

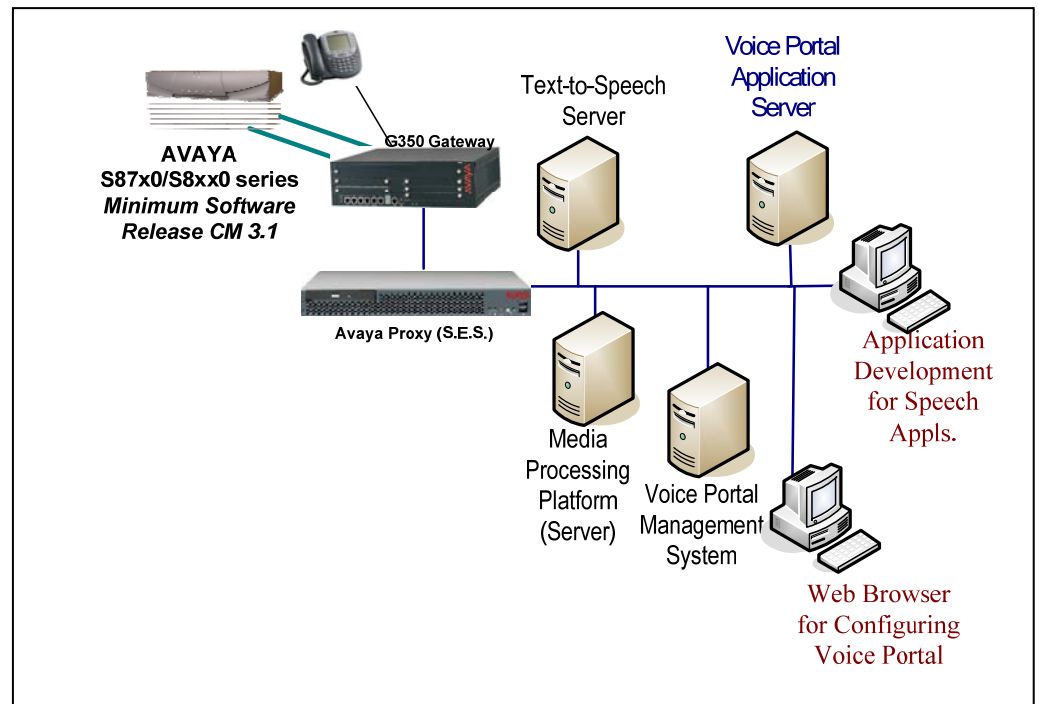


Avaya

Voice Portal

Configuration Note 3911 – Version B (12/08)

Avaya Voice Portal (Software application) - SIP Integration



Important: This Configuration Note specifies the minimum configuration required by voice portal. It is advisable to consult with your application designer to ensure you have a configuration to meet your specific

The Avaya SIP Enablement Server (SES) serves as the proxy allowing communication between the Avaya PBX, Gateway, and Voice Portal

1.0 METHOD OF INTEGRATION

The Session Initiation Protocol (SIP) provides connectivity with the Avaya PBX over a Local Area Network (LAN). IP-connected SIP trunks connect the Avaya SIP Enablement Server (SES) that serves as a Proxy allowing communication between the Avaya PBX, an Avaya Gateway, and the Voice Portal's Media Processing Platform (MPP).

PBX hardware requirements**2.0 Voice Portal Ordering INFORMATION**

For details on server hardware and software requirements please refer to:

- Avaya Voice Portal – Concepts and Planning
- Avaya Voice Portal – Installation and Configuration Guide

3.0 PBX HARDWARE REQUIREMENTS

Before performing the installation ensure the customer site has had a Avaya Network Assessment and the customer has implemented the recommendations.

S87x0/S8500/S84x0:

- TN2302/TN2602* IP Media Processor for voice processing
(Note: Should have latest firmware version)

*FOR FAX Support: TN2302 Firmware 111 minimum /
TN2602AP Firmware 24 minimum
- TN799D C-LAN for signaling

S8300:

- PROCR
- MM760/On-board VOIP(G250/G350)

3.1 PBX SOFTWARE REQUIREMENTS

Minimum Supported Software:

- Communication Manager 3.1 (Load 623 minimum)

Important: Before ordering, account teams should check with Avaya Services to determine if there are any applicable patches for customer specific configuration.

PBX/SES software requirements**3.2 SES SOFTWARE/HARDWARE REQUIREMENTS**

Minimum Supported Software:

- SIP Enablement Services 3.1 (Load 18 minimum) + Patch 1001

Hardware Required:

- SES Home Server and SES Edge Server or SES Home/Edge Server

FEATURES

3.3 CONNECTIVITY

- Ethernet LAN connectivity - TCP/IP

3.4 CUSTOMER-PROVIDED EQUIPMENT

- Wiring/equipment necessary to support the physical LAN (CAT 5 minimum)

4.0 SUPPORTED FEATURES

The Voice Portal system provides either full or partial automation of telephone transactions that would otherwise be performed by an operator, attendant, or contact center agent. These automated transactions are driven by speech applications where each speech application is designed and developed to satisfy specific customer needs.

- continued on next page -

PBX Configuration

5.0 SWITCH CONFIGURATION FOR IP INTEGRATION

The following tasks must be completed in the following order when programming the PBX to integrate. PBX programming is intended for certified PBX technicians/engineers.

- Verify customer option for SIP trunking
- Assign Local Node Number
- Administer C-LAN and IP Media Processor circuit packs (S8500/S87xx only)
- Assign IP node names and IP addresses to C-LAN, IP Media Processor (S8500/S87xx only)
- Define IP interfaces (S8500/S87xx only)
- Administer IP Network Regions
- Add SES Server to the node names
- Create SIP signaling group to the SES server
- Create a SIP trunk group associated to the SIP signaling group
- Create Route Pattern for SIP trunking
- Modify AAR/ARS Analysis Table
- Modify AAR Digit Conversion Table
- Modify ARS Digit Conversion Table
- Define Public Numbering Format

Note: The screens shown in this section are taken from an Avaya Site Administration (ASA) terminal. Some parameters may not appear on all software releases.

5.1 Verify customer options for SIP trunking

Ensure all required software features are enabled on the PBX. Access the System Parameters Customer Options form. Below is an example of the forms required for SIP integration, with the required features in **boldface**.

IMPORTANT: Only change the recommended fields.

NOTE:

These are license based changes.

Proper SIP licenses are required. Please refer to "SIP 3.1 Avaya Solution Designer Rules" to obtain proper codes.

```
display system-parameters customer-options                               Page 1 of 10
                                OPTIONAL FEATURES
G3 Version: V13
Location: 1                      RFA System ID (SID): 1
Platform: 6                      RFA Module ID (MID): 1

                                USED
Platform Maximum Ports: 44000 1105
Maximum Stations: 36000 1013
Maximum XMOBILE Stations: 0 0
Maximum Off-PBX Telephones - EC500: 100 0
Maximum Off-PBX Telephones - OPS: 100 28
Maximum Off-PBX Telephones - SCCAN: 100 0
```

(NOTE: You must logoff & login to effect the permission changes.)

NOTICE:

The screens in this Config Note are only for illustration purposes.

It is recommended that a qualified technician review the customer's configuration for accuracy.

```
display system-parameters customer-options                             page 2 of 10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
Maximum Administered H.323 Trunks: 100 0
Maximum Concurrently Registered IP Stations: 500 0
Maximum Administered Remote Office Trunks: 0 0
Maximum Concurrently Registered Remote Office Stations: 0 0
Maximum Concurrently Registered IP eCons: 0 0
Max Concur Registered Unauthenticated H.323 Stations: 0 0
Maximum Video Capable H.323 Stations: 0 0
Maximum Video Capable IP Softphones: 0 0
Maximum Administered SIP Trunks: 5000 70

Maximum Number of DS1 Boards with Echo Cancellation: 0 0
Maximum TN2501 VAL Boards: 1 0
Maximum G250/G350/G700 VAL Sources: 0 0
Maximum TN2602 VoIP Channels: 0 0

Maximum Number of Expanded Meet-me Conference Ports: 0 0
```

(NOTE: You must logoff & login to effect the permission changes.)

display system-parameters customer-options Page 3 of 10
 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)? y
ASAI Link Plus Capabilities? n	DCS Call Coverage? y
Async. Transfer Mode (ATM) PNC? n	DCS with Rerouting? y
Async. Transfer Mode (ATM) Trunking? y	Digital Loss Plan Modification? y
ATM WAN Spare Processor? n	DS1 MSP? n
ATMS? n	DS1 Echo Cancellation? n
Attendant Vectoring? n	

(NOTE: You must logoff & login to effect the permission changes.)

NOTICE:

The screens in this
 Config Note are only
 for illustration
 purposes.

It is recommended that
 a qualified technician
 review the customer's
 configuration for
 accuracy.

display system-parameters customer-options Page 4 of 10
 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? n	Malicious Call Trace? n
External Device Alarm Admin? n	Media Encryption Over IP? n
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)? y
Hospitality (G3V3 Enhancements)? n	Multimedia Call Handling (Enhanced)? y
IP Trunks? y	
IP Attendant Consoles? n	

```
display system-parameters customer-options                               Page 5 of 10
                                OPTIONAL FEATURES

      Multinational Locations? n      Station and Trunk MSP? n
Multiple Level Precedence & Preemption? n      Station as Virtual Extension? n
      Multiple Locations? n

      Personal Station Access (PSA)? n      System Management Data Transfer? n
      Posted Messages? n      Tenant Partitioning? n
      PNC Duplication? n      Terminal Trans. Init. (TTI)? y
      Port Network Support? y      Time of Day Routing? n
      Processor and System MSP? n      Usage Allocation Enhancements? y
      Private Networking? y      TN2501 VAL Maximum Capacity? y
      Processor Ethernet? n      Wideband Switching? n
      Remote Office? n      Wireless? n
Restrict Call Forward Off Net? y
      Secondary Data Module? y
```

- continued on next page -

- On the System-Parameters Features page, enable the following:

display system-parameters features Page 1 of 17

```

FEATURE-RELATED SYSTEM PARAMETERS
  Self Station Display Enabled? n
    Trunk-to-Trunk Transfer: all*
Automatic Callback - No Answer Timeout Interval (rings): 3
  Call Park Timeout Interval (minutes): 10
  Off-Premises Tone Detect Timeout Interval (seconds): 20
    AAR/ARS Dial Tone Required? y
      Music/Tone on Hold: none
    Music (or Silence) on Transferred Trunk Calls? no
      DID/Tie/ISDN/SIP Intercept Treatment: attd
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
  Automatic Circuit Assurance (ACA) Enabled? n

```

```

Abbreviated Dial Programming by Assigned Lists? n
Auto Abbreviated/Delayed Transition Interval (rings): 2
  Protocol for Caller ID Analog Terminals: Bellcore
Display Calling Number for Room to Room Caller ID Calls? n

```

*** NOTE:**

Trunk-to-trunk transfer
should be set to none
and COS used to
access this feature.

- Change features-access-codes and assign your private network access code, in this example we assigned 6

change feature-access-codes Page 1 of 8

```

FEATURE ACCESS CODE (FAC)
  Abbreviated Dialing List1 Access Code:
  Abbreviated Dialing List2 Access Code:
  Abbreviated Dialing List3 Access Code:
  Abbreviated Dial - Prgm Group List Access Code:
    Announcement Access Code:
    Answer Back Access Code:
    Attendant Access Code:
    Auto Alternate Routing (AAR) Access Code: 6
  Auto Route Selection (ARS) - Access Code 1: 9
    Automatic Callback Activation:
  Call Forwarding Activation Busy/DA: All:
    Call Park Access Code:
    Call Pickup Access Code:
  CAS Remote Hold/Answer Hold-Unhold Access Code:
    CDR Account Code Access Code:
    Change COR Access Code:
    Change Coverage Access Code:
    Contact Closure Open Code:
    Contact Closure Pulse Code:
    Access Code 2:
    Deactivation:
    Deactivation:
    Close Code:

```

IMPORTANT:

In some Avaya CM
releases AAR CODES
cannot start with #.
This was corrected in later
Avaya CM releases.

- Assign Local Node Number. Ensure PBX has an assigned Local Node Number. If there is no assigned number, enter 1.

```
display 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
```

- Administer C-LAN and IP Media Processor circuit packs (S8500/S87xx only)

```
display circuit-packs
CIRCUIT PACKS
Page 1 of 5

Cabinet: 1
Cabinet Layout: five-carrier
Carrier: A
Carrier Type: expansion-control

Slot Code  Sf Mode  Name
01:
02: TN799  D      CONTROL-LAN
03: TN2302      IP MEDIA PROCESSOR
04:
05:
06:
07:
08:
09: TN747  B      CO TRUNK
10:

Slot Code  Sf Mode  Name
11:
12:
13:
14: TN754  B      DIGITAL LINE
15: TN2181      DIGITAL LINE
16:
17:
18:
19:
```

- Assign IP Node names IP addresses to C-LAN, IP Media Processor (S8500/S8700 only). Enter the appropriate IP addresses for the installation.

```
display node-names ip
```

IP NODE NAMES

Name	IP Address
clan1	135.9 .84 .79
med1	135.9 .84 .82

- Define IP interfaces (S8500/S8700 only). Enter the appropriate Gateway address for the installation.

```
list ip-interface all
```

IP INTERFACES

ON	Type	Slot	Code	Sfx	Node Name/ IP-Address	Subnet Mask	Gateway Address	Net Rgn	VLAN
y	C-LAN	01A02	TN799	D	clan1 135.9.84.79	255.255.255.0	135.9.84.254	1	n
y	MEDPRO	01A03	TN2302		med1 135.9.84.82	255.255.255.0	135.9.84.254	1	n

- Define the Ethernet data module for the C-LAN board:

```
display data-module 8999
```

DATA MODULE

Data Extension: 8999	Name: clan1
Type: ethernet	
Port: 01A0217	
Link: 1	

```
Network uses 1's for Broadcast Addresses? y
```

Multiple Network Regions:

If you plan to use multiple network regions please read Consideration 8.5 in this Configuration Note.

IMPORTANT:

Avaya Media Encryption is NOT supported. These fields only appear if Media Encryption is enabled in system-parameter customer-options. Leave them set to none (default)

- Define the IP Codec Set and ensure G.711 is added. You can use G.711 mu-law or G.711 a-law or have both entries in the set.

Note: Frames per packet should be set to 2 and packet (ms) size 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: G.711A	n	2	20
3:			
4:			
5:			
6:			
7:			

Media Encryption:

1: none
2: none
3: none

display ip-codec-set 1
Page 2 of 2

IP Codec Set

Allow Direct-IP Multimedia? n

	Mode	Redundancy
FAX	off	0
Modem	off	0
TDD/TTY	US	3
Clear-channel	n	0

FAX: Starting with Voice Portal 4.1 a fax tone detection feature is available. For this feature to work, you must configure the PBX correctly.

Note: Turn **off** the special fax handling as shown here in this codec set. This allows the fax to be handled as an ordinary voice call. With a codec set that uses G.711, this setting is required to send faxes to non-Avaya systems that do not support T.38 fax.

- Define IP Network Regions. In this example network region '1' is selected. Define the local domain for the SIP network in this example "avaya.com" is used.

Authoritative Domain:

The name entered here (*our example shows avaya.com*) must match what is used on the Signaling Group or a call from the Voice Portal Server to the CM will not authenticate.

```
display ip-network-region 1                               Page 1 of 19
IP NETWORK REGION
Region: 1
Location: 1      Authoritative Domain: avaya.com
Name:
MEDIA PARAMETERS
Codec Set: 1      Intra-region IP-IP Direct Audio: yes
                  Inter-region IP-IP Direct Audio: yes
                  IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 34
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 7
Audio 802.1p Priority: 6
H.323 IP ENDPOINTS
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
RTCP Reporting Enabled? y
RTCP MONITOR SERVER PARAMETERS
Use Default Server Parameters? y
AUDIO RESOURCE RESERVATION PARAMETERS
RSVP Enabled? n
```

- Add the SIP Proxy to the IP Nodes Names. Enter the IP address assigned to the Home SES or Home/Edge SES.

```
display node-names ip
IP NODE NAMES
Name      IP Address
clan1    135.9 .84 .79
med1     135.9 .84 .82
sip-proxy 135.9 .84 .111
```

- Create the signaling group for SIP. The Near-end Node Name is the name assigned to the C-LAN above. The Far-end Node Name is the name assigned to the SES Server above. For this example signal group 8 was selected using TLS transport with port 5061.

Far-end Domain: The name entered here (our example shows **avaya.com**) must match what's in the Author Domain field on the NR or inbound calls (SIP messages) to CM from the Voice Portal Server may not work.

To support *shuffling*, set *IP-IP Audio Connections* and *IP Audio Hairpinning* to "Y"

```
change signaling-group 8                               Page 1 of 1
                                     SIGNALING GROUP
Group Number: 8                                     Group Type: sip
                                           Transport Method: tls

Near-end Node Name: clan1                         Far-end Node Name: sip-proxy
Near-end Listen Port: 5061                     Far-end Listen Port: 5061
                                           Far-end Network Region:

Far-end Domain: avaya.com

                                           Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                         Direct IP-IP Audio Connections? y
                                           IP Audio Hairpinning? y
Session Establishment Timer(min): 120
```

- Create the trunk group for SIP.

Only 1 Trunk Group needs to be programmed between the PBX and SES. This Trunk Group can be used by all applications. You will need to confirm how many members it has.

Additionally, you may want to look at COR on the PBX to prevent inbound/outbound calls on that trunk group as required.

Note: The COR controls only calls from the VP in the event outcalling or follow-me is used. If different COR permissions are needed for different applications multiple trunk groups would be used.

```
display trunk-group 7                               Page 1 of 20
                                     TRUNK GROUP
Group Number: 7                                     Group Type: sip
Group Name: to sip-proxy                         COR: 1       CDR Reports: y
Direction: two-way                             Outgoing Display? n   TN: 1       TAC: 107
Dial Access? n
Queue Length: 0
Service Type: tie                               Auth Code? n
                                           Night Service:

                                           Signaling Group: 8
                                           Number of Members: 40
```

When the format is set to **Public** this indicates that the CCITT Recommendation E.164 number plan is used, meaning that the Type of Number is formatted as national.

```
display trunk-group 7
TRUNK FEATURES
    ACA Assignment? n
    Measured: none
    Maintenance Tests? y
    Format: public
    Replace Unavailable Numbers? n
```

- continued on next page -

- Add Hunt Group. This hunt group's extension number is going to be used as the VOICE PORTAL Access Number. This hunt group is configured with no members assigned to it, and should be configured as follows:

A Hunt Group provides a dialable number (Pilot #) to reach a trunk group.

Note: The Hunt Group name in CM (shown here in our example as **voiceportal**) must match the name defined in Voice Portal exactly. Bear in mind Voice Portal only allows lower case characters.

```
change hunt-group 4                                     Page 1 of 60
                                                    HUNT GROUP

Group Number: 4                                         ACD? n
Group Name: voiceportal                               Queue? n
Group Extension: 4321                                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: mbr-name
```

Voice Mail Handle must match the adjunct System name as shown in the Add Adjunct System screen on page 21 in this CN. This field indicates the SIP Enablement Services (SES) handle that can receive voice mail.

Note: In our example on the right we show the *Voice Mail Handle* as **VoicePortalSystem**. If this is a name then the SES must have it entered as a name on the MAP. Alternately this Handle can be a number.

```
change hunt-group 15                                     Page 2 of 60
                                                    HUNT GROUP

Message Center: sip-adjunct

Voice Mail Number      Voice Mail Handle      Routing Digits
                       (e.g., AAR/ARS Access Code)
4321                 VoicePortalSystem          6
```

- Create a Route Pattern for the SIP trunk group created earlier. For this example route pattern 9 is used, with trunk group 7.

Note: *Ensure Secure SIP is set to n.* (Sets the call as TLS end-to-end.)

display route-pattern 9															Page 1 of 3										
										Pattern Number: 9		Pattern Name: siproute													
										SCCAN? n		Secure SIP? n													
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted								DCS/	IXC									
No			Mrk	Lmt	List	Del	Digits								QSIG										
							Dgts								Intw										
1:	7	0						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		BAND		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									

- Within the AAR Digit Analysis Table, create a dialed string that will map calls to the newly created Route Pattern. The dialed string created in the AAR Digit Analysis Table should contain a map to the Pilot Number for the VOICE PORTAL system. This is how calls will route to the trunk. Below is an example of an AAR dialed string in **boldface**.

display aar analysis 2						Page 1 of 2	
AAR DIGIT ANALYSIS TABLE							
						Percent Full: 1	
	Dialed String	Total		Route	Call	Node	ANI
		Min	Max	Pattern	Type	Num	Reqd
2		7	7	999	aar		n
3		5	5	2	aar		n
4		7	7	999	aar		n
5		5	5	5	aar		n
5		7	7	999	aar		n
6		5	5	2	aar		n
7		4	4	9	aar		n
8		4	4	2	aar		n
9		5	5	4	aar		n

ALTERNATE MEANS OF SETTING UP ROUTING –

Should you need more than 10 application IDs (aka hunt groups), one way to achieve this is to use the *uniform-dialplan* table on CM.

The same signaling and trunk group shown in this Configuration Note will work, however you will need to modify the AAR Digit Analysis Table and Route Pattern.

- Create a Route Pattern for the SIP trunk group created earlier. For this example route pattern 9 is used, with trunk group 7. If you choose to use a three digit AAR number, you will need to delete these digits in the route pattern you have going to the SES/MPP. In the example below, we configured trunk group 7 to route to our SES, and under No. Del Dgts, we delete 3 to remove the 007 that you're going to insert later.

Note: Ensure Secure SIP is set to n. (Sets the call as TLS end-to-end.)

display route-pattern 9

Page 1 of 3

Pattern Number: 9

Pattern Name: siproute

SCCAN? n

Secure SIP? n

Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC
No			Mrk	Lmt	List	Del	Digits	QSIG	
						Dgts		Intw	
1:	7	0				3		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	BAND	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

- continued on next page -

How does this Alternate configuration work?

If you dial **7100**, the UDP Table will see the number is 4 digits on length and then insert **006** in front of it, making the new number **0067100**.

The **0067100** number is then presented to the AAR Analysis Table and since it is a 7-digit number (Min/Max) that begins with **006** it knows it is destined for Route Pattern **9**.

When the number **0067100** is presented to Route Pattern **9**, the Route Pattern sees that this call is destined for Trunk Group **7** but before it goes there, the Route Pattern must delete 3 (leading) digits as you have administered it to do. The number that remains is **7100**.

- The example AAR Digit Analysis Table shown below, assumes a 3-digit AAR pattern of 006 and 4-digits extensions are being use. We use 7 as the Min and Max as it is the total number of digits derived from 4-digit extensions + a 3-digit AAR.
- Create a dialed string that will map calls to the newly created Route Pattern (*see previous page*). We chose a 3-digit number to represent the trunk group number. Since the trunk group going to the SES is 7, we use 007.

Note: What this does is circumvent the need to make an entry in the AAR Analysis table for “every” matching pattern going to the SES.

display aar analysis 2 Page 1 of 2

AAR DIGIT ANALYSIS TABLE							
Percent Full: 1							
	Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
006		7	7	9	aar		n
5		5	5	5	aar		n
5		7	7	999	aar		n
6		5	5	2	aar		n
8		4	4	2	aar		n
9		5	5	4	aar		n

In the example below we use a matching pattern of 71. This means that all extensions 7100-7199 will route to the SES (or MPP). Under Insert Digits we add 007, which is our AAR.

display uniform-dial plan 1 Page 1 of 2

UNIFORM DIAL PLAN TABLE						
Percent Full: 0						
Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
71	4	0	006	aar	n	
					n	
					n	
					n	
					n	
					n	
					n	
					n	

After you've done this, you'll need to add these extensions as Application ID's on the SES and on Voice Portal. You should then be able to call from CM to Voice Portal application with extensions 7100-7199.

- Modify the AAR Digit Conversion to ensure a Matching Pattern for all extensions the SES server will be using

Note: If matching patterns covers the Pilot number, you may get an error within CM and forwarding to voice portal will not work.

display aar digit-conversion 0								Page	1 of	2
AAR DIGIT CONVERSION TABLE								Percent Full: 0		
Matching Pattern	Min	Max	Del	Replacement String	Net	Conv	ANI	Req		
0	1	28	0		ars	y		n		
1	4	28	0		ars	y		n		
x11	3	3	0		ars	y		n		
8	4	4	0		ext	n		n		

- Modify the ARS Digit Conversion (if needed) to allow SES to dial and transfer to local PBX extensions. Ensure to administer a Matching Pattern for all extensions the SES server will be dialing.

display ars digit-conversion 0								Page	1 of	2
ARS DIGIT CONVERSION TABLE								Percent Full: 10		
Location: all										
Matching Pattern	Min	Max	Del	Replacement String	Net	Conv	ANI	Req		
2	4	4	0		ext	n		n		
3	4	4	0		ext	n		n		
5	4	4	0		ext	n		n		
7	4	4	0		ext	n		n		

- Set the route pattern for the switch location.

The **Proxy Selection Route Pattern** field identifies the routing pattern that is used to get to the proxy server. Basically, this route pattern points to the SIP trunk so that outbound calls over ISDN trunks will know where to send updated ISDN messages.

Example of use: When an ISDN "Disconnect" message needs to change to a SIP "Bye" message so it can be sent over the SIP trunk to drop that leg of the call.

display locations

LOCATIONS

ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc. No.	Name	Timezone Offset	Rule	NPA	Proxy Sel. Rte. Pat.
1:	Main	+ 00:00	0		9

- Define Public Numbering. Ensure to administer an entry to match each extension Voice Portal will be supporting. For this example extension 8XXX is used. For the trunk group use the same trunk group number created above (7 for example).

Note: No more than 7 digits should be sent, so administer with a blank *CPN Prefix*. *Ext Len* and *CPN Len* values should not be more than 7.

NUMBERING - PUBLIC/UNKNOWN FORMAT

Ext Len	Ext Code	Trk Grp(s)	Total CPN Prefix	CPN Len	Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len
4	8	7		4					

Note: The screens used in this section serve only as examples. The Names, IP addresses, and login information will be different than what you need for your install.

5.3 Configuring the SES Proxy Server

The following tasks must be completed to integrate the proxy server with the switch.

- Create a media server
- Add an address map to the media server consisting of a media server contact and a map entry.
- Create an adjunct system
- For each Voice Portal add one adjunct server to the adjunct system
- For each SIP phone administered on the PBX add a user with a media extension. (see [note](#) below)

Note: Administration always takes place on the Edge and is pushed to the home. Therefore, stations are integrated to the CM, not on the CM.

From the main edge proxy administration page:

1. Click **Media Servers**
2. Click **Add another Media Server Interface**.
3. The Host is the home proxy of the VOICE PORTAL interface.
4. **Select** the desired link type of **TCP/TLS**.
5. **SIP Trunk** refers to the CLAN/PROCR shown on the switch IP Node Names screen.
6. **Enter the login/password information** for the switch along with the switch name or IP (in the case of S87xx this should be the “active” shared IP-address).

This screen is where you ADD the Avaya CM (shown) or a Voice Portal Server to the SES

The help feature is very useful and can provide information that will aid the installation.

HOST: Our example shows an IP address. But if DNS is used this could be a domain name.

CM Login: The login/password show as **craft** is only an example. Normally this field would be administered as a different super user since the SES cannot do ASG authentication when it talks to CM.

The screenshot displays the 'Add Media Server Interface' form within the Avaya Integrated Management SIP Server Management interface. The form includes the following fields and options:

- Media Server Interface Name*:** CLAN1
- Host:** 148.147.98.93 (selected from a dropdown menu)
- SIP Trunk Link Type:** TCP (selected), TLS
- SIP Trunk IP Address*:** 135.9.84.79
- Media Server Admin Address:** 135.9.84.75
- Media Server Admin Login:** craft
- Media Server Admin Password:** *****
- Media Server Admin Password Confirm:** *****

Fields marked * are required. An 'Add' button is at the bottom of the form. A sidebar on the left contains a navigation menu with options like Top, Users, Conferences, Media Server, Extensions, Emergency Contacts, Hosts, Media Servers, List, Add, Adjunct Systems, Trusted Hosts, Services, Server Configuration, Certificate Management, IM logs, Trace Logger, and Export/Import to Provision. A 'Help' link is visible at the top left of the main content area.

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From the **List Media Servers**:

1. Click on **Map** for the interface just created to define an address map.
2. Click on **Add another Map** and enter a name and pattern that will map to the desired extensions on the PBX.

Notes: Multiple maps may be necessary. The example screen below allows for any extension beginning with 80 that describes the dial plan on the example PBX. For 4-digit extensions you would use "80[0-9]{2}" where the {2} indicates only 2-digits follow "80."

Media server address maps are ONLY required when CM receives an inbound SIP message from a non-administered OPTIM resource.

NOTE: This screen is where you define, or map, the extension numbers in the Pattern field allowing the SES to match a SIP invite message (connection) to an extension.

These screenshots are only examples. Information should be specific for your installation.

NOTE: Please refer to *Installing and Administering SIP Enablement Services* for further information regarding MAPs.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header shows the Avaya logo and the title 'Integrated Management SIP Server Management' with the server IP '148.147.98.93'. A navigation menu on the left lists various system components. The main content area is titled 'Add Media Server Address Map' and contains a form with the following fields:

- Name***: A text input field containing '80xx'.
- Pattern***: A text input field containing '^sip:80[0-9]*@avaya.c'.
- Replace URI**: A checkbox that is checked.

Below the form, a note states 'Fields marked * are required.' and an 'Add' button is visible.

- continued on next page -

Starting from the main proxy administration page, perform the following actions:

- 1) Expand “**Adjunct Systems**”
- 2) Click “**Add**”
- 3) In the “**System Name**” field, enter the desired name for the system.
This MUST match the entry for the Voice Portal Server Handle defined on the PBX pilot hunt group.
- 4) From the **Host** Name drop-down, select the name of the home proxy the Voice Portal System is using.
- 5) Click the **Add** button and then the Continue button.

NOTE: This screen is where you **ADD** the Voice Portal System SIP server(s) name and associated Name / IP Address.

System Name must match the Voice Portal handle specified on page 2 of the VPS Hunt Group in CM as shown on page 14 in this CN.

HOST: Our example shows an IP address but if DNS is used this could be a domain name.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title 'Integrated Management SIP Server Management', and the server IP '148.147.98.93'. A navigation menu on the left lists various system components, with 'Adjunct Systems' expanded. The main content area shows the 'Add Adjunct System' form, which includes fields for 'System Name*' (containing 'VoicePortalSystem') and 'Host' (a dropdown menu showing '148.147.98.93'). A note below the fields states 'Fields marked * are required.' and an 'Add' button is at the bottom.

- continued on next page -

The list of available Adjunct Systems is now displayed.

- 6) Click on “**List Adjunct Servers**” for the system (just defined above)
- 7) Click on “**Add Another Adjunct Server**”

NOTE: This screen shows there are 7 systems SIP Adjunct Systems (under the System column) known to the SES.

Your screen may only show 1.



Integrated Management
SIP Server Management
Server: 148.147.98.93

Help
Exit
Update

Top

- ▣ Users
- ▣ Conferences
- ▣ Media Server
 - Extensions
 - Emergency Contacts
- ▣ Hosts
- ▣ Media Servers
- ▣ Adjunct Systems
 - List
 - Add
- ▣ Trusted Hosts
 - Services
- ▣ Server Configuration
- ▣ Certificate
 - Management
 - IM logs
- ▣ Trace Logger
- ▣ Export/Import to
 - ProVision
 - Update


List Adjunct Systems

Commands				System	Host
Edit	Delete	List Adjunct Servers(0)	List Application IDs(1)	VoicePortalSystem	148.147.98.93
Edit	Delete	List Adjunct Servers(1)	List Application IDs(2)	alexsb	148.147.98.93
Edit	Delete	List Adjunct Servers(1)	List Application IDs(7)	avayaibm	148.147.98.93
Edit	Delete	List Adjunct Servers(1)	List Application IDs(5)	avrh4	148.147.98.93
Edit	Delete	List Adjunct Servers(2)	List Application IDs(6)	dxperf	148.147.98.93
Edit	Delete	List Adjunct Servers(2)	List Application IDs(7)	ixperf	148.147.98.93
Edit	Delete	List Adjunct Servers(1)	List Application IDs(1)	kw	148.147.98.93

[Add Another Adjunct System](#)

- continued on next page –

- 8) In the “**Server Name**” field (*below*), enter a name for the VP
- 9) Select the **TCP/TLS** setting as needed
- 10) Enter the **IP address** of the MPP.
- 11) Click **Add** and **Continue**.
- 12) **Repeat steps 8 to 14** for each MPP in the system.
- 13) Click **Update** when complete.

NOTE: This screen shows where you add each MPP comprising the Voice Portal system.

NOTE:

The Server ID is where you would enter the name of each MPP.

The Server IP address is that MPP's associated IP Address. If the SES is configured with DNS only then would you be able to use a domain name.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top bar includes the Avaya logo, 'Integrated Management SIP Server Management', and the server IP '148.147.98.93'. A navigation menu on the left lists various system components. The main area displays the 'Add Adjunct Server' form with the following fields:

- Host: 148.147.98.93
- System: VoicePortalSystem
- Server Name*: VPS1
- Server ID: MPP1
- Link Type: ☒ TCP ☐ TLS
- Server IP Address*: 135.9.84.76

A note at the bottom of the form states: 'Fields marked * are required.' There is an 'Add' button at the bottom right of the form.

NOTE: Adjunct servers are simply members of the SIP Adjunct System. (i.e., think of this as you would members in a hunt group.)

- The Server Name has no relevance to either.
- The Server IP address is where you want calls to go.

– continued on next page –

From the List Adjunct Systems screen.

- 14) Click on “**List Application IDs**” for the system you want to add a Pilot # for. In our example screen it is listed as VoicePortalSystem.

AVAYA Integrated Management SIP Server Manager
Server: 148.147.98.93

Help Exit Update

Top
 Users
 Conferences
 Media Server
 Extensions
 Emergency Contacts
 Hosts
 Media Servers
 Adjunct Systems
 List
 Add
 Trusted Hosts
 Services
 Server Configuration
 Certificate Management
 IM logs
 Trace Logger
 Export/Import to ProVision
 Update

List Adjunct Systems

Commands		System	Host
Edit	Delete	List Adjunct Servers(0) List Application IDs(1) VoicePortalSystem	148.147.98.93
Edit	Delete	List Adjunct Servers(1) List Application IDs(2) alexsb	148.147.98.93
Edit	Delete	List Adjunct Servers(1) List Application IDs(7) avayaibm	148.147.98.93
Edit	Delete	List Adjunct Servers(1) List Application IDs(5) avrh4	148.147.98.93
Edit	Delete	List Adjunct Servers(2) List Application IDs(6) dxperf	148.147.98.93
Edit	Delete	List Adjunct Servers(2) List Application IDs(7) ixperf	148.147.98.93
Edit	Delete	List Adjunct Servers(1) List Application IDs(1) kw	148.147.98.93

Add Another Adjunct System

- 15) Click on “**Add another Application ID**”

AVAYA Integrated Management SIP Server Manager
Server: 148.147.98.93

Help Exit Update

Top
 Users
 Conferences
 Media Server
 Extensions
 Emergency Contacts
 Hosts
 Media Servers
 Adjunct Systems
 List
 Add
 Trusted Hosts
 Services
 Server Configuration
 Certificate Management
 IM logs
 Trace Logger
 Export/Import to ProVision
 Update

Add Application ID for system VoicePortalSystem

Application ID*

Host 148.147.98.93

Fields marked * are required.

Add

On this [example](#) screen, the Application ID field is where the pilot number for the Voice Portal System is entered. This number is taken from the Hunt Group screen.

Please note that our example screen shows 4321 as the number. Your entry for a Pilot Number will correspond to that shown on the Hunt Group on your Avaya CM.

Testing the Application

6.0. CONFIGURING the Voice Portal Server for SIP

To configure your Voice Portal System to use SIP, please refer to the [Avaya Voice Portal Installation and Configuration Guide](#).

7.0 TESTING

7.1 Testing the Application

- Call the pilot number for your system. Validate the functionality.

8.0 CONSIDERATIONS / ALTERNATIVES

8.1 Known Issues:

- a. **CM may require administration to remove “ - ” (hyphen)** from the called number string sent to VOICE PORTAL. Until CM defsw054628 is fixed (CM build 626 at the earliest), if aaa-bbbb is being sent rather than aaabbbb, perform the following administration.
 - On the dial-plan parameters form ("cha dial parameters" at the CM SAT) change the "7-Digit Extension Display Format: " field to be xxxxxxx (remove the "-" that defaults in this format).

Note: Check with customer as this will change the display format on the stations/phones.
- b. **Call diversion interoperability between QSIG and SIP (QSIG/SIP Interworking) is not supported.** A solution for those being served from remote PBX's is to change the Voicemail Huntgroup type on those PBX's to SIP and let them cover directly to the Voice Portal over a SIP trunk. It should be possible to leave QSIG in place between the PBX's for feature transparency of CM features, and still configure SIP coverage to Voice Portal from each PBX independently. This solution was used in Alpha trial to allow UCC coverage and requires CM load 625 (or later).
- c. **Vectors are not supported.** Calls that route to Voice Portal from a vector pass the vector number and not the called party information.
- d. **Multiple Call Forwarding will only pass caller information of the last forwarded party.** When a call covers to another/intermediate person(s) before reaching Voice Portal the original called party will not be sent, but instead that of the last party where the call was forwarded from before reaching Voice Portal.

Important notes regarding this integration

8.2 SIP integrations may not be reliable for TTY if the IP network is unable to support uncompressed audio with no packet loss. For this reason we currently do not support TTY with this SIP integration.

8.3 Multiple Network Regions – If multiple network regions exist where call flow on the switch can travel to/from the network region used by Voice Portal, additional settings are necessary to ensure the codec defined for use by Voice Portal is among each of those network regions. In this case, it is recommended that Voice Portal be assigned its own network region. That network region number should then be placed in the “Far-end Network Region” field of the SIP Signaling Group used by Voice Portal as follows:

Step 1. Edit page 1 of Voice Portal's ip-network-region form to use the VOICE PORTAL codec set.

Step 2. Go to page 3 of the form and enter the Voice Portal codec set number next to all other network regions that may carry calls to / from Voice Portal.

CHANGE HISTORY

Revision	Issue Date	Reason for Change
Rev. A	09/23/08	Initial release.
Rev. B	12/15/08	Added new screen shots and Alternate Means of Routing for handling more than 10 Application IDs.

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