Esna Officelinx: Direct SIP Integration with Avaya CM

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ESNA OFFICELINX: DIRECT SIP INTEGRATION

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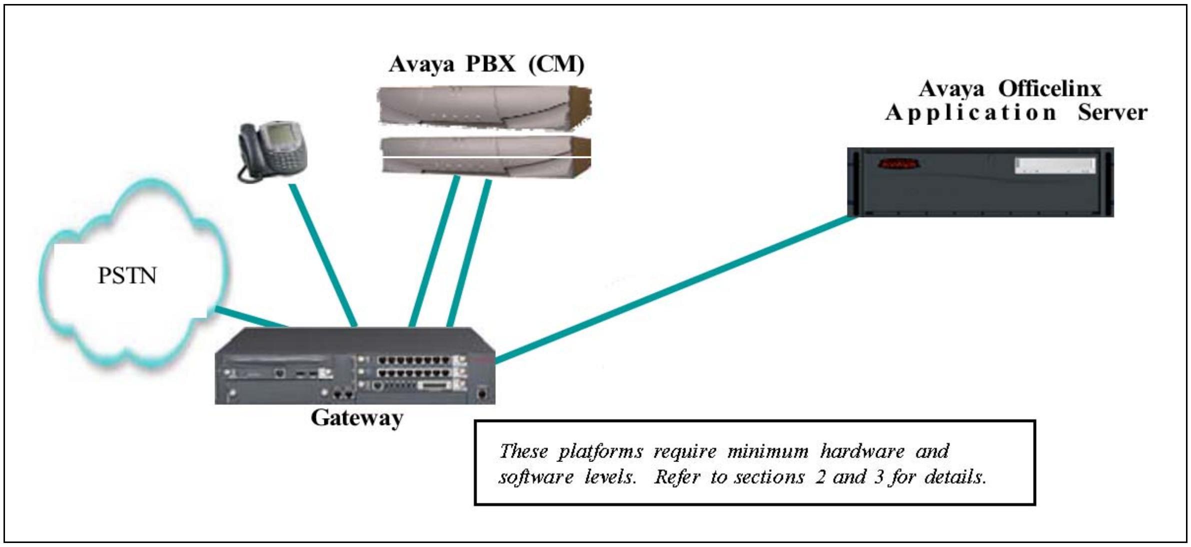
Esna Officelinx: Direct SIP Integration with Avaya CM

Introduction

This configuration note is intended for Certified Esna Officelinx technicians and engineers who are familiar with Officelinx Application Server procedures and terminology. It also assumes that you are Avaya certified or very familiar with the features and functionality of the Avaya PBXs supported in this document, and with the SIP protocol.

Use this document in conjunction with the Officelinx server Installation Guide and the Avaya PBX Administration Guide.

Please read the entire document before attempting any configuration.



Method of Integration

Session Initiation Protocol (SIP) integration provides connectivity with the Avaya PBX over a Local Area Network (LAN). The connectivity between the Officelinx Application Server and the Avaya Communication Manager PBX is achieved over IP-connected SIP trunks. This integration passes call information and MWI using SIP packets.

Pre-requisites

Esna Officelinx Application Server Requirements

Esna Officelinx version 8.x or higher is required to perform this integration.

PBX Hardware Requirements

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

S85x0/S87x0/S8x00:

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

  Note:
  FOR FAX Support: TN2302 Firmware 111 minimum / TN2602AP Firmware 24 minimum.

  Note:
  N2302 IP Media Processors DO NOT support SRTP. If you are using SRTP use the TN2602.

- TN799D C-LAN for signaling (only in G650 gateways)

Avaya S8xx0 server with Processor Ethernet:

- PROCR (for signaling [in place of CLAN card])
- MM760/On-board VOIP

  Note:
  The MM760 is used to add additional VOIP resources that may be required based on traffic requirements.

PBX Software Requirements

For Single Officelinx configurations: Avaya CM 5.2.1 and later.

For Multiple Officelinx Server Configurations: The minimum software releases that can be used are, CM 5.2.1 and CM 6.0.1.

Note:
Support for multiple voice server configurations with CM5.2.1 is only available upon request. This requires seed patch 18479 to be re-written for your specific CM5.2.1 SP (18481 is available for CM5.2.1 SP4). Please contact your Avaya representative for more information.

Connectivity

- Ethernet LAN connectivity - TCP/IP

Customer Provided Equipment

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

Supported Integration Features

● - Items that are supported

System Forward to Personal Greeting	
All Calls	●
Ring/no answer	●
Busy	●
Busy/No Answer	●

Station Forward to Personal Greeting	
All Calls	●
Ring/no answer	●
Busy	●

Auto Attendant	●
Call Me	●
Direct Call	●
External Call ID (ANI)	●
Fax *	●
Find Me	●
Internal Call ID	●
Message Waiting Indication (MWI)	●
Multiple Call Forward	●
Multiple Greetings	●
N+1	●
Outcalling	●
Queuing	●
Return to Operator	●

* - T.38 (Internal) Fax is supported.

Important:
PBX options or features not described in this Configuration Note are not supported with this integration. To implement options/features not described here, please contact the Avaya Switch Integration product manager.

Switch Configuration for IP Integration

The following tasks must be completed in the described order when programming the PBX for integration. PBX programming is intended for Certified PBX technicians or engineers.

- Verify customer option for SIP trunking
- Assign Local Node Number
- Administer C-LAN and IP Media Processor circuit packs (if using an S8xxx that requires this)
- Assign IP node names and IP addresses to C-LAN, IP Media Processor (if using an S8xxx that requires this)
- Define IP interfaces (if using an S8xxx that requires this)
- Administer IP Network Regions
- Create SIP signaling groups
- Create SIP trunk groups associated with SIP signaling groups
- Create Hunt Groups (Pilot Numbers)
- Create Coverage Paths to Pilot Hunts
- Create Route Patterns 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.

The table shows the values used in the examples throughout this document with regard to the S8300/S84x0/S85x0/S87x0 setup.

Page #	Field Value
-	Extension Length = 8
21	Local Node Number= 1 CLAN & MedPro Circuit Packs: 01A08 = TN799D C-LAN 01A09 = TN2602 IP Media Processor
22	IP Node Names: clan2-mtn135.9.81.29 clan3-mtn135.9.81.111 officelinxipaddr 148.147.35.88 mountain-prow3135.9.81.214 mountain-prow2135.9.81.52 Gateway001135.9.81.254 IP Interfaces (refer to CLAN & MedPro Circuit Packs above)
26	IP Network Regions = 1
28	SIP Signaling Group = 15 & 16
29	Trunk Group = 15 & 16
32	Hunt group = 252, 253 Pilot # 25281100, 25281099
33	Coverage Path = 252, 253
34	Route Pattern = 15, 16 AAR Analysis = 25281099 / 25281100
35	AAR Digit Conversion: Digits = n/a
36	Public Numbering Format: Public Extension Length = 8
39	Subscriber extensions = 252xxxxx

Note:
These are sample entries provided for illustration only. Consult your customer for the actual system values.

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**.

Note:
OPS Licenses are needed for all SIP stations (telephones). They are considered non-native / off-premise to CM. OPS Licenses are not needed for SIP far-end appliances such as Officelinx, MM & AAM.

Important:
Only change the recommended fields.

display system-parameters customer-options		Page	1 of 10
	OPTIONAL FEATURES		
G3 Version: V15	Software Package:	Standard	
Location: 1	RFA System ID (SID):	1	
Platform: 12	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 - PBFMC:	0	0
	Maximum Off-PBX Telephones - PVFMC:	0	0
	Maximum Off-PBX Telephones - SCCAN:	100	0
(NOTE: You must logoff & login to effect the permission changes.)			

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	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		
ATM WAN Spare Processor?	n	Digital Loss Plan Modification?	y
ATMS?	n	DS1 MSP?	n
Attendant Vectoring?	n	DS1 Echo Cancellation?	n

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

display system-parameters customer-options		Page 4 of 11	
	OPTIONAL FEATURES		
Emergency Access to Attendant?	y	IP Stations?	y
Enable 'dadmin' Login?	y		
Enhanced Conferencing?	y	ISDN Feature Plus?	n
Enhanced EC500?	y	ISDN/SIP Network Call Redirection?	n
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 Call Handling (Basic)?	n
Hospitality (Basic)?	y	Multimedia Call Handling (Enhanced)?	n
Hospitality (G3V3 Enhancements)?	n	Multimedia IP SIP Trunking?	n
IP Trunks?	y		
IP Attendant Consoles?	n		
(NOTE: You must logoff & login to effect the permission changes.)			

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

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

display system-parameters features	Page 1 of 18
FEATURE-RELATED SYSTEM PARAMETERS	
Self Station Display Enabled?	n
Trunk-to-Trunk Transfer:	**all***
Automatic Callback with Called Party Queuing?	n
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: music Type:	port 01C1001
Music (or Silence) on Transferred Trunk Calls?	all
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

* - 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 **799**.

display feature-access-codes		Page 1 of 7	
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:	799		
Auto Route Selection (ARS) - Access Code 1:	9	Access Code 2:	
Automatic Callback Activation:		Deactivation:	#21
Call Forwarding Activation Busy/DA: All:	*21	Deactivation:	
Call Forwarding Enhanced Status: Act:		Deactivation:	
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:			
Conditional Call Extend Activation:		Deactivation:	
Contact ClosureOpen Code:		Close Code:	

```
display dialplan parameters
                              DIAL PLAN PARAMETERS
                      Local Node Number:  1              ETA Node Number:
         UDP-ARS Calls Considered Offnet?  n           ETA Routing Pattern:
            UDP Extension Search Order:  local-extensions-first

   Retry ARS/AAR Analysis If All-Location Entry Inaccessible?  n

EXTENSION DISPLAY FORMATS
                                    Inter-Location/SAT      Intra-Location
           6-Digit Extension:       xx.xx.xx                xx.xx.xx
           7-Digit Extension:       xxx-xxxx                xxx-xxxx
           8-Digit Extension:       xxxxxxxx                xxxxxxxx
           9-Digit Extension:       xxx-xxx-xxx             xxx-xxx-xxx
          10-Digit Extension:       xxx-xxx-xxxx            xxx-xxx-xxxx
          11-Digit Extension:       xxxx-xxx-xxxx           xxxx-xxx-xxxx
          12-Digit Extension:       xxxxxx-xxxxxx           xxxxxx-xxxxxx
          13-Digit Extension:       xxxxxxxxxxxxx           xxxxxxxxxxxxx
```

- Administer C-LAN and IP Media Processor circuit packs (if using an S8xxx that requires this).

display circuit-packs					Page		1 of 5		
				CIRCUIT PACKS					
Cabinet:	1				Carrier:	A			
Cabinet Layout:	five-carrier				Carrier Type:	expansion-control			
Slot	Code	Sf	Mode	Name	Slot	Code	Sf	Mode	Name
01:					11:				
02:					12:				
03:					13:				
04:	TN744	E		CALL CLASSIFIER	14:				
05:	TN744	E		CALL CLASSIFIER	15:				
06:	TN744	E		CALL CLASSIFIER	16:				
07:	TN744	E		CALL CLASSIFIER	17:				
08:	**TN799**	**D**		**CONTROL-LAN**	18:				
09:	**TN2602**			**IP MEDIA PROCESSOR**	19:				
10:									
				'#' indicates circuit pack conflict.					

- Assign IP Node names IP addresses to C-LAN, IP Media Processor (if using an S8xxx that requires this). Enter the appropriate IP addresses for the installation.
  Each Officelinx Server needs to be added in node-

list node-names all

Page 1

NODE NAMES

Type	Name	IP Address
AUDIX	apollo1	135.9.80.216
AUSIX	shuttle	135.9.80.217
IP	Gateway001	135.9.81.254
IP	bard-clan	135.9.82.122
IP	carrera-icc	135.9.127.240
IP	**clan1**	**135.9.81.203**
IP	clan2-mtn	135.9.81.29
IP	clan3-mtn	135.9.81.111
IP	clan4-mtn	135.9.81.112
IP	clan5-mtn	135.9.81.123
IP	**offcielinxipaddr**	**148.147.35.88**
IP	d2f20mmsip	135.9.84.111
IP	default	0.0.0.0
IP	gateway	135.9.81.254
IP	mountain-prow	135.9.81.131
IP	mountain-prow2	135.9.81.52
IP	mountain-prow3	135.9.81.214

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

- Define the Ethernet data module for the C-LAN board (no longer used with CM 5.2 or greater):

```
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
```

```
display ip-interface 1a08                          Page    1 of 3

                        IP INTERFACES

             Type:  C-LAN
             Slot:  01A17        Target socket load and Warning level:  400
      Code/Suffix:  TN799 D        Receive Buffer TCP Window Size:  8320
Enable Interface?  y                        Allow H.323 Endpoints?  y
            VLAN:  n                         Allow H.248 Gateways?  y
   Network Region:  1                          Gatekeeper Priority:  5

                        IPV4 PARAMETERS
                    Node Name:  clan1
                  Subnet Mask:  /21
            Gateway Node Name:  Gateway001

                Ethernet Link:  2
Network uses 1's for Broadcast Addresses?  y
```

- 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.

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

                              IP Codec Set

              Codec set:   1

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

              Media Encryption
           1: 1-srtp-aescm128-hmac80
           2:
           3:
```

Important:
"Media Encryption" will only appear on the ip-codec-set screen if it is enabled in Customer Options. Several types of encryption are available. The encryption type "1-srtp-aescm128- hmac80" shown here is one example. Please consult with the appropriate technical resources to determine what type is needed for your PBX.
SRTP to HIGH or LOW and correspond to:
High = 1-srtp-aescm128-hac80
Low = 2-srtp-aescm128-hmac32

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

display ip-codec-set 1		Page	2 of 2
	IP Codec Set		
		Allow Direct-IP Multimedia?	n
	Mode	Redundancy	
FAX	t.38-standard	0	
Modem	off	0	
TDD/TTY	US	3	
Clear-channel	n	0	

Note:
If you plan to use internal fax, you must administer FAX Mode as “t.38-standard”

- Define IP Network Regions. In this example network region ‘1’ is selected. Define the local domain for the SIP network in this example “cmapsv.avaya.com” is used.

```
display ip-network-region 1                                        Page    1 of 19
                                   IP NETWORK REGION
              Region:  1
            Location:                      Authoritative Domain:  cmapsv.avaya.com
                Name:
MEDIA PARAMETERS                               Intra-region IP-IP Direct Audio:  yes
           Codec Set:  1                       Inter-region IP-IP Direct Audio:  yes
        UDP Port Min:  2048                               IP Audio Hairpinning?  y
        UDP Port Max:  8001
DIFFSERV/TOS PARAMETERS                                 RTCP Reporting Enabled?  y
Call Control PHB Value:      34              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:  7
        Audio 802.1p Priority:  6
        Video 802.1p Priority:  5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323IP ENDPOINTS                                               RSVP Enabled?  n
            H.323 Link Bounce Recovery?  y
             Idle Traffic Interval (sec):  20
              Keep-Alive Interval (sec):  5
                       Keep-Alive Count:  5            Keep-Alive Interval (sec):  5
                       Keep-Alive Count:  5
```

Note:
The Authoritative Domain should match what is used on the Signaling Group so calls placed from the Officelinx to the CM will authenticate properly. This is the Near Region Domain and corresponds to the CLAN or PROCR Region.

Important:
If using multiple IP Network Regions, where Officelinx may be in a different region than subscribers' IP Phones, make sure to administer Inter Network Region Connection Management in the IP Network Regions so calls will complete properly.

```
change ip-network-region 1                                      Page    2 of 19
                                IP NETWORK REGION
INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
                                   Incoming LDN Extension:
                    Conversion To Full Public Number - Delete:          Insert:
                  Maximum Number of Trunks to Use for IGAR:
                  Dial Plan Transparency in Survivable Mode?  n

BACKUP SERVERS(IN PRIORITY ORDER)               H.323 SECURITY PROFILES
1                                               1      challenge
2                                               2
3                                               3
4                                               4
5
6                                             Allow SIP URI Conversion?  y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
                  Near End Establishes TCP Signaling Socket?  y
                                    Near End TCP Port Min:  61440
                                    Near End TCP Port Max:  61444
```

Allow SIP URI Conversion? (default is "y") - Used to specify whether a SIP Uniform Resource Identifier (URI) is permitted to change. For example, if "sips:// in the URI is changed to "sip://" then the call would be less secure but this may be necessary to complete the call. If you enter n for 'no' URI conversion, then calls made from SIP endpoints that support SRTP to other SIP endpoints that do not support SRTP will fail. Enter "y" to allow conversion of SIP URIs.

- 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 OFFICELINX APPLICATION Server. For this example signal group 15 was selected using TLS transport with port 5061.

```
display signaling-group 15
                              SIGNALING GROUP
  Group Number:   1                         Group Type:   sip
                                      Transport Method:   tls
  IMS Enabled?    n

     Near-end Node Name:  clan1              Far-end Node Name:  officelinxipaddr
     Near-end Listen Port:  5061             Far-end Listen Port:  5061
                                        Far-end Network Region:  2
          Far-end Domain:  cmapsv.avaya.com
                                 Bypass If IP Threshold Exceeded?  n
            DTMF over IP:  rtp-payload
                                  Direct IP-IP Audio Connections?  y
    Enable Layer 3 Test?   n                 IP Audio Hairpinning?  y
  Session Establishment Timer(min):   3     Alternate Route Timer(sec):  6
```

Far-end Domain: The value here should match the Authoritative Domain field on the IP Network Region screen to allow inbound calls (SIP messages) to CM from the Aura Messaging to work properly.

Far-end Node Name: This is the Node Name for the Officelinx Address in its SIP Specific Configuration screen (see Section 6.0).

IP Audio: For shuffling IP-IP Audio Connections and IP Audio Hairpinning may be set to ”Y”. Shuffling may need to be set to no if re-invites are occurring.

Enable Layer 3 Test? For Single voice server configurations set to “N”. For Multiple voice server configurations, set to “Y”.

- Create the trunk group for SIP.

```
display trunk-group 15                                  Page    1 of 21
                                TRUNK GROUP

   Group Number:  15              Group Type:  sip       CDR Reports:  y
     Group Name:  To_MM_SIP              COR:  1      TN: 1      TAC:  715
      Direction:  two-way       Outgoing Display?  n
   Dial Access?   n                         Night Service:
   Queue Length:  0
   Service Type:  tie              Auth Code?  n
                                       Member Assignment Method:  manual
                                                Signaling Group:  15
                                              Number of Members:  255
```

When you are using CM5.2.1 patched for multiple voice server integration (see section 3.1) or CM6.0.1 you can set the member assignment to manual.

Doing this will allow you to manually distribute the calls for load balancing. This can be seen in the Trunk Group screen for GROUP MEMBER ASSIGNMENTS on the next Page.

```
display trunk-group 15                                    Page   2 of 21
        Group Type:  sip


TRUNK PARAMETERS
      Unicode Name:  yes
                                      Redirect On OPTIM Failure:  5000


             SCCAN?  n                       Digital Loss Group:  18
              Preferred Minimum Session Refresh Interval (sec):  600
```

```
display trunk-group 15                                    Page   3 of 20
TRUNK FEATURES
          ACA Assignment?  n              Measured:  none
                                              Maintenance Tests?  y
                    Numbering Format:  public


                                    Replace Unavailable Numbers?  n
```

display trunk-group 15			Page 5 of 20
	TRUNK GROUP		
		Administered Members (min/max):	1/8
GROUP MEMBER ASSIGNMENTS		Total Administered Members:	8

	Port	Name	Night	SIG Group
1:	T00001			15
2:	T00002			20
3:	T00003			15
4:	T00004			20
5:	T00005			15
6:	T00006			20
7:	T00007			15
8:	T00008			20
9:	T00009			15
10:	T00010			20
11:	T00011			15
12:	T00012			20
13:	T00013			15
14:	T00014			20
15:	T00015			15

This example shows the set up the Trunk Group Members in an interleaving fashion to distribute calls between two Voice servers, each assigned their own signaling group. Sig Grp 15 is used with a single voice server. Sig Grp 20 points to a second voice server (if installed).

Note:

The Sig Grp field is Displayed ONLY when the Member Assignment Method is set to “manual” on page 1 of the Trunk Group screen (see previous page). If the Member Assignment Method is set to “auto” the Sig Grp field will not be displayed and manual assignment is not allowed. Administer member assignment so calls are distributed/interleaved among the servers.

Add Hunt Group(s): Configure a Hunt Group to be used as the Call Coverage Point for the Call Coverage Path assigned to the Officelinx subscribers. This hunt group's extension number will be used as the Officelinx Access Number. This hunt group is configured with no members assigned to it, and should be configured as follows:

display hunt-group 252		Page	1 of 60
	HUNT GROUP		
Group Number:	252	ACD?	n
Group Name:	**Apollo12**	Queue?	n
Group Extension:	**25281100**	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**		

- In the "Routing Digit (e.g. AAR/ARS Access Code)" field, enter your PBX's AAR Access Code as defined on page 19.

change hunt-group 252		Page 2 of 60
	HUNT GROUP	
	Message Center: sip-adjunct	
Voice Mail Number	Voice Mail Handle	Routing Digits (e.g., AAR/ARS Access Code)
25281100	**25281100**	**799**

Tip:
With Direct Integration, the Voice Mail Number can be used (again) as the Voice Mail Handle. With CM 5.2.x and CM 6.x, the Voice Mail Hunt Group Pilot number may not be available to the VXIBrowser. Making the "voice mail handle" match the "voice mail number" corrects this.

- Setup a coverage path for the subscriber's extensions. Assign to it the pilot hunt group number created in the earlier step.

```
display coverage path 252
                              COVERAGE PATH

              Coverage Path Number:  252
  Cvg Enabled for VDN Route-To Party?  n            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:   h252              Rng:              Point2:
          Point3:                                       Point4:
          Point5:                                       Point6:

       Command:
```

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

display route-pattern 15 Page 1 of 3

Pattern Number: 15 Pattern Name: sm1-2

SCCAN? n Secure SIP? **n** Grp FRL NPA Pfx Hop Toll No. Inserted

DCS/ IXC

	No		Mrk	Lmt	List	Del Dgts	Digits	QSIG Intw	
1:	15	0				0		n	user
2:								n	user
3:								n	user
4:								n	user
5:								n	user
6:								n	user

BCC VALUE	TSC	CA-TSE Request	ITC	BCIE	Service/ Feature	PARM	No. Dgts Subaddress	Numbering Format	LAR
0 1 2 M 4 W									
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 here should contain a map to the Pilot Number for the Aura Messaging Server system. Below is an example of an AAR dialed string shown in **boldface**.

Display aar analysis Page 1 of 2

AAR DIGIT ANALYSIS REPORT

Location: all

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Number
13000	5	5	130	aar	
131	5	5	130	aar	
13999	5	5	30	aar	
14000	5	5	130	aar	
25281099	8	8	16	aar	
25281100	**8**	**8**	**15**	**unk**	
26341000	8	8	10	aar	

AAR is a public numbering format. The Type of Number /Numeric Plan Indicator is national/ E.164. Although uses AAR for private network routing, the encoding of the Call Type remains public. If you are using an Avaya CM 6.x and set the Call Type in the AAR Analysis screen to aar, CM will add a ‘+’ prefix to the CPN and calls may not integrate properly. Setting the Call Type to “unk” will prevent the “+” from being added as a prefix. Alternatively, change the Numbering Format on the Route Pattern to private.

- Set the route pattern for the switch location.

display locations					Page	1 of 2
			LOCATIONS			
		ARS Prefix 1 Required For 10-Digit NANP Calls? y				
Loc. No.	Name	Timezone Offset	Rule	NPA	Proxy Rte	Sel Pat
1:	Main	+ 00:00	0		15	

The Proxy Selection Route Pattern field identifies the routing pattern that is used to reach the CM. This route pattern points to the SIP trunk so that outbound calls over ISDN trunks will know where to send updated ISDN messages. For example, 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.

- **Define Public Numbering**. Be sure to administer an entry to match each extension the message server will be supporting. In this example, extension 8XXX was used. For the trunk group, use the same trunk group number created above.

list public-unknown-numbering			Page	1 of 2
		NUMBERING - PUBLIC/UNKNOWN FORMAT		
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len
8	**2**			**8**
5	3			5
5	3	130		5
4	4	13	1415263	11

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.

Subscriber Administration

Subscriber administration has several parts: Administering the MWI, assigning the call coverage path, and specifying softphone capability.

Follow these steps to program the subscribers stations assigned to the OFFICELINX APPLICATION SERVER.

The screens for station 25281101 show how to administer for a non-SIP phone. The screens for station 25281110 show how to administer for a SIP phone which includes off-PBX administration.

Important:
The screens shown below are only provided as an example. Please refer to Installing and Administering SIP Enablement Services for further information.

Administering a Non-SIP Station

change station 25281101				Page	1 of 5
		STATION			
Extension:	25281101	Lock Messages?	n	BCC:	0
Type:	7406+	Security Code:	25281101	TN:	1
Port:	01C1702	Coverage Path 1:	252	COR:	1
Name:	apollo12 x25281101	Coverage Path 2:	2	COS:	1
		Hunt-to Station:			
STATION OPTIONS					
				Time of Day Lock Table:	
Loss Group:	2			Personalized Ringing Pattern:	1
Data Module?	n			Message Lamp Ext:	25281101
Display Module?	y				
Display Language:	english				

Survivable COR:	internal	Media Complex Ext:	
Survivable Trunk Dest?	y	IP SoftPhone?	n

change station 25281101		Page	2 of 5
	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		
H.320 Conversion?	n	Per Station CPN - Send Calling Number?	
Service Link Mode:	as-needed	EC500 State:	disabled
Multimedia Mode:	basic	Audible Message Waiting?	n
MWI Served User Type:	**sip-adjunct**	Display Client Redirection?	n
		Select Last Used Appearance?	n
		Coverage After Forwarding?	s
		Direct IP-IP Audio Connections?	y
Emergency Location Ext:	25281101	IP Audio Hairpinning?	n

Important:
Set each user's MWI Served User Type as "sip-adjunct". Otherwise MWI interrogation (polling) will not work.

Tip:
See the Considerations and Alternatives section in this document for information about changing the MWI Served User Type for many users.

Administering a SIP Station

display station 25281112				Page	1 of 5
		STATION			
Extension:	25281112	Lock Messages?	n	BCC:	0
Type:	4620	Security Code:		TN:	1
Port:	S00000	Coverage Path 1:	253	COR:	1
Name:	apollo12 x25281112	Coverage Path 2:		COS:	1
		Hunt-to Station:			
STATION OPTIONS					
				Time of Day Lock Table:	
Loss Group:	19			Personalized Ringing Pattern:	1
				Message Lamp Ext:	26341112
Speakerphone:	2-way			Mute Button Enabled?	y
Display Language:	english				
Survivable GK Node Name:					
Survivable COR:	internal			Media Complex Ext:	
Survivable Trunk Dest?	y			IP SoftPhone?	n
				Customizable Labels?	y

display station 25281112		Page	2 of 5
	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		
		EMU Login Allowed?	n
H.320 Conversion?	n	Per Station CPN - Send Calling Number?	
Service Link Mode:	as-needed	EC500 State:	disabled
Multimedia Mode:	enhanced	Audible Message Waiting?	n
MWI Served User Type:	**sip-adjunct**	Display Client Redirection?	n
		Select Last Used Appearance?	n
		Coverage After Forwarding?	s
		Direct IP-IP Audio Connections?	y
Emergency Location Ext:	25281112	IP Audio Hairpinning?	n

Important:
Set each user's MWI Served User Type as "sip-adjunct". Otherwise MWI interrogation (polling) will not work.

Tip:
See the Considerations and Alternatives section in this document for information about changing the MWI Served User Type for many users.

Create an 'Off-PBX' Station Mapping

Create an “Off-PBX” station mapping using the SIP trunk defined earlier.

In our previous example screens we had used trunk 15. Your trunk number may be different.

display off-pbx-telephone station-mapping 25281112 Page 1 of 3

STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station Extension	Application	Dial	CC Prefix	Phone Number	Trunk Selection	Config Set	Dual Mode
25281112	OPS		-	25281112	aar	1	

Configuring the Officelinx Application Server

Configuring the Esna Officelinx platform for proper PBX integration requires settings be set as indicated in the Officelinx Technical Operating Guidelines document.

Considerations and Alternatives

- **SIP integrations may not be reliable for TTY/TDD** if the IP network is unable to support uncompressed audio with no packet loss. For this reason Avaya does not support TTY/TDD with this SIP integration.
- **Multiple Network Regions** – If multiple network regions exist where call flow on the switch can travel to and from the network region used by the Officelinx Application Server, additional settings are necessary to ensure the codec defined for use with Officelinx is among each of those network regions. In this case, it is recommended that Officelinx 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 Officelinx as follows:

  Step 1. Edit page 1 of the Officelinx Application Server ip-network-region form to use the proper codec set.

  Step 2. Go to page 3 of the form and enter the Officelinx codec set number next to ALL network regions that may carry calls to / from Officelinx.
- In reference to supported "transport CODECs", AAM supports only G.711. Ensure the far end SIP end point (SIP gateway or SIP PBX) is set accordingly. Failure may result in undesirable or what's perceived to be a non-working or dysfunctional OL. G.711 is the front-ended transport CODEC, OL's back-end storage allows for both GSM and G.711 CODECs to the message store. This latter switch setting is found within the SMI of "System Parameters".
- If using the ONE-STEP Recording feature, the Recording Delay Timer setting in Feature-Related System Parameters must be set to 2000 msecs. If not, the originator may hear a call answer greeting when using this feature.

  Note:
  Customers using One-Step record may experience a slight delay of 2-4 seconds before recording begins.
- If you are using Outlook and attempt to Play a message on phone that requires an outside trunk and the call gets rejected/fails, check to see if service provide is blocking calls with names.

Appendix A: Revision History

Date	Issue	Change Summary
3 March, 2016	1.0	• Initial Release