

Avaya Solution & Interoperability Test Lab

Configuring Avaya Communication Manager in a Centralized Avaya Gatekeeper Environment - Issue 1.0

Abstract

These Application Notes describe connectivity between a network of Avaya Media Servers running Avaya Communication Manager in a centralized gatekeeper environment using Avaya Gatekeepers. This is one approach to minimizing administrative complexity for enterprises deploying a large and growing number of IP-enabled PBXs and call servers.

1. Introduction

These Application Notes describe one approach to minimizing administrative complexity for enterprises deploying a large and growing number of IP-enabled PBXs and call servers. As the number of independent sites that need to be interconnected via the data WAN scales up, it may be desirable to:

- Minimize the configuration requirements (e.g., H.323 signaling groups and trunk groups) at each of the sites
- Centralize routing intelligence at fewer locations
- Manage admission of calls to the IP network more centrally
- Retain alternate routing paths such that calls denied admission to the enterprise IP network could automatically be completed via alternate PSTN routes.

The configuration described in these Application Notes is one approach to satisfying these objectives. Each IP-enabled PBX or call server, referred to as an "end system", need only be configured for simple routing and IP Trunking to a site designated as its "regional gatekeeper". Regional gatekeeper(s) require routing and IP Trunk provisioning only to the end systems in their region and one or more peer regional gatekeepers. This arrangement obviates the need for a mesh configuration, where each node on the network needs an awareness of all other nodes.

The arbitration of whether a new call request should be allowed to traverse the enterprise IP network can be managed by the regional gatekeepers. The regional gatekeepers use the Avaya Communication Manager Call Admission Control feature to monitor network utilization, either by number of calls or by bandwidth utilization. If the regional gatekeeper denies admission of the call to the IP network based on configured thresholds, the call signaling that rejects the call back to the "end system" can result in the automatic re-routing of the call over local PSTN trunks in the end systems, if desired. The Avaya Communication Manager Look-ahead Routing feature enables the end systems to automatically re-route calls when network congestion is encountered on the preferred IP trunk paths.

In **Figure 1**, the Avaya DEFINITY® Server R (US1) and the Avaya DEFINITY® Server R (US2) are "end systems" whose "regional gatekeeper" (GK1) is a site with an Avaya S8700 Media Server pair. The second regional gatekeeper (GK2) is a site with an Avaya S8500 Media Server, and GK2 provides services for the end system with the S8700 Media Server pair (UK2). The sites are labeled to suggest an example approach where US PBX sites are spokes on the left, with GK1 serving as a regional US gatekeeper hub. Sites on the right are UK PBX sites with GK2 serving as a regional UK hub. Obviously, customer circumstances will vary.

Two basic topologies are supported in an Avaya centralized gatekeeper environment, fully interconnected as well as hub and spoke. Combinations of these topologies are also supported. The configuration covered here is two adjacent hub and spoke configurations.

The configuration steps covered in these Application Notes focus on the trunking, signaling, and call management connectivity between the locations. **Figure 1** shows a logical representation of the connections between the Gateways and Gatekeepers. There are two signaling groups and trunk groups between location US1 and Gatekeeper 1. This was simply done for redundancy. There is only one signaling group and trunk group between all other systems. A LAN infrastructure was used to physically interconnect the PBXs, but this was a standard configuration, that supported QoS, and is beyond the scope of these Application Notes.

The call flow between location US1 and US2 is as follows:

- 1) Communication Manager in location US1 sends a call request to Gatekeeper 1.
- 2) Gatekeeper 1 determines if the Call Admission Control limit between Gatekeeper 1 and location US2 has been reached. If the limit has been reached, the call request is rejected, and Communication Manager in location US1 can use Look Ahead Routing (LAR) to find an alternate path to location US2. If the limit has not been reached, Gatekeeper 1 sends a call request to location US2.
- 3) When Communication Manager in location US2 accepts the call, if the call shuffles, between US1 and US2 (or the endpoints), Gatekeeper 1 makes note of the direct utilization between US1 and US2. If the call does not shuffle, Gatekeeper 1 makes note of the utilization for its path to US1 and its path to US2.

The call flow between location US1 and UK1 is as follows:

- 1) Communication Manager in location US1 sends a call request to Gatekeeper 1.
- 2) Gatekeeper 1 determines if the Call Admission Control limit between Gatekeeper 1 and Gatekeeper 2 has been reached. If the limit has been reached, the call request is rejected, and Communication Manager in location US1 can use Look Ahead Routing (LAR) to find an alternate path to location UK1. If the limit has not been reached, Gatekeeper 1 sends a call request to Gatekeeper 2.
- 3) Gatekeeper 2 determines if the Call Admission Control limit between Gatekeeper 2 and UK1 has been reached. If the limit has been reached, the call request is rejected all the way back to US1, and Communication Manager in location US1 can use Look Ahead Routing (LAR) to find an alternate path to location UK1. If the limit has not been reached, Gatekeeper 2 sends a call request to UK1.
- 4) When Communication Manager in UK1 accepts the call, whether or not the call shuffles, Gatekeeper 1 makes note of the utilization for its path to US1 and its path to Gatekeeper 2, and Gatekeeper 2 makes note of the utilization for its path to UK1 and its path to Gatekeeper 1.

While the calls are being set up, resources at the gatekeepers are being utilized, so there should be sufficient C-LAN and MEDPRO resources in the gatekeepers to support calls flowing through them. Even when calls shuffle, trunk port members are allocated in each of the associated trunk groups, so there should be sufficient members in the trunk groups to support all calls.



US 2

Notes:

----- Indicates the number of calls allowed by Call Admission Control between regions. Region 11 to 12 allows 3 shuffled/direct calls.
Region 11 to 12 allows 3 tandem calls (without shuffling) through Gatekeeper 1. Region 11 to 15 or 12 to 15 allows 2 calls (the bottleneck is between the Gatekeepers).

Figure 1: Logical Network Configuration

2. Equipment and Software Validated

Quantity **Device Description** Version 2 Avaya DEFINITY® Server R R011.r.03.1.535 (1.3.1) Avaya S8700/G600 Server/Gateway R011x.03.1.535 (1.3.1) 1 1 Avaya S8700/G600 Server/Gateway (GK1) R012x.01.0.411.7 (2.1) Avaya S8500/G650 Server/Gateway (GK2) R012x.01.0.411.7 (2.1) 1 4 Avaya 8410 Digital Telephones N/A 4 Avaya 4612 IP Telephone 1.8 4 Avaya 4620 IP Telephone 2.0

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

3. Configuration

The configuration steps covered in these Application Notes focus on the trunking, signaling, and call management connectivity for Avaya Communication Manager at the different locations.

3.1. Configuring the Avaya S8700 Server and G600 Gateway serving as Gatekeeper 1

Gatekeeper 1 had 2 C-LAN and 2 MEDPRO cards, assigned to VLAN 50, and in Network Region 13.

1. Add Node Names for Gatekeeper 1's C-LAN and MEDPRO cards, and assign them the appropriate IP Address. Add Node Names for the remote C-LANs to which Gatekeeper 1 will have signaling groups.

Name	:	IP A	ddre	SS	
33R1-CLAN1	60	.1	.1	.10	
G3R1-CLAN2	70	.1	.1	.10	
G3R2-CLAN1	40	.1	.1	.10	
GK1-CLAN1	50	.1	.1	.10	
GK1-CLAN2	50	.1	.1	.20	
GK1-MEDPRO1	50	.1	.1	.11	
GK1-MEDPRO2	50	.1	.1	.21	
GK2-CLAN1	30	.1	.1	.10	
default	0	.0	.0	.0	
procr					





4. Add a data module for each C-LAN. Select a unique "Data Extension" and "Link" for each data module. Note that the "Name" for the data module does not need to match the "Node Name" associated with the C-LAN. The association is done by the port field, which is a combination of the slot (01A04) and the signaling port (17). Set the "Type" field to "ethernet".

	DATA MODULE	
Data Extension: 20 Type: et Port: 0 Link: 1	0000 Name: CLAN1 thernet LA0417	
Network uses l's for	r Broadcast Addresses? y	
add data-module 2000)1 DATA MODULE	
add data-module 2000 Data Extension: 20 Type: et Port: 00 Link: 2	DATA MODULE DATA MODULE Name: GK1-CLAN2 Chernet LA0717	

5. There are 2 signaling groups between Gatekeeper 1 and Avaya G3R Server 1. When adding a signaling group, the "Trunk Group for Channel Selection" field must be left blank until the trunk group is configured. After the trunk group is configured use the "change signaling-group 1" command to populate the trunk group number. The "Supplementary Service Protocol" field was set to "b" to use QSIG signaling, but QSIG is not required for this configuration to work. The "Near-end Node Name" field should be set to the node name of a local C-LAN. The "Far-end Node Name" field should be set to the node name of a remote C-LAN. The "Near-End Listen Port" and "Far End Listen Port" should be "1719". The "Far-End Network Region" associated with Avaya G3R Server 1 is "11". "LRQ Required" should be set to "y". "Direct IP-IP Audio Connections" and "IP Audio Hairpinning" should be set to "y".



6.	There are 2 trunk groups between Gatekeeper 1 and Avaya G3R Server 1. Trunk group
	614 is shown here, which is associated with signaling group 1. Most of the fields are set
	to their defaults. Trunk group 615 is configured similarly and is associated with
	signaling group 2.
	display trunk-group 614 Page 1 of 22
	TRUNK GROUP
	Group Number: 614 Group Type: isdn CDR Reports: y
	Group Name: To G3R1 COR: 1 TN: 1 TAC: 136
	Direction: two-way Outgoing Display? n Carrier Medium: 1P Dial Access? n Busy Threshold: 255 Night Service:
	Queue Length: 0
	Service Type: tie Auth Code? n TestCall ITC: rest
	TestCall BCC: 4
	TRUNK PARAMETERS
	Codeset to Send Display: 6 Codeset to Send National IEs: 6
	Supplementary Service Protocol: b Digit Handling (in/out): enbloc/enbloc
	Trunk Hunt: ascend QSIG Value-Added? n
	Incoming Calling Number - Delete: Insert: Format:
	Bit Rate: 1200 Synchronization: async Duplex: full
	Disconnect Supervision - In? y Out? n Answer Supervision Timeout: O
	display trunk-group 614 Page 2 of 22
	TRUNK FEATURES
	ACA Assignment? n Measured: none Wideband Support? n
	Data Restriction? n NCA-TSC Trunk Member: 1
	Send Name: y Send Calling Number: y
	Used for DCS? n Hop Dgt? n Suppress # Outpulsing? n Format: unknown
	Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider
	Penlace Pestricted Numbers? n
	Replace Unavailable Numbers? n
	Send Connected Number: y
	Modify Tandem Calling Number? n Send HHI TE? v
	Send UCID? n
	Send Codeset 6/7 LAI IE? y
	Path Replacement with Retention? n
	Path Replacement Method: better-route
	SBS? n Network (Japan) Needs Connect Before Disconnect? n
	when entering the Port members, simply type "ip". After the form is submitted, virtual
	port numbers will be assigned by software. When the form is displayed, the virtual port
	value (i.e. "100001") is displayed.
	display trunk-group 614 Page 6 of 22
	Administered Members (min/max): 1/10
	GROUP MEMBER ASSIGNMENTS Total Administered Members: 10
	Port Code Sfx Name Night Sig Grp
	1: T00001
	2: T00002

7.	Signaling group 3 and Trunk group 616 are used to connect Gatekeeper 1 to Avaya G3R Server 2. The signaling group and trunk group are configured as shown in previous steps. The signaling group is shown here to highlight the association of Network Region 12 with Avaya G3R Server 2.								
	display signaling-group 3 SIGNALING GROUP								
	Group Number: 3 Group Type: h.323 Remote Office? n Max number of NCA TSC: 10 SBS? n Max number of CA TSC: 10 Trunk Group for Channel Selection: 616 Supplementary Service Protocol: b								
	Near-end Node Name: GK1-CLAN1 Near-end Listen Port: 1719Far-end Node Name: G3R2-CLAN1 Far-end Listen Port: 1719LRQ Required? y RRQ Required? nCalls Share IP Signaling Connection? n H245 Control Addr On FACility? n Bypass If IP Threshold Exceeded? nDTMF over IP: in-bandDirect IP-IP Audio Connections? y IP Audio Hairpinning? y Interworking Message: PROGress								
8	Signaling group 5 and Trunk group 617 are used to connect Gatekeeper 1 to Gatekeeper								
0.	2. The signaling group and trunk group are configured as shown in previous steps. The signaling group is shown here to highlight the association of Network Region 14 with Gatekeeper 2.								
	display signaling-group 5 SIGNALING GROUP								
	Group Number: 5 Group Type: h.323 Remote Office? n Max number of NCA TSC: 0 SBS? n Max number of CA TSC: 0 Trunk Group for Channel Selection: 617 Supplementary Service Protocol: b								
	Near-end Node Name: GK1-CLAN1Far-end Node Name: GK2-CLAN1Near-end Listen Port: 1719Far-end Listen Port: 1719Far-end Network Region: 14Calls Share IP Signaling Connection? nRRQ Required? nH245 Control Addr On FACility? nBypass If IP Threshold Exceeded? y								
	DTMF over IP: in-band Direct IP-IP Audio Connections? y IP Audio Hairpinning? y Interworking Message: PROGress								

9. Call and Bandwidth limits are configured on page 3 of the IP Network Region form. Gatekeeper 1 is logically associated with Network Region 13. On Page 1 of the IP Network Region, set "Intra-region IP-IP Direct Audio" and "Inter-region IP-IP Direct Audio" to "yes. Set the "IP Audio Hairpinning" field to "y". display ip-network-region 13 Page 1 of 19 IP NETWORK REGION Region: 13 Location: Home Domain: Name: Intra-region IP-IP Direct Audio: yes Inter-region IP-IP Direct Audio: yes AUDIO PARAMETERS Codec Set: 1 IP Audio Hairpinning? y UDP Port Min: 2048 UDP Port Max: 3028 RTCP Reporting Enabled? y RTCP MONITOR SERVER PARAMETERS DIFFSERV/TOS PARAMETERS Use Default Server Parameters? y Call Control PHB Value: 34 Audio PHB Value: 46 802.1P/Q PARAMETERS Call Control 802.1p Priority: 7 Audio 802.1p Priority: 6 AUDIO RESOURCE RESERVATION PARAMETERS H.323 IP ENDPOINTS RSVP Enabled? n H.323 Link Bounce Recovery? y Idle Traffic Interval (sec): 20 Keep-Alive Interval (sec): 5 Keep-Alive Count: 5 On Page 3 specify the limits between the direct regions controlled by Gatekeeper 1. What is shown here is that 3 calls will be allowed between Region 13 and Region 11; 3 calls between Region 13 and Region 12; 2 calls between Region 13 and Region 14. The limit can also be specified by bandwidth (e.g. 256:Kbits). When using bandwidth, take into consideration the Codec used and the packet size; for example G.711 with 20 ms packets use approximately 83 Kbits/sec. Appendix 1 shows bandwidth guidelines based on codec and frame size used. For a detailed description of bandwidth calculations and setting WAN bandwidth limits between network regions, consult the Administrator's Guide for Avaya Communication Manager. display ip-network-region 13 Page 3 of 19 Inter Network Region Connection Management src dst codec direct Dynamic CAC rgn rgn set WAN WAN-BW-limits Intervening-regions Gateway 13 1 13 2 13 3 13 4 13 5 13 б 13 7 13 8 13 9 13 10 11 3:Calls 13 1 У 12 13 1 3:Calls У 13 13 1 13 14 1 2:Calls У 13 15

10.	Page 3 of the screens for Regions 11 and 12 show that 3 calls are allowed between
100	Regions 11 and 12 Page 3 of the screens for Regions 11 12 and 14 show that
	connectivity between Degions 11 and 14, as well as 12 and 14 was indirect, with
	connectivity between Regions 11 and 14, as wen as 12 and 14 was induced, with
	intervening region 13. when a connection is indirect, the limits for the intervening
	region apply, for example if a call is to go from region 11 to 14, the limits to go from 11
	to 13 and 13 to 14 will apply.
	display ip-network-region 11 Page 3 of 19
	src dst codec direct Dynamic CAC
	rgn rgn set WAN WAN-BW-limits Intervening-regions Gateway
	11 2 11 3
	11 4
	11 5 11 6
	11 7
	11 9 11 10
	11 11 1
	11 12 1 y 3:Calls
	$\begin{array}{c ccccccccccccccccccccccccccccccccccc$
	11 15
	display ip-network-region 12 Page 3 of 19
	src dst codec direct Dynamic CAC
	rgn rgn set WAN WAN-BW-limits Intervening-regions Gateway
	12 1 12 2
	12 3
	12 4
	12 6
	12 7
	12 8
	12 10
	12 11 1 y 3:Calls
	12 12 1 12 13 1 y 3:Calls
	12 14 1 n 13: : :
	12 15
	display 1p-network-region 14 Inter Network Region Connection Management
	src dst codec direct Dynamic CAC
	rgn rgn set WAN WAN-BW-limits Intervening-regions Gateway
	14 1
	14 3
	14 4
	14 6
	14 7
	14 8 14 9
	14 10
	14 11 1 n 13: : :
	$\begin{array}{cccccccccccccccccccccccccccccccccccc$
	14 14 1

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13 of 28 AvayaCentralGKs **11.** Automatic Route Selection was used to route calls to the appropriate locations. This solution would also work with Automatic Alternate Routing and a Uniform Dial Plan. Since these are standard routine configurations, additional detail was not deemed necessary. It is worth re-emphasizing here that in this configuration the end PBXs only had signaling groups, trunk groups, and route patterns to their respective Gatekeepers. In a large configuration this significantly reduces administration required for trunks, signaling groups, and routing at each end PBX. All call routing takes place by the gatekeepers.

3.2. Configuring the Avaya S8500 Server and G650 Gateway serving as Gatekeeper 2

Gatekeeper 2 had 1 C-LAN and 1 MEDPRO, assigned to VLAN 30, and in Network Region 14.

1. Add Node Names for Gatekeeper 2's C-LAN and MEDPRO cards, and assign them the appropriate IP Address. Add Node Names for the remote C-LANs to which Gatekeeper 2 will have signaling groups.

				1	NODE NAMES	
Name	II	P Ad	ldre	SS		
GK1-CLAN1	50.	.1	.1	.10		
GK2-CLAN1	30.	.1	.1	.10		
GK2-MEDPRO1	30.	.1	.1	.11		
PBX3-CLAN1	20 .	.1	.1	.10		
default	0.	. 0	.0	.0		
procr				•		

2.	Add an IP Interface for the local C-LAN card,	located in slot a02. The "Node Name"								
	must match the one assigned in the Node Nam	es form. Assign the appropriate "Network								
	Region" and "VLAN" (in this case 14 and 30	respectively). Set the "Enable Ethernet								
	Port" field to "v".	, , , , , , , , , , , , , , , , , , ,								
	add ip-interface a02									
	TD INTERPACES									
	IP INIERFACE	5								
	Time: C-IAN	FTHFONFT ODTIONS								
	Slot: 01A02	Auto? n								
	Code/Suffix: TN799 D	Speed: 100Mbps								
	Node Name: GK2-CLANI	Duplex: Full								
	Subnet Mask: 255.255.255.0									
	Gateway Address: 30 .1 .1 .1									
	Enable Ethernet Port? y									
	VLAN: 30									
2										
3.	3. Add an IP Interface for the local MEDPRO card, located in slot a03. The IP Interface									
	a MEDPRO is configured in the same way as for a C-LAN.									
	add ip-interface aus									
	IP INTERFACE	S								
	Type: MEDPRO	ETHERNET OPTIONS								
	Slot: 01A03	Auto? n								
	Code/Suffix: TN2302	Speed: 100Mbps								
	IP Address: 30 .1 .1 .11	Duplex. Full								
	Subnet Mask: 255.255.255.0									
	Gateway Address: 30 .1 .1 .1									
	Network Region: 14									
	VLAN: 30									

4. Add a data module for the C-LAN. Select a unique "Data Extension" and "Link" for each data module. Note that the "Name" for the data module does not need to match the "Node Name" associated with the C-LAN. The association is done by the port field, which is a combination of the slot (01A02) and the signaling port (17). Set the "Type" field to "ethernet".

Data Exte	nsion: 20000	Name: GK2-CLAN1	
	Type: ethernet		
	Port: 01A0217		

5. Gatekeeper 2 has a signaling group and trunk group to Gatekeeper 1. There is also a signaling group and trunk group to the Avaya S8700 Server and G600 Gateway that is directly connected to Gatekeeper 2. The signaling group and trunk groups are configured in the same manner as shown for Gatekeeper 1. The configuration for the signaling group to Gatekeeper 1 is shown here for comparison. The "Far-end Network Region" for Gatekeeper 1 is 13. The configuration screens for trunk group 617, to Gatekeeper 1, are shown in the next step. display signaling-group 5 SIGNALING GROUP Group Number: 5 Group Type Remote Office? n SBS? n Group Type: h.323 Max number of NCA TSC: 0 Max number of CA TSC: 0 Trunk Group for NCA TSC: Trunk Group for Channel Selection: 617 Supplementary Service Protocol: b Network Call Transfer? n Near-end Node Name: GK2-CLAN1 Far-end Node Name: GK1-CLAN1 Near-end Listen Port: 1719Far-end Listen Port: 1719Near-end Listen Port: 1719Far-end Listen Port: 1719LRQ Required? yCalls Share IP Signaling Connection? nRRQ Required? nH245 Control Addr On FACility? nMedia Encryption? nBypass If IP Threshold Exceeded? y DTMF over IP: in-band Direct IP-IP Audio Connections? y IP Audio Hairpinning? y Interworking Message: PROGress The configuration for the signaling group to the Avaya S8700 Server and G600 Gateway is shown here for comparison. The "Far-end Network Region" for the Avaya S8700 Server and G600 Gateway is 15. The screens for trunk group 619 are not shown, but are very similar to those of trunk group 617, shown in the next step. display signaling-group 1 SIGNALING GROUP Group Type: h.323 Group Number: 1 Remote Office? n SBS? n Max number of NCA TSC: 0 Max number of CA TSC: 0 Trunk Group for NCA TSC: Trunk Group for Channel Selection: 619 Supplementary Service Protocol: b Network Call Transfer? n Near-end Node Name: GK2-CLAN1 Far-end Node Name. FDAS Far-end Listen Port: 1719 Far-end Node Name: PBX3-CLAN1

 Near-end Listen Port: 1719
 Far-end Listen Port: 1719

 Near-end Listen Port: 1719
 Far-end Listen Port: 1719

 LRQ Required? y
 Calls Share IP Signaling Connection? n

 RRQ Required? n
 H245 Control Addr On FACility? n

 Bypass If IP Threshold Exceeded? y

 DTMF over IP: in-band Direct IP-IP Audio Connections? y IP Audio Hairpinning? y Interworking Message: PROGress

These screen shots show how the trunk group on Gatekeeper 2, that connects to 6. Gatekeeper 1, was configured. display trunk-group 617 1 of 22 Page TRUNK GROUP Dup Number:617Group Type:isdnCDR Reports:yGroup Name:IP Trunk to GK1COR:1TN:1TAC:13Direction:two-wayOutgoing Display?nCarrier Medium:IPial Access?nPure Thurstylla access?1T Group Number: 617 Group Name: IP Trunk to GK1 TAC: 139 Dial Access? n Busy Threshold: 255 Night Service: Queue Length: 0 Service Type: tie Auth Code? n TestCall ITC: rest Far End Test Line No: TestCall BCC: 4 TRUNK PARAMETERS Codeset to Send Display: 6 Codeset to Send National IEs: 6 Max Message Size to Send: 260 Charge Advice: none Supplementary Service Protocol: b Digit Handling (in/out): enbloc/enbloc QSIG Value-Added? n Trunk Hunt: cyclical Digital Loss Group: 18 ing Number - Delete: Insert: Bit Rate: 1200 Synchronization: async Incoming Calling Number - Delete: Format: Duplex: full Disconnect Supervision - In? y Out? n Answer Supervision Timeout: 0 display trunk-group 617 2 of 22 Page TRUNK FEATURES Measured: none ACA Assignment? n Wideband Support? n Internal Alert? n Maintenance Tests? y ata Restriction? n NCA-TSC Trunk Member: Send Name: y Send Calling Number: y Data Restriction? n Used for DCS? n Hop Dgt? n Suppress # Outpulsing? n Format: public UUI IE Treatment: service-provider Outgoing Channel ID Encoding: preferred Replace Restricted Numbers? n Replace Unavailable Numbers? n Send Connected Number: y Modify Tandem Calling Number? n Send UUI IE? y Send UCID? n Send Codeset 6/7 LAI IE? y Path Replacement with Retention? n Path Replacement Method: better-route SBS? n Network (Japan) Needs Connect Before Disconnect? n 6 of 22 display trunk-group 617 Page TRUNK GROUP Administered Members (min/max): 1/10 GROUP MEMBER ASSIGNMENTS Total Administered Members: 10 Port Code Sfx Name Night Sig Grp 1: T00005 5 2: T00006 5 3: T00007 5 4: T00008 5

7. Call and Bandwidth limits are configured on page 3 of the IP Network Region form. Gatekeeper 2 is logically associated with Network Region 14. On Page 1 of the IP Network Region, set "Intra-region IP-IP Direct Audio" and "Inter-region IP-IP Direct Audio" to "yes. Set the "IP Audio Hairpinning" field to "y". change ip-network-region 14 Page 1 of 19 IP NETWORK REGION Region: 14 Location: Home Domain: Name: Intra-region IP-IP Direct Audio: yes AUDIO PARAMETERS Inter-region IP-IP Direct Audio: yes Codec Set: 1 IP Audio Hairpinning? y UDP Port Min: 2048 UDP Port Max: 3028 RTCP Reporting Enabled? y RTCP MONITOR SERVER PARAMETERS DIFFSERV/TOS PARAMETERS Use Default Server Parameters? y Call Control PHB Value: 34 Audio PHB Value: 46 802.1P/Q PARAMETERS Call Control 802.1p Priority: 7 Audio 802.1p Priority: 6 AUDIO RESOURCE RESERVATION PARAMETERS H.323 IP ENDPOINTS RSVP Enabled? n H.323 Link Bounce Recovery? y Idle Traffic Interval (sec): 20 Keep-Alive Interval (sec): 5 Keep-Alive Count: 5 On Page 3 specify the limits between the direct regions controlled by Gatekeeper 2. What is shown here is that 2 calls will be allowed between Region 14 and Region 13; and 3 calls between Region 14 and Region 15. Specify the limit by bandwidth (e.g. 256:Kbits). When using bandwidth, take into consideration the codec used and the packet size; for example G.711 with 20 ms packets use approximately 83 Kbits/sec. For a detailed description of bandwidth calculations check the Administrator's Guide for Avaya Communication Manager [1], or Appendix 1. change ip-network-region 14 3 of 19 Page Inter Network Region Connection Management src dst codec direct Dynamic CAC rgn rgn set WAN WAN-BW-limits Intervening-regions Gateway 14 1 14 2 14 3 14 4 5 14 14 б 14 7 14 8 14 9 14 10 14 11 14 12 2:Calls 14 13 1 v 14 14 1 3:Calls 14 15 1 У

0	Connacti	uity ha	twoon D	agiona 12 and	15 was indirect. The	following	aaraar	a show Dogo
0.							screen	is show rage
	3 of regio	$\frac{13}{5}$	and 15,	snowing the in	direct configuration,	and the in	terven	ing region
	being 14.	For ex	ample 1	f a call is to go	from region 13 to 15	, the limits	s to go	from 13 to
	14 and 14	to 15	will app	oly.		-	2 6	1.0
	display i	o-netwo	rk-regio	n 13		Page	3 of	19
			Inter	Network Region	Connection Managemen	.t		
	era det	codec	direct			Dunamia C	۸C	
	rgn rgn	set	WAN	WAN-BW-limits	Intervening-regions	Gateway	nc	
	13 1							
	13 2 13 3							
	13 4							
	13 5 12 6							
	13 6							
	13 8							
	13 9 13 10							
	13 11							
	13 12	-						
	13 13 13 14	1	v	2:Calls				
	13 15	1	n	2.00115	14: : :			
	display ip	p-netwo	rk-regio	n 15		Page	3 of	19
			Inter	Network Region	Connection Managemen	t		
	src dst	codec	direct	WAN_BW_limita	Intervening-regions	Dynamic C	AC	
	15 1	bet	WAIN	WAN-DW-IIIIICS	Incervening regions	Gateway		
	15 2							
	15 3 15 4							
	15 5							
	15 6							
	15 7 15 8							
	15 9							
	15 10							
	15 11 15 12							
	15 13	1	n		14: : :			
	15 14 15 15	1	У	3:Calls				
0	Automati		o Solact	ion was used to	route calls to the ope	propriate 1	ocatio	ne Thie
у.	Automati			I was used it	tio Alternate Dauting	ond a U	former	Dial Dlar
	solution v	vouid a	uso wor	k with Automa	tic Alternate Routing	and a Un	llorm	Dial Plan.
	Since the	se are s	standard	routine config	urations, additional d	etail was i	not dee	emed
	necessary	v. It is v	worth re	-emphasizing h	ere that in this config	guration th	e end	PBXs only
	had signa	ling gr	oups, tr	unk groups, and	l route patterns to the	eir respecti	ve Ga	tekeepers. In
	a large co	onfigura	ation thi	s significantly	reduces administratio	on required	l for tr	unks,
	signaling	groups	s, and ro	outing at each e	nd PBX. All call rout	ing takes 1	blace b	by the
	gatekeene	ers.		0				-
	8p							

3.3. Configuring the Avaya DEFINITY® Server R, S8700 Servers, MCC1 Media Gateways, and G600 Media Gateway acting as PBX Nodes

Configuration of the PBX nodes involves similar steps to configuring the Gatekeepers, with the exception of the Call Admission Control (CAC) entries in the network regions. Connection management and CAC limitations are only done at the Gatekeepers, therefore the third page of the network regions is left empty on the PBX nodes.

Follow steps 1 through 4 of the previous sections to administer the Node Names, IP interfaces, and Data Channels. Note that in early software versions the "Change IP-Interface" command was used to bring up a form that allowed the input of multiple IP Interfaces at the same time. In later software versions, each IP Interface was configured separately as shown in section 3.1, steps 2 and 3.

On Avaya G3R Server 1 the following trunk and signaling groups must be configured to match the configuration on Gatekeeper 1:

- Trunk Group 614 and its corresponding signaling group
- Trunk Group 615 and its corresponding signaling group

On Avaya G3R Server 2 the following trunk and signaling groups must be configured to match the configuration on Gatekeeper 1:

• Trunk Group 614 and its corresponding signaling group

On the Avaya S8700 Server the following trunk and signaling groups must be configured to match the configuration on Gatekeeper 2:

• Trunk Group 619 and its corresponding signaling group

The following steps show the configuration screens for the IP Interfaces, trunk groups and signaling groups on the Avaya S8700 server. The configurations on the Avaya DEFINITY® Server R is done in a similar manner. Also note that there may be minor screen changes between Avaya Communication Manager software versions. The screens shown here correspond to Software Version 1.3.

1.	The older version of the IP Interfaces form is shown here. The "ON" field should be set
	to "y" to enable the IP Interface. The "Type" field is the interface type, "C-LAN" or
	"MEDPRO". The "Slot" field is the card location. The "Code" and "Sfx" fields are
	automatically populated when the "Slot" is entered. The "Node Name" is the name
	association for the interface in the "IP Node Names" table (see sample screen in Section
	3.1, step 1). The "Subnet Mask" and "Gateway Address" are the respective mask and
	gateway for the IP Interface. The "Net Rgn" is the network region associated with the IP
	Interface, in this case 15. The "VLAN" is the VLAN associated with the IP Interface, in
	this case 20.

change ip-interfaces		I	Page	1	of	19
IP I	NTERFACES					
					Net	:
ON Type Slot Code Sfx Node Name	Subnet Mask	Gateway	Addr	ress	Rgn	VLAN
y C-LAN 01A04 TN799 D SGINT-CLAN1	255.255.255.0	20 .1	.1	.1	15	20
Y MEDPRO 01A05 TN2302 SGINT-MEDPRO1	255.255.255.0	20 .1	.1	.1	15	20
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0					n
n	255.255.255.0		•			n
n	255.255.255.0		•			n
n	255.255.255.0		•			n

2. There is one signaling group between the Avaya S8700 Server and Gatekeeper 2. When adding a signaling group, the "Trunk Group for Channel Selection" field must be left blank until the trunk group is configured. After the trunk group is configured use the "change signaling-group 1" command to populate the trunk group number. The "Supplementary Service Protocol" field was set to "b" to use QSIG signaling. The "Near-end Node Name" field should be set to the node name of a local C-LAN. The "Far-end Node Name" field should be set to the node name of a remote C-LAN. The "Near-End Listen Port" and "Far End Listen Port" should be "1719". The "Far-End Network Region" associated with Gatekeeper 2 is "14". "LRQ Required" should be set to "y".

SIGNA.	
Group Number: 1 Group T Remote Off Trunk Group for Channel Select Supplementary Service Proto	ype: h.323 ice? n Max number of NCA TSC: 0 SBS? n Max number of CA TSC: 0 Trunk Group for NCA TSC: ion: 619 col: b
Near-end Node Name: SGINT-CLA Near-end Listen Port: 1719 LRQ Required? y RRQ Required? n	N1 Far-end Node Name: GK2-CLAN1 Far-end Listen Port: 1719 Far-end Network Region: 14 Calls Share IP Signaling Connection? n H245 Control Addr On FACility? n Bypass If IP Threshold Exceeded? y
DTMF over IP: in-band	Direct IP-IP Audio Connections? y IP Audio Hairpinning? y Interworking Message: PROGress

These screen shots show how the trunk group on the Avaya S8700 Server that connects 3. to Gatekeeper 2 was configured. display trunk-group 619 1 of 22 Page TRUNK GROUP Group Type: isdn CDR Reports: y Group Name: IP Trunk to GK2 COR: 1 TN: 1 TAC: 141 Direction: two-way Outgoing Display? n Carrier Medium: IP ial Access? n Busy Threshold: 255 Night Service: Sue Length: 0 Group Number: 619 Group Name: IP Trunk to GK2 Direction Dial Access? n Queue Length: 0 Service Type: tie Auth Code? Far End Test Line No: Auth Code? n TestCall ITC: rest TestCall BCC: 4 TRUNK PARAMETERS Codeset to Send Display: 6 Codeset to Send National IEs: 6 Max Message Size to Send: 260 Charge Advice: none Supplementary Service Protocol: b Digit Handling (in/out): enbloc/enbloc Trunk Hunt: cyclical QSIG Value-Added? n Digital Loss Group: 18 Digital Loss Group: 18Calling Number - Delete:Insert:Numbering Format:Bit Rate: 1200Synchronization: asyncDuplex: full Disconnect Supervision - In? y Out? n Answer Supervision Timeout: 0 display trunk-group 619 Page 2 of 22 TRUNK FEATURES Measured: none Wideband Support? n Internal Alert? n Maintenance Tests? y Data Restriction? n NCA-TSC Trunk Member: ACA Assignment? n Wideband Support? n Send Name: y Send Calling Number: y Hop Dgt? n Used for DCS? n Hop Dgt? n Suppress # Outpulsing? n Numbering Format: public Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider Replace Restricted Numbers? n Replace Unavailable Numbers? n Send Connected Number: y Send UUI IE? y Send UCID? n Send Codeset 6/7 LAI IE? y Path Replacement with Retention? n Path Replacement Method: better-route SBS? n Network (Japan) Needs Connect Before Disconnect? n display trunk-group 619 Page 6 of 22 TRUNK GROUP Administered Members (min/max): 1/10 GROUP MEMBER ASSIGNMENTS Total Administered Members: 10 Code Sfx Name Night Port Sig Grp 1: T00001 1 2: T00002 1 3: T00003 1 4: T00004 1

4. This screen shot shows the route pattern used for routing calls between UK1 and US1. Trunk group 619 is the first preference and trunk group 620 is the second preference. "LAR" is enabled on the route pattern, and set to "next". This allows calls to re-route over the second trunk group if the first one is unavailable or calls over the first trunk group are rejected.

isplay route-patt	ern 1		P	age 1	of 3
	Pattern Number: 1	Pattern Name: In-Co	untry E	xt	
Grp FRL NPA Pf	x Hop Toll No. Inse	erted		DCS	/ IXC
No Mr	rk Lmt List Del Dig:	its		QSI	G
	Dgts			Int	W
1: 619 0 1	L			n	user
2: 620 0 1				n	user
3:				n	user
4:				n	user
5:				n	user
6:				n	user
BCC VALUE TS	SC CA-TSC ITC BCI	E Service/Feature BAND	No. N	umbering	LAR
01234W	Request		Dqts F	ormat	
	-	Su	baddres	S	
1: ууууул г	n both ept	t			next
2: ууууул г	n both ept	t			none
3: ууууул г	n rest				none
4: yyyyyn n	n rest				none
5: yyyyyn r	n rest				none
6: ууууул г	n rest				none

4. Verification Steps

To verify if the trunks are up and working, use the "Status Trunk" command in Avaya Communication Manager. For example, "status trunk 617". For trunk ports with active calls the status should be "in-service/active". For idle trunk ports the status should be "in-service/idle".

To verify calls are routing properly over the appropriate trunks, use the Avaya Communication Manager "Status Trunk" command, or the "Monitor Trunk" command.

To verify calls are using the appropriate codecs, use the Avaya Communication Manager "status station" command, when the station is active on a call.

To verify that the Gatekeepers are properly managing the limits specified for the network regions, place calls between the regions, and when the limits are reached, the call should re-route over an alternate path if LAR is configured, or fast busy should be heard if no alternate path is available.

5. Conclusion

As illustrated by these Application Notes, Avaya Communication Manager can be used in a Centralized Gatekeeper environment to minimize the administrative complexity for enterprises deploying a large and growing number of IP-enabled PBXs and call servers.

Appendix 1

The following table can be used to help assess how much bandwidth (in Kbits/sec) is used for various types of codecs and packet sizes. The values in this table assume a 7 byte L2 WAN header and are rounded up.

Packet Size	10 ms	20 ms	30 ms	40 ms	50 ms	60 ms
G.711	102	83	77	74	72	71
G.729	46	27	21	18	16	15
G.723-6.3	NA	NA	19	NA	NA	13
G.723-5.3	NA	NA	18	NA	NA	12

The values shown here were derived from the Administrator's Guide for Avaya Communication Manager, Issue 8, June 2004. Refer to the Administrator's Guide for additional detail on these calculations.

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