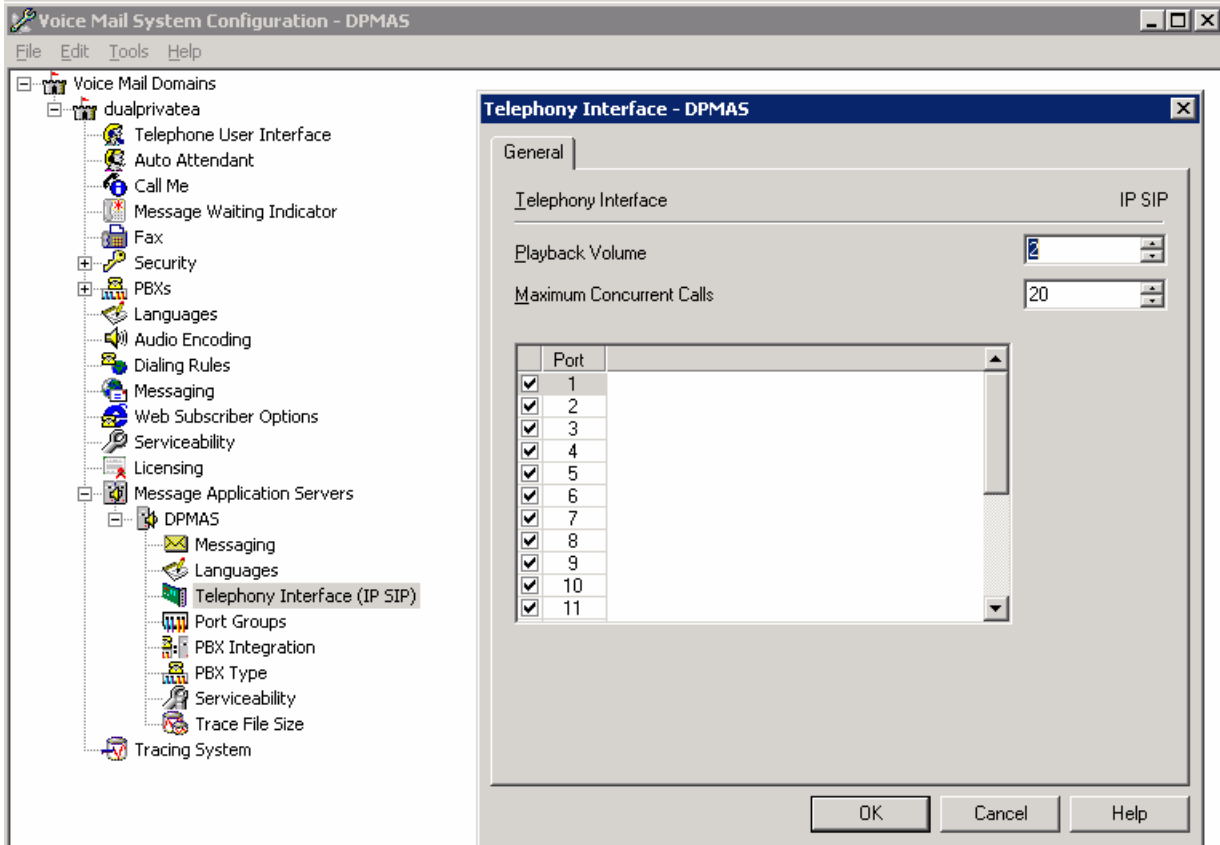


The screenshot displays the 'Voice Mail System Configuration - DPMAS' application window. On the left, a tree view shows the 'Voice Mail Domains' folder expanded, with 'PBXs' selected. Under 'PBXs', 'Avaya CM (IP SIP)' is highlighted. On the right, the 'Avaya CM (IP SIP) PBX Configuration - Voice Mail Domain' dialog box is open, showing the 'Transfer/Outcall' tab. The 'Transfer Mode' is set to 'Full'. The dialog box has 'OK', 'Cancel', and 'Help' buttons at the bottom.

Step	Description
3.	<p>Double-click the Message Waiting Indicator option and select Enable Message Waiting Indicator (MWI) option to enable the MWI feature with the options defined below. Click the OK button to save changes. The MAS will prompt to restart the services, but wait until the next step has been completed to restart.</p>
	 <p>The screenshot displays the 'Voice Mail System Configuration - DPMAS' window. On the left, a tree view shows the configuration hierarchy under 'Voice Mail Domains', with 'Message Waiting Indicator' selected. The main pane shows the 'Message Waiting Indicator - Voice Mail Domain' dialog box. This dialog has two tabs: 'General' and 'Update Schedule'. The 'General' tab is active, showing the following settings:</p> <ul style="list-style-type: none"> Enable Message Waiting Indicator (MWI): Checked (indicated by a checkmark in a box). MAS MWI server: DPMAS (text field with a browse button). Scheduled MWI updates: Active (dropdown menu). Limit requests: Unchecked (checkbox). Maximum requests per minute: 60 (spin box). Message Application Servers that support MWI: A list box containing 'DPMAS'. <p>At the bottom of the dialog are 'OK', 'Cancel', and 'Help' buttons.</p>

Step	Description
4.	<p>Double-click the Audio Encoding option, and select <i>G.711 mu-law</i> or <i>G.711 a-law</i> accordingly. This setting must match the Avaya Communication Manager IP Codec Set defined in Section 4.5.</p>



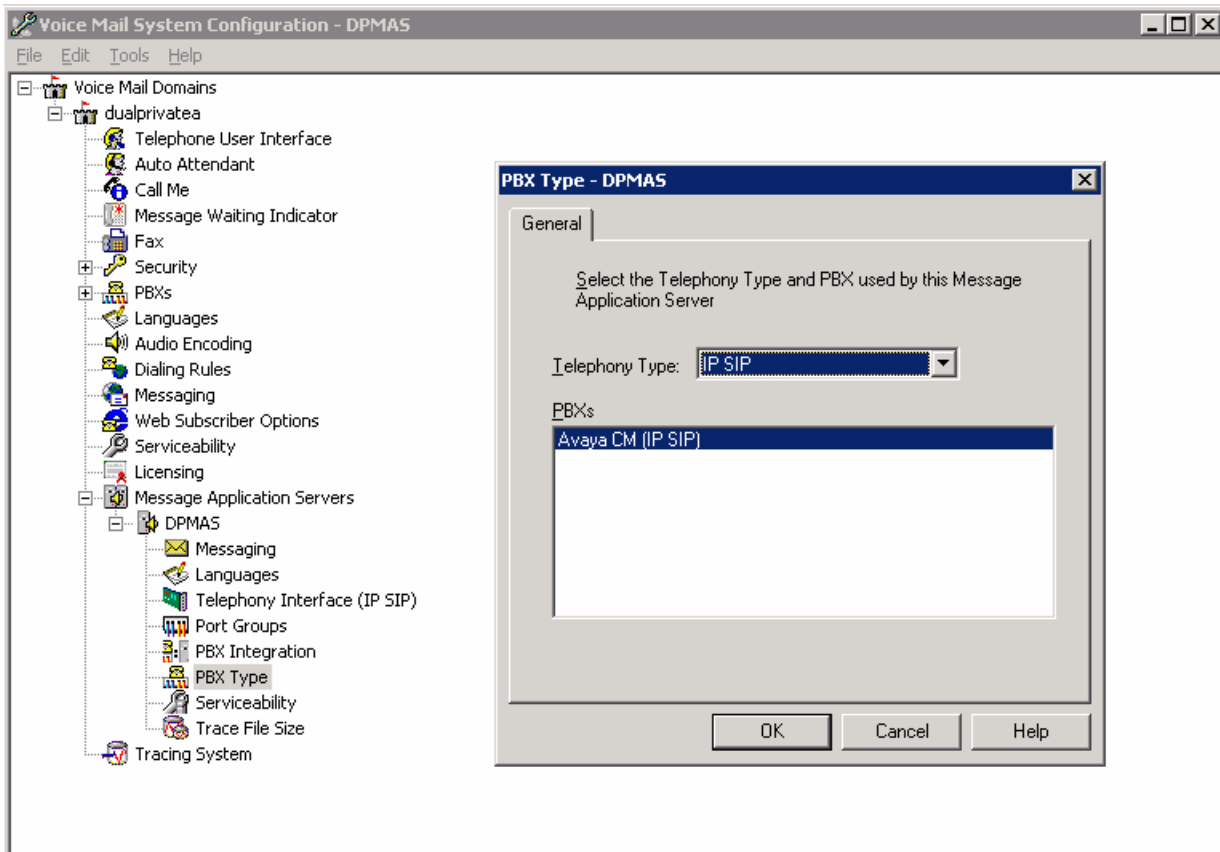
Step	Description
5.	<p>From the Voice Mail Domains, expand Message Application Servers and expand the MAS listed (i.e. <i>DPMAS</i>) to be modified. Double-click Telephony Interface (IP SIP). Set the Playback Volume to the desired level and check the port numbers to be used. Click OK to save changes.</p> 

Step	Description
6.	<p>Next double-click on PBX Integration to set the following options:</p> <ul style="list-style-type: none"> • Corporate IP Address = IP address assigned to the MAS. • SIP Domain = domain assigned in the IP Network Region form on Avaya Communication Manager. This was defined in section 4.5. • RTP Port Range = the minimum and maximum IP port numbers to specify for real-time protocol. 7000 to 7900 should be used. • Under MAS Listen Protocols, <ul style="list-style-type: none"> • TLS Port Number = Port number used for Transport Layer Security. This is usually 5061 • TCP Port Number = Port number used for Transaction Control Protocol. This is usually 5060. • Under SIP Proxies, for the Avaya SES in the network, select the checkbox and enter the IP Address/FQDN of the proxy server. The default port is <i>5061</i>. <p>Click OK to save changes and restart the MAS before continuing.</p>

The screenshot displays the 'Voice Mail System Configuration - DPMAS' window. On the left, a tree view shows various configuration categories, with 'PBX Integration' highlighted. The main area on the right is titled 'PBX Integration - DPMAS' and contains several sections:

- IP SIP**: A section header.
- MAS Details**: Fields for 'Corporate IP Address' (5.1.1.23), 'SIP Domain' (trade.com), 'RTP Port Range' (7000 - 7900), and 'Packet Size Bytes' (40).
- MAS Listen Protocols**: Fields for 'TLS Port Number' (5061) and 'TCP Port Number' (5060) with an 'Enable' checkbox.
- SIP Proxies**: A table with columns for Proxy, IP Address/FQDN, TCP/TLS, and Port. The first row is checked.

Proxy	IP Address/FQDN	TCP/TLS	Port
<input checked="" type="checkbox"/> 1	5.1.1.14	TCP	5061
<input type="checkbox"/> 2		TLS	5061
<input type="checkbox"/> 3		TLS	5061
<input type="checkbox"/> 4		TLS	5061

Step	Description
7.	<p>Double-click PBX Type and access the General tab. Select IP SIP for Telephony Type. Verify that Avaya CM (IP SIP) is selected under PBXs. Click the OK button to save any changes and restart the Message Application Server services to apply the changes.</p> 

7. Configure Subscribers on the Message Storage Server

There is nothing unique about the Avaya Modular Messaging subscriber configuration. Refer to the Message Storage Server (MSS) configuration guide [5] for information on how to create, modify, and/or delete subscribers.

For these Application Notes, a MSS server named DPMSS at IP address 5.1.1.25 was configured to add an Avaya Modular Messaging subscriber for extension 30000. Refer back to section 4.14 for details on how to create station 30000 in Avaya Communication Manager.

8. Verification Steps

These verification steps can be performed to check the Avaya Modular Messaging SIP integration.

1. Place a call to subscriber extension 30000.
2. Verify the call goes to coverage, is answered, and the correct subscriber greeting is played.
3. From SAT, perform a **list trace tac X**, where *X* is the trunk access code for the SIP trunk configured for Avaya SES to route the calls to Avaya Modular Messaging. Begin the trace as the call to the subscriber starts. The trace screen should show routing and caller information as the call is forwarded to Avaya Modular Messaging. The output on the **list trace** screen further verifies that the call was routed over the correct trunk.
4. Leave a message for the subscriber.
5. Verify that the MWI lamp is lit on the subscriber's phone.
6. From the subscriber extension, call the number for Voice Mail.
7. Verify the subscriber can log in and hear the message that was left in step #3.
8. Verify that the MWI lamp extinguishes on the subscriber's phone after the message has been retrieved.

9. Conclusion

These Application Notes describe the steps required to configure Avaya Modular Messaging SIP integration with an Avaya S8710 Media Server running Avaya Communication Manager via Avaya SES server.

10. Additional References

The following Avaya product documentation can be found at <http://support.avaya.com>.

- [1] Release 3.0 Messaging Application Server Administration Guide, Issue 1.
- [2] Avaya Modular Messaging Configuration Note 88010 (CN 88010) - "Avaya S8300/S8500/S8700/S8710 Session Initiated Protocol (SIP)".
- [3] Installing and Administering SIP Enablement Services Release 3.1, Issue 1.5.
- [4] Modular Messaging Software Message Application Server Guide Release 3, Issue 1.
- [5] Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 3 - Installation and Upgrades.

© 2006 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com