



Avaya Solution & Interoperability Test Lab

Configuring Avaya Communication Manager Release 4.0 To Use Secure Real-Time Transport Protocol (SRTP) over H.323 IP Trunks – Issue 1.0

Abstract

These Application Notes illustrate the configuration and operation of the Secure Real-Time Transport Protocol (SRTP) introduced in Avaya Communication Manager Release 4.0. The sample configuration described in these Application Notes includes two independent Avaya Communication Manager systems networked using H.323 IP Trunks. One site uses an Avaya S8720 Media Server pair controlling two Avaya G650 Media Gateway cabinets. Another site uses an Avaya S8300 Media Server with an Avaya G350 Media Gateway. Each site has a mix of Avaya 4600-Series IP Telephones, Avaya 9600-Series IP Telephones, and non-IP telephones such as Avaya 2400-Series Digital Telephones. These Application Notes focus on illustrating the configuration of the H.323 IP Signaling Groups, Trunk Groups, and IP codec sets for secure inter-system connectivity using SRTP. Although not the focus of these Application Notes, the configuration also enables Avaya 9600-Series IP Telephones to communicate directly with one another across the H.323 IP Trunk using the G.722 wideband codec, another capability introduced in Avaya Communication Manager Release 4.0.

1. Introduction and Scope

These Application Notes illustrate the configuration and operation of the Secure Real-Time Transport Protocol (SRTP) introduced in Avaya Communication Manager Release 4.0. The sample configuration described in these Application Notes includes two independent Avaya Communication Manager systems networked using H.323 IP Trunks. One site uses an Avaya S8720 Media Server pair controlling two Avaya G650 Media Gateway cabinets. Another site uses an Avaya S8300 Media Server with an Avaya G350 Media Gateway. Each site has a mix of Avaya 4600-Series IP Telephones, Avaya 9600-Series IP Telephones, and non-IP telephones such as Avaya 2400-Series Digital Telephones. These Application Notes focus on illustrating the configuration of the H.323 IP Signaling Groups, Trunk Groups, and IP codec sets for secure inter-system connectivity using SRTP. Although not the focus of these Application Notes, the configuration also enables Avaya 9600-Series IP Telephones to communicate directly with one another across the H.323 IP Trunk using the G.722 wideband codec, another capability introduced in Avaya Communication Manager Release 4.0.

Figure 1 provides an overview of the network used to verify these Application Notes. Note that the approach documented in these Application Notes could be used with other Avaya Media Servers and Media Gateways.

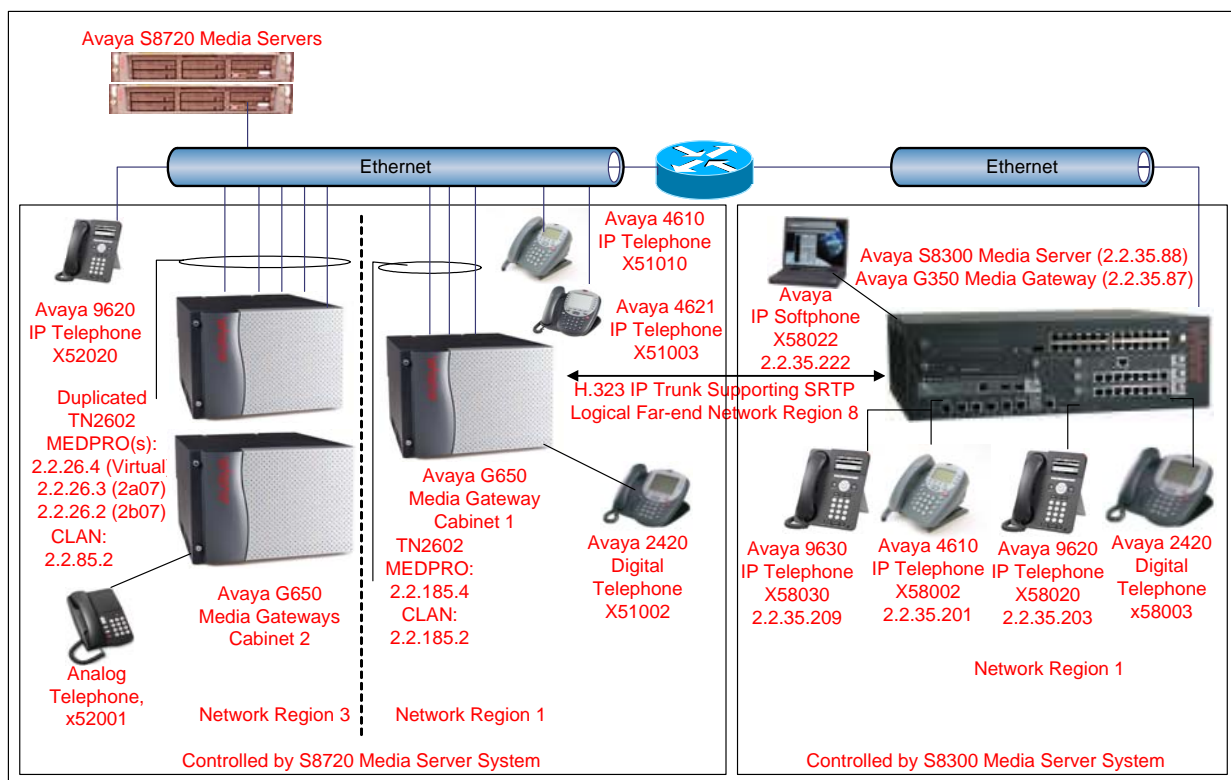


Figure 1: Network Overview

1.1. Example Codec / Security Objectives for the Figure 1 Network

For the purposes of illustrating the flexibility and considerations that apply to codec selection and encryption, these Application Notes assume the following high-level objectives for the **Figure 1** network. Of course, each customer's objectives will vary. Section 5 shows a representative sample of calls fulfilling these objectives:

- Media connections within any site (system) should be optimized for the highest quality audio and the strongest security natively supported by the communicating devices. By using the highest quality audio and strongest security natively supported by the devices, intra-site connections will utilize the fewest resources from media gateway VoIP processors. The intra-system objectives will be satisfied by configuring a codec set that includes G.722-64K and G.711MU, and prefers SRTP but allows AES media encryption.
- Media connections between sites (systems) will again be optimized for the highest quality audio codec, but SRTP will be required for connections between sites, even if satisfying the SRTP requirement means that media gateway resources must be utilized for the inter-system connectivity. That is, this objective assumes that calls between systems always require the strongest security procedures, independent of the native capabilities of the end devices. These objectives may be appropriate where the preponderance of calls are intra-system, and the less frequent inter-system calls have stringent security requirements, either given the nature of the calls or the networking path followed. These inter-system objectives will be satisfied by configuring a codec set with the same audio codec preferences used for intra-site calling, but with a requirement to use SRTP for the inter-system media connection. End-user devices incapable of native SRTP will have media processing resources assigned automatically to honor the requirement that SRTP be used for the inter-system connection, while still retaining AES encryption over the LAN (intra-system).

A five digit Uniform Dial Plan (UDP) is used so that any user may dial a five-digit number to reach any other user. Unique ranges of extensions are associated with each system as follows. The Avaya S8720 Media Server system uses local extensions in the range 51XXX and 52XXX. The Avaya S8300 Media Server system uses local extensions in the range 58XXX. The Avaya S8720 Media Server and Avaya S8300 Media Server systems are connected with an H.323 signaling group and trunk group. The UDP feature is configured so that inter-system calls use the H.323 IP trunk group.

Section 5 illustrates detailed status and call trace verifications to reinforce the behavior achieved using the relatively simple configuration presented in Section 4. Although the focus of these Application Notes is the inter-system connectivity over the H.323 IP Trunk, note that the procedures also result in encrypted intra-system connections among the IP Telephones and gateways controlled by the same Avaya Communication Manager.

1.2. Avaya Communication Manager Release 4.0 SRTP Summary

SRTP is a media encryption standard, defined in RFC 3711 (Reference [4]) as a profile of RTP. The Avaya Communication Manager Release 4.0 implementation of SRTP enables the following, with the SRTP codec profile illustrated in these Application Notes shown in parenthesis where applicable:

- Encryption of Real-time Transport Protocol (RTP) (e.g., AES-128 in counter mode)
- Authentication of RTP and Real-time Transport Control Protocol (RTCP) streams (e.g., HMAC-SHA1-80 keyed hash algorithm). An authentication value appended to the packet covers the entire RTP or RTCP packet. A receiver may authenticate a packet before decrypting the packet payload, minimizing the computational expense of processing a bad packet.
- Direct SRTP connectivity between devices natively supporting SRTP
- Capability harmonization among parties in a call, and insertion of gateway media processing resources where required

In addition to the Avaya G350 Media Gateway and TN2602 Media Processor illustrated in these Application Notes, the Avaya G250 Media Gateway and Avaya G700 Media Gateway also support SRTP. At time of writing, the TN2302 Media Processor, like the Avaya 4600-Series IP Telephones, support AES media encryption, but not SRTP. If the Avaya Integrated Management VoIP Monitoring Manager is used, note that the VoIP Monitoring Manager server will receive unencrypted, unauthenticated RTCP information, allowing use of the existing application to collect and display data pertaining to media connections that use SRTP. As this is primarily a configuration document, product documentation should be consulted for further details.

2. Equipment and Software Validated

Table 1 shows the equipment and version information used in the sample configuration. Note that the Avaya Communication Manager Release 4.0 load used was a beta version. Therefore, minor differences in screen content may be observed.

Network Component	Version Information
Avaya S8720 Media Servers	Avaya Communication Manager 4.0 Load 730.0
Avaya S8300 Media Server	Avaya Communication Manager 4.0 Load 730.0
Avaya G350 Media Gateway	26.28.0
Avaya TN799DP in G650 Media Gateways	HW01 FW012
Avaya TN2602AP in G650 Media Gateways	HW2 FW 30
Avaya 4600-Series IP Telephones (H.323)	2.7
Avaya 9600-Series IP Telephones (H.323)	1.2
Avaya IP Softphone	5.2.3.6

Table 1 – Equipment Version Information

3. Conventions, Assumptions, and Licensing

In these Application Notes, Avaya Communication Manager administration software screens are shown with a gray shaded background. These screens are also referred to as “SAT” (System Access Terminal) screens. In some instances, the information from the original screen has been edited or annotated for brevity or clarity in presentation. Unless otherwise noted, each screen capture is preceded by the text that references the screen capture.

Steps that aid in understanding the configuration, but are not central to the scope, are shown with “display” or “list” commands, to emphasize that the screens should not be interpreted as part of a procedurally complete configuration flow. Only aspects of the configuration related to configuring H.323 IP Trunks, codec selection, and SRTP are illustrated.

To verify that the installed license grants permission to use the features illustrated in these Application Notes, use the command “display system-parameters customer-options”. Verify sufficient H.323 trunk and IP station capacity are available. Note that the S8720 Media Server must be equipped with one or more TN2602 Media Processor circuit packs to use SRTP as shown in these Application Notes. The TN2302 Media Processor does not support SRTP.

display system-parameters customer-options		Page 2 of 10
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
	Maximum Administered H.323 Trunks: 50	48
	Maximum Concurrently Registered IP Stations: 10	3

The bolded fields shown on Page 4 are features that must be enabled in the license to achieve the configuration presented in these Application Notes.

display system-parameters customer-options		Page 4 of 10
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 Network Call Redirection? n
Enterprise Survivable Server? n		ISDN-BRI Trunks? n
Enterprise Wide Licensing? n		ISDN-PRI? y
ESS Administration? n		Local Survivable Processor? n
Extended Cvg/Fwd Admin? y		Malicious Call Trace? y
External Device Alarm Admin? n		Media Encryption Over IP? y
Five Port Networks Max Per MCC? n		Mode Code for Centralized Voice Mail? n
Flexible Billing? n		
Forced Entry of Account Codes? n		Multifrequency Signaling? y
Global Call Classification? y		Multimedia Call Handling (Basic)? n
Hospitality (Basic)? y		Multimedia Call Handling (Enhanced)? n
Hospitality (G3V3 Enhancements)? n		
	IP Trunks? y	

Paging forward, the bolded fields in the following screen show the routing features that must be enabled in the license to achieve the UDP configuration presented in these Application Notes.

display system-parameters customer-options		Page 5 of 10
OPTIONAL FEATURES		
Multinational Locations? n	Station and Trunk MSP? y	
Multiple Level Precedence & Preemption? n	Station as Virtual Extension? n	
Multiple Locations? y		
Personal Station Access (PSA)? y	System Management Data Transfer? y	
Posted Messages? y	Tenant Partitioning? n	
PNC Duplication? n	Terminal Trans. Init. (TTI)? y	
Port Network Support? n	Time of Day Routing? y	
	Uniform Dialing Plan? y	
Processor and System MSP? y	Usage Allocation Enhancements? y	
Private Networking? y	TN2501 VAL Maximum Capacity? y	
Processor Ethernet? y	Wideband Switching? n	

If the license does not grant sufficient permissions or capacity, contact your authorized Avaya sales representative.

4. Avaya Communication Manager Configuration

This section illustrates the relevant Avaya Communication Manager configuration. Since the procedures to configure the Avaya S8300 Media Server and Avaya S8720 Media Server for the H.323 IP Trunks are nearly identical, only the Avaya S8300 Media Server configuration screens are presented here. The Avaya S8720 Media Server configuration is illustrated in detail in Reference [1].

4.1. Node Names

Node names are logical mappings of names to IP addresses that are used in other configuration screens, such as the near-end and far-end of an H.323 signaling group. The following illustrates the relevant node names on the Avaya S8300 Media Server. The node name “procr” refers to the Avaya S8300 Media Server processor Ethernet. The node name “CLAN185-s8720” refers to a CLAN in an Avaya G650 Media Gateway controlled by the Avaya S8720 Media Servers.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
CLAN185-s8720	2.2.185.2	
procr	2.2.35.88	

4.2. H.323 Signaling Group and Trunk Group

Configure the signaling group using the “add signaling group” command. In this case, signaling group 9 is created. The parameters relevant to these Application Notes are shown in bold below.

The **Group Type** is set to “h.323”. The **Near-end Node Name** is set to “procr”, the S8300 Media Server processor Ethernet. The **Far-end Node Name** is set to “CLAN185-s8720”, the node name assigned to a CLAN circuit pack in the Avaya G650 Media Gateway. When configuring the corresponding H.323 signaling group on the S8720 Media Server, the near-end

will be the CLAN, and the far-end will be the node name of the S8300 Media Server. The **Near-end Listen Port** and **Far-end Listen Port** are set to 1719, consistent with the use of **LRQ Required**. With this configuration, a Location Request (LRQ) is sent to the far-end before each new call using the signaling group. A positive Location Confirm (LCF) is necessary before the call is routed to the corresponding trunk group. Although not illustrated in these Application Notes, the failure to receive an LCF in response to LRQ can be used to trigger look-ahead routing to alternate trunk groups.

The **Media Encryption** field is set to “y”, and a **Passphrase** is entered. The configured **Passphrase** must match on the Avaya S8300 Media Server and Avaya S8720 Media Server representing the near-end and far-end of this signaling group. The **H.235 Annex H Required** field is set to “y” to enhance security of control signaling. The **Far-end Network Region** parameter is set to 8, a network region that is different from the near-end devices on the Avaya S8300 Media Server. By assigning a unique network region to the far-end of the signaling group, different codec and media encryption options may be used for calls that use the H.323 signaling group and trunk group.

```

add signaling-group 9                                     Page 1 of 1
                                     SIGNALING GROUP

Group Number: 9                Group Type: h.323
                                Remote Office? n           Max number of NCA TSC: 10
                                SBS? n                     Max number of CA TSC: 10
                                IP Video? n               Trunk Group for NCA TSC: 9
                                Trunk Group for Channel Selection:
                                TSC Supplementary Service Protocol: b
                                T303 Timer(sec): 10

Near-end Node Name: procr      Far-end Node Name: CLAN185-s8720
Near-end Listen Port: 1719    Far-end Listen Port: 1719
                                Far-end Network Region: 8
                                Calls Share IP Signaling Connection? n
                                LRQ Required? y
                                RRQ Required? n
                                Media Encryption? y        Bypass If IP Threshold Exceeded? n
                                Passphrase: *              H.235 Annex H Required? y
                                DTMF over IP: out-of-band  Direct IP-IP Audio Connections? y
                                Link Loss Delay Timer(sec): 90 IP Audio Hairpinning? y
                                Enable Layer 3 Test? y      Interworking Message: PROgress
                                                                DCP/Analog Bearer Capability: 3.1kHz

```

The command “add trunk-group” can be used to add a trunk group to be associated with the H.323 signaling group. The use of SRTP does not affect the trunk group configuration. Therefore, details are omitted.

```

add trunk-group 9                                       Page 1 of 21
                                     TRUNK GROUP

Group Number: 9                Group Type: isdn           CDR Reports: y
Group Name: To-S8720-Right     COR: 1                    TN: 1          TAC: 109
Direction: two-way            Outgoing Display? y      Carrier Medium: H.323
Dial Access? n                Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie              Auth Code? n
                                Member Assignment Method: manual

```

In the sample configuration, four trunk members were added, allowing four simultaneous calls in progress using the trunk group. The keyword “ip” is entered in the **Port** field, and the H.323 signaling group number is entered in the **Sig Grp** field. After submitting the form, Avaya Communication Manager assigns a unique trunk identifier to each trunk member. In the sample configuration, Avaya Communication Manager assigned trunk identifiers T00037 – T00040. These trunk identifiers can be observed in the verification screens shown in Section 5.

add trunk-group 9		Page 5 of 21		
TRUNK GROUP				
		Administered Members (min/max):	0/0	
GROUP MEMBER ASSIGNMENTS		Total Administered Members:	0	
Port	Name	Night	Sig Grp	
1: ip			9	
2: ip			9	
3: ip			9	
4: ip			9	

Return to the signaling group configuration with the command “change signaling-group 9”. In the field **Trunk Group for Channel Selection**, enter the number of the trunk group to be associated with incoming calls on this signaling group. In the sample configuration, trunk group 9 is used.

change signaling-group 9		Page 1 of 1	
SIGNALING GROUP			
Group Number: 9	Group Type: h.323		
	Remote Office? n	Max number of NCA TSC: 10	
	SBS? n	Max number of CA TSC: 10	
IP Video? n		Trunk Group for NCA TSC: 9	
Trunk Group for Channel Selection: 9			
TSC Supplementary Service Protocol: b	T303 Timer(sec): 10		
Near-end Node Name: procr	Far-end Node Name: CLAN185-s8720		
Near-end Listen Port: 1719	Far-end Listen Port: 1719		
	Far-end Network Region: 8		
LRQ Required? y	Calls Share IP Signaling Connection? n		
RRQ Required? n			
Media Encryption? y	Bypass If IP Threshold Exceeded? n		
Passphrase: *	H.235 Annex H Required? y		
DTMF over IP: out-of-band	Direct IP-IP Audio Connections? y		
Link Loss Delay Timer(sec): 90	IP Audio Hairpinning? y		
Enable Layer 3 Test? y	Interworking Message: PROGRESS		
	DCP/Analog Bearer Capability: 3.1kHz		

4.3. Assignment of Endpoints to Network Regions

IP telephones can be explicitly assigned a network region based on an IP address mapping, or can derive a network region from the region of the entity (e.g., CLAN, processor Ethernet) to which the telephone registers. The following screen illustrates a subset of the IP network map configuration used to verify these Application Notes. In the sample configuration, all the Avaya IP Telephones registered to the Avaya S8300 Media Server are assigned to network region 1. For telephones registered to the Avaya S8720 Media Server, some are mapped to network region 1 and some are mapped to network region 3, as detailed in Reference [1].

```
change ip-network-map                                     Page 1 of 32
```

IP ADDRESS MAPPING							Emergency
From IP Address	(To IP Address	Subnet	Region	VLAN	Location	Extension	
	or Mask)						
2 .2 .35 .200	2 .2 .35 .209		1	n			

Non-IP telephones (analog, digital) derive a network region from the region of the cabinet or gateway to which the telephone is connected. For example, digital station 58003 is in network region 1 because the Avaya G350 Media Gateway to which it is connected is in network region 1. When a non-IP device such as a digital phone makes a call that is routed over an H.323 IP Trunk Group, a media processing resource (e.g., TN2602 in the Avaya G650 Media Gateway, or the integrated VoIP processing of an Avaya G350 Media Gateway) is required. These media processing resources are capable of SRTP.

The following screen shows the overall configuration of the Avaya G350 Media Gateway. Note that link encryption is used, and that the gateway is configured in network region 1.

```
display media-gateway 1
```

MEDIA GATEWAY		
Number: 1	Registered? y	
Type: g350	FW Version/HW Vintage: 26 .28 .0 /1	
Name: G350-Right	IP Address: 2 .2 .35 .87	
Serial No: 03IS07589449	Controller IP Address: 2 .2 .35 .88	
Encrypt Link? y	MAC Address: 00:04:0d:29:c9:8d	
Network Region: 1		
Location: 1	Site Data:	
Recovery Rule: none		
Slot	Module Type	Name
V1:	S8300	ICC MM
V2:		
V3:		
V4:	MM712	DCP MM
V5:		
V6:	MM314	ETH 24P MM
V7:	1T+2L-Integ-Analog	ANA IMM Max Survivable IP Ext: 8
V8:		
V9:	gateway-announcements	ANN VMM

4.4. Network Region Configuration

The following screens illustrate the configuration for network region 1. As shown in the previous section, all the endpoint devices served by the Avaya S8300 Media Server are in network region 1. From Page 1, observe that the **Codec Set** to be used for connections within region 1 is set to “1”.

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

The following screen illustrates Page 2 for network region 1. The **H.323 Security Profiles** list uses “any-auth” to allow for the procedures appropriate for the device. However, “pin-eke” can be used to require the stronger security procedures. As evidenced in Section 5, “any-auth” will result in Annex H procedures for the Avaya 4600-Series and Avaya 9600-Series IP Telephones running the firmware shown in **Table 1**.

```
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  any-auth
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
```

The following screen illustrates Page 3 for network region 1, which defines the inter-network region connectivity. The focus of these Application Notes is the connectivity between “local” network region 1 endpoints and network region 8, the region assigned as the “far-end” of the H.323 Signaling Group. These connections will use codec set 2.

```
change ip-network-region 1                                     Page 3 of 19
Inter Network Region Connection Management
src dst codec direct  Total          Video          Dyn
rgn rgn set   WAN  WAN-BW-limits  Norm Prio  Shr Intervening-regions  CAC IGAR
1   1   1
1   2   3   y      :NoLimit          Y          n
1   3   1   y      :NoLimit          Y          n
1   4
1   5
1   6
1   7   7   y      :NoLimit          Y          n
1   8   2   y      :NoLimit          n          n
```

The foregoing assignment of the interconnectivity between network region 1 and 8 automatically creates a symmetrical configuration for region 8. This is illustrated with the display command shown below. Other network region 8 default parameters also apply.

```
display ip-network-region 8                                   Page 3 of 19
Inter Network Region Connection Management
src dst codec direct  Total          Video          Dyn
rgn rgn set   WAN  WAN-BW-limits  Norm Prio  Shr Intervening-regions  CAC IGAR
8   1   2   y      :NoLimit          n          n
```

4.5. Codec Set Configuration

The network region configuration allows configuration of a codec set to be used for intra-region connections, and the option of a different codec set to be used for inter-region connections for any given network region pair. In these Application Notes, the intent for intra-region connections is to use the best quality codec and strongest encryption methods supported natively by the communicating devices. Therefore, codec set 1, used for intra-region connections, is configured as shown in the following screen. Observe that the profile “1-srtp-aescm128-hmac80” as well as “aes” are listed as options under the **Media Encryption** heading.

```
change ip-codec-set 1                                         Page 1 of 2
IP Codec Set
Codec Set: 1
Audio          Silence      Frames      Packet
Codec          Suppression  Per Pkt     Size(ms)
1: G.722-64K   n           2           20
2: G.711MU    n           2           20
3:
4:
5:
6:
7:
Media Encryption
1: 1-srtp-aescm128-hmac80
2: aes
3:
```

In these Application Notes, calls over the H.323 Signaling Group are configured to require SRTP while still using the best quality codec supported by the SRTP-communicating devices. Therefore, codec set 2, used for inter-system connections between network region 1 and network region 8, is configured as shown in the following screen. Observe that the profile “1-srtp-aescm128-hmac80” is the only option listed under the **Media Encryption** heading.

```
change ip-codec-set 2                                     Page 1 of 2
                IP Codec Set
Codec Set: 2
Audio          Silence      Frames   Packet
Codec          Suppression  Per Pkt  Size(ms)
1: G.722-64K   n                2        20
2: G.711MU     n                2        20
3:
4:
5:
6:
7:
Media Encryption
1: 1-srtp-aescm128-hmac80
2:
3:
```

4.6. Call Routing to the H.323 IP Trunk Group

Although there is nothing unique about routing due to the use of SRTP, the following screens illustrate the call routing configuration to the H.323 IP trunk group for completeness. This may aid in understanding the verification traces in Section 5.

Any call of the form 51XXX or 52XXX is routed to AAR after internal conversion of the dialed number to 111-1XXX or 111-2XXX respectively.

```
display uniform-dialplan 5                               Page 1 of 2
                UNIFORM DIAL PLAN TABLE
                Percent Full: 0
Matching      Len Del      Insert      Node
Pattern       Len Del      Digits      Net Conv Num
51            5 1        111        aar n
52            5 1        111        aar n
```

AAR analysis routes the call to Route Pattern 111 as shown below.

```
display aar analysis 111                                Page 1 of 2
                AAR DIGIT ANALYSIS TABLE
                Percent Full: 3
Dialed        Total      Route      Call      Node      ANI
String        Min Max    Pattern    Type      Num      Reqd
111          7 7        111        aar      n
```

Route Pattern 111 contains trunk group 9, the H.323 trunk group. The digits are manipulated such that the original dialed digits (51XXX or 52XXX) are sent to the far-end as the called party

number. The QSIG configuration also apparent in the screen below is not the subject of these Application Notes.

```

display route-pattern 111                                     Page 1 of 3
                    Pattern Number: 111 Pattern Name: to S8720
                    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: 9      0                3  5                n  user
2:                3  5                n  user
3:                3  5                n  user
4:                3  5                n  user
5:                3  5                n  user
6:                3  5                n  user

  BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM  No. Numbering LAR
  0 1 2 M 4 W      Request          Dgts Format
                    Subaddress
1: y y y y y n  y  as-needed  bothept          unk-unk  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

```

4.7. Saving Configuration Changes

The command “save translation all” can be used to save configuration changes.

5. Verification

The illustrated configuration has been verified. The following subsections illustrate the expected behavior for a representative sample of calls using the configuration presented in these Application Notes. Unless otherwise noted, all SAT screens are from the Avaya S8300 Media Server.

The following screen illustrates the registered IP stations, versions, and network regions. At the time of this screen capture, an Avaya IP Softphone is registered as extension 58022.

```
list registered-ip-stations
```

REGISTERED IP STATIONS							
Station Ext/ Orig Port	Set Type	Product ID	Prod Rel	Station IP Address	Net Rgn	Gatekeeper IP Address	TCP Skt
58002	4610	IP_Phone	2.700	2.2.35.201	1	2.2.35.88	y
58020	9620#	IP_Phone	1.200	2.2.35.203	1	2.2.35.88	y
58022	4620	IP_Soft	5.236	2.2.35.222	1	2.2.35.88	y
58030	9630#	IP_Phone	1.200	2.2.35.209	1	2.2.35.88	y

The following screen shows that the Avaya G350 Media Gateway is registered, and is in network region 1. The IP Address of the Avaya G350 Media Gateway is 2.2.35.87. This IP Address will feature prominently in the verifications.

```
list media-gateway
```

MEDIA-GATEWAY REPORT							
Num	Name	Serial No/ FW Ver/HW	Vint	IP Address/ Cntl IP Addr	Type	NetRgn	Reg? RecRule
1	G350-Right	03IS07589449	2	.2 .35 .87	g350	1	y
		26 .28 .0 /1	2	.2 .35 .88			none

The following screen illustrates the “in-service/idle” initial status of the H.323 IP Trunks.

```
status trunk 9
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0009/001	T00037	in-service/idle	no
0009/002	T00038	in-service/idle	no
0009/003	T00039	in-service/idle	no
0009/004	T00040	in-service/idle	no

The following screen, taken from the S8720 Media Server, illustrates the registered IP stations and corresponding network regions for the telephones included in the verification of these Application Notes. Reference [1] contains additional information on the S8720 Media Server configuration.

```
list registered-ip-stations
```

REGISTERED IP STATIONS							
Station Ext/ Orig Port	Set Type	Product ID	Prod Rel	Station IP Address	Net Rgn	Gatekeeper IP Address	TCP Skt
51003	4621	IP_Phone	2.700	2.2.185.200	1	2.2.85.20	y
51010	4610	IP_Phone	2.700	2.2.1.110	1	2.2.85.20	y
52020	9620#	IP_Phone	1.200	2.2.1.103	3	2.2.185.2	y

The following screen shows the network regions assigned to the IP Media Processors on the Avaya S8720 Media Server configuration detailed in Reference [1]. No unique configuration of the TN2602 Media Processor is required to use SRTP. These IP addresses will appear in the verification screens below. Cabinet 1 contains a TN2602 (2.2.185.4) configured for network region 1. Cabinet 2 contains a duplicated TN2602 pair (active/standby) configured for network region 3, where the virtual IP address of the pair is 2.2.26.4 as shown below.

```
list ip-interface medpro
```

IP INTERFACES									
ON	Slot	Code	Sfx	IP Address/Mask	Gateway Address	Net Rgn	VLAN	Virtual Address	Address
y	02A07	TN2602		2.2.26.3 /24	2.2.26.1	3	n	2.2.26.4	
y	02B07	TN2602		2.2.26.2 /24	2.2.26.1	3	n	2.2.26.4	
y	01A11	TN2602		2.2.185.4 /24	2.2.185.1	1	n		

5.1. Extension 58020 (9620) Calls Extension 52020 (9620)

This section illustrates the behavior when an Avaya 9600-Series IP Telephone (x58020) calls another Avaya 9600-Series IP Telephone (x52020), and the call is routed over the H.323 IP Trunk. Since the Avaya 9600-Series IP Telephones support G.722-64K, this preferred codec is used. Also, since the Avaya 9600-Series IP Telephones support SRTP, the media path can “shuffle” to “ip-direct”, meaning that no media processing resources from the gateways are required to ensure that SRTP is used over the IP Trunk. The following screen shows the annotated trace output for such a call, taken from the Avaya S8300 Media Server. From this trace, it can be observed that the call is routed using UDP and AAR to route pattern 111 and trunk group 9, the H.323 trunk group.

```
list trace station 58020
```

LIST TRACE		Page	1
time	data		
14:41:01	active station 58020 cid 0x15f		
	/** Telephone at 2.2.35.203 draws G711MU dialtone from G350 Gateway VoIP 2.2.35.87 **/		
14:41:01	G711MU ss:off ps:20 rn:1/1 2.2.35.203:2178 2.2.35.87:2054		
14:41:01	xoip: fax:Relay modem:off tty:US 2.2.35.87:2054 uid:0x954		
14:41:04	dial 52020 route:UDP AAR		
14:41:04	term trunk-group 9 cid 0x15f		
14:41:04	dial 52020 route:UDP AAR		
14:41:04	route-pattern 111 preference 1 cid 0x15f		
14:41:04	seize trunk-group 9 member 4 cid 0x15f		
14:41:04	Calling Number & Name NO-CPNumber NO-CPName		
14:41:04	Setup digits 52020		
14:41:04	Calling Number & Name 58020 Joe Everyday		
14:41:04	Proceed trunk-group 9 member 4 cid 0x15f		
14:41:04	Alert trunk-group 9 member 4 cid 0x15f		
	/** Call has routed to destination telephone over trunk 9, ringback is heard **/		
14:41:04	G711MU ss:off ps:20 rn:8/1 2.2.185.4:2800 2.2.35.87:2056		
14:41:04	xoip: fax:Relay modem:off tty:US 2.2.35.87:2056 uid:0x50028		
14:41:04	G711MU ss:off ps:20 rn:1/8 2.2.35.203:2178 2.2.185.4:2800		
14:41:04	G711MU ss:off ps:20 rn:8/1 2.2.185.4:2800 2.2.35.203:2178		
14:41:10	active trunk-group 9 member 4 cid 0x15f		
	/** After call answered, "shuffles" to G.722-64K ip-direct 9620-phone-9620-phone **/		
14:41:10	G72264K ss:off ps:20 rn:8/1 2.2.1.103:2884 2.2.35.203:2178		
14:41:10	G72264K ss:off ps:20 rn:1/8 2.2.35.203:2178 2.2.1.103:2884		

The following screen shows a subset of the information available from the “status station” command for the originating station. The telephone (2.2.35.203) is registered to the S8300 processor Ethernet (2.2.35.88) and is in region 1.

```

status station 58020                                     Page 3 of 7
                                CALL CONTROL SIGNALING
Port: S00007          Switch-End IP Signaling Loc: PROCR   H.245 Port:
      IP Address                                     Port  Node Name      Rgn
Switch-End:  2.  2. 35. 88                           1720
Set End:     2.  2. 35.203                             3177
  
```

The codec used is G.722-64K and the connection is “ip-direct” between the two 9620 IP Telephones. Note that the region of the “Other-End” is region 8, the far-end network region configured for the H.323 signaling group used for the call.

```

status station 58020                                     Page 4 of 7
                                AUDIO CHANNEL Port: S00007
G.722-64K          Switch-End Audio Location:
      IP Address                                     Port  Node Name      Rgn
Other-End:  2.  2.  1.103                             2884
Set-End:    2.  2. 35.203                             2178
Audio Connection Type: ip-direct
  
```

The telephone’s “Authentication Type” is Annex H (as a result of the IP-Network-Region configuration for the network region of this 9600-Series IP Telephone).

```

status station 58020                                     Page 5 of 7
                                IP ENDPOINT DATA
      Port: S00007
Product ID-Release: IP_Phone  1.200          H.245 Tunneled? does not apply
Registration Status: registered-authenticated  MAC Address: 00:04:0d:eb:cb:45
Authentication Type: Avaya AnxH              Dependency Mode: main
Native NAT Address: not applicable
ALG NAT WAN Address: not applicable
  
```

The media path details in the following screen show the end-to-end use of G.722-64K and the “1-srtp-aescm128-hmac80” codec profile by the two Avaya 9620 IP Telephones, one local, and the other at the far-end of the H.323 Trunk.

```

status station 58020                                     Page 6 of 7
                                SRC PORT TO DEST PORT TALKPATH
src port: S00007
S00007:TX:2.2.35.203:2178/g722-64/20ms/1-srtp-aescm128-hmac80
T00040:RX:2.2.1.103:2884/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: T00040
  
```

The following shows similar information for this same call obtained using the “status trunk” command for the trunk member handling the call.

```

status trunk 9/4                                     Page 1 of 2
                                     TRUNK STATUS
Trunk Group/Member: 0009/004                Service State: in-service/active
      Port: T00040                          Maintenance Busy? no
Signaling Group ID:                        CA-TSC state: none
  IGAR Connection? no
    Connected Ports: S00007

      Port      Near-end IP Addr : Port      Far-end IP Addr : Port
Signaling: 01A0017  2. 2. 35. 88 : 10740    2. 2.185. 2 : 1720
  H.245: 01A0017  2. 2. 35. 88 : 10740    2. 2.185. 2 : 1720
G.722-64K Audio:      2. 2. 35.203 : 2178      2. 2. 1.103 : 2884
  Video:
Video Codec:
H.245 Tunneled in Q.931? yes                Authentication Type: Avaya AnxH
Audio Connection Type: ip-direct

```

The second page of “status trunk” confirms the media path, codec, and SRTP profile.

```

status trunk 9/4                                     Page 2 of 2
                                     SRC PORT TO DEST PORT TALKPATH
src port: T00040
T00040:TX:2.2.1.103:2884/g722-64/20ms/1-srtp-aescm128-hmac80
S00007:RX:2.2.35.203:2178/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: S00007

```

5.2. Extension 58020 (9620) Calls Extension 51010 (4610)

This section illustrates the behavior when an Avaya 9600-Series IP Telephone (x58020) calls an Avaya 4600-Series IP Telephone (x51010), and the call is routed over the H.323 IP Trunk. Since the Avaya 4600-Series IP Telephones used in these Application Notes do not support G.722-64K, the G.711MU codec is negotiated. Also, since the remote Avaya 4600-Series IP Telephones does not support SRTP, but the configured inter-system codec set has been configured to require SRTP, a media processor is utilized at the far-end to ensure SRTP is used over the IP Trunk. The far-end system will use AES media encryption between the far-end media processor and the 4600-Series IP Telephone. No media processor is required from the G350 Media Gateway, since the originating 9620 IP Telephone can use SRTP to communicate with the far-end media processor.

The following screen shows the annotated trace output for such a call, taken from the Avaya S8300 Media Server. From this trace, as previously observed, the call is routed using UDP and AAR to route pattern 111 and trunk group 9, the H.323 trunk group. However, in this case, the final bolded rows show the use of G.711MU

```
list trace station 58020                                     Page 1
LIST TRACE
time      data
15:12:59  active station 58020 cid 0x168
15:12:59  G711MU ss:off ps:20 rn:1/1 2.2.35.203:2178 2.2.35.87:2054
15:12:59  xoip: fax:Relay modem:off tty:US 2.2.35.87:2054 uid:0x954
15:13:01  dial 51010 route:UDP|AAR
15:13:01  term trunk-group 9 cid 0x168
15:13:01  dial 51010 route:UDP|AAR
15:13:01  route-pattern 111 preference 1 cid 0x168
15:13:01  seize trunk-group 9 member 4 cid 0x168
15:13:01  Calling Number & Name NO-CPNumber NO-CPName
15:13:01  Setup digits 51010
15:13:01  Calling Number & Name 58020 Joe Everyday
15:13:01  Proceed trunk-group 9 member 4 cid 0x168
15:13:01  Alert trunk-group 9 member 4 cid 0x168
15:13:01  G711MU ss:off ps:20 rn:8/1 2.2.26.4:31792 2.2.35.87:2052
15:13:01  xoip: fax:Relay modem:off tty:US 2.2.35.87:2052 uid:0x50028
15:13:01  G711MU ss:off ps:20 rn:1/8 2.2.35.203:2178 2.2.26.4:31792
15:13:01  G711MU ss:off ps:20 rn:8/1 2.2.26.4:31792 2.2.35.203:2178
15:13:07  active trunk-group 9 member 4 cid 0x168
/** G.711 media path from local 9620 to remote TN2602 media processor at 2.2.185.4 **/
15:13:07  G711MU ss:off ps:20 rn:8/1 2.2.185.4:2960 2.2.35.203:2178
15:13:07  G711MU ss:off ps:20 rn:1/8 2.2.35.203:2178 2.2.185.4:2960
```

The following screen shows a subset of the information available from the “status station” command for the originating station. The greater details presented in previous sections are omitted to focus on the relevant differences. The codec used is G.711MU, and from the point of view of the S8300 Media Server, the connection is “ip-direct” from the local 9620 IP Telephone at 2.2.35.203 to the remote TN2602 at 2.2.185.4. The “1-srtp-aescm128-hmac80” profile is in evidence.

```
status station 58020                                     Page 6 of 7
SRC PORT TO DEST PORT TALKPATH
src port: S00007
S00007:TX:2.2.35.203:2178/g711u/20ms/1-srtp-aescm128-hmac80
T00040:RX:2.2.185.4:2960/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00040
```

From the S8720 Media Server, the status of station 51010 further illustrates the connectivity. The Avaya 4610 IP Telephone (IP Address 2.2.1.110) uses G.711MU and AES media encryption to connect to the TN2602 Media Processor (slot 1A11, IP Address 2.2.185.4), which in turn uses G.711MU and SRTP to communicate across the H.323 Trunk.

```

status station 51010                                     Page 7 of 8
                SRC PORT TO DEST PORT TALKPATH
src port: S00011
S00011:TX:2.2.1.110:2848/g711u/20ms/aes
01A1104:RX:2.2.185.4:2956/g711u/20ms/aes:TX:tdm:a249
01A1103:RX:tdm:a249:TX:2.2.185.4:2960/g711u/20ms/srtp-aescm128-hmac80
T00013:RX:2.2.35.203:2178/g711u/20ms/srtp-aescm128-hmac80
dst port: T00013

```

5.3. Extension 58002 (4610) Calls Extension 52020 (9620)

This section illustrates the behavior when an Avaya 4600-Series IP Telephone (x58002) calls an Avaya 9600-Series IP Telephone (x52020), and the call is routed over the H.323 IP Trunk. Since the local Avaya 4600-Series IP Telephones does not support SRTP, but the configured inter-system codec set has been configured to require SRTP, a media processor from the G350 Media Gateway is utilized at the near-end to ensure SRTP is used over the IP Trunk. Since the G350 Media Gateway media processor does not support G.722-64K, the G.711MU codec is used. No media processor is required from the far-end in the final connection, since the answering 9620 IP Telephone can use SRTP to communicate directly with the G350 Media Gateway media processor.

For the sake of diversity of presentation, the following trace is taken from the S8720 Media Server, illustrating the incoming call on the H.323 Trunk group. The bold rows show the final media path from the Avaya 9620 IP Telephone at 2.2.1.103 to the Avaya G350 Media Gateway at 2.2.35.87. For ease of understanding, the far-end of the H.323 signaling group on the Avaya S8720 Media Server was also configured to be network region 8.

```

list trace tac 109                                     Page 1
                LIST TRACE
time          data
14:43:52      Calling party trunk-group 9 member 1 cid 0x621
14:43:52      Calling Number & Name 58002 Jane Ten
14:43:52      active trunk-group 9 member 1 cid 0x621
14:43:52      dial 52020
14:43:52      ring station 52020 cid 0x621
14:43:52      G711MU ss:off ps:20 rn:3/1 2.2.1.103:2884 2.2.185.4:2860
14:43:52      xoip: fax:Relay modem:off tty:US 2.2.185.4:2860 uid:0x8c51
14:43:52      G711MU ss:off ps:20 rn:8/1 2.2.35.87:2054 2.2.185.4:2856
14:43:52      xoip: fax:Relay modem:off tty:US 2.2.185.4:2856 uid:0x5000d
14:43:58      active station 52020 cid 0x621
14:43:58      G711MU ss:off ps:20 rn:8/3 2.2.35.87:2054 2.2.1.103:2884
14:43:58      G711MU ss:off ps:20 rn:3/8 2.2.1.103:2884 2.2.35.87:2054

```

The following screen shows a subset of the information available from the “status station” command for the originating station. The greater details presented in previous sections are omitted to focus on the relevant differences.

The intra-region codec used is G.711MU, and the connection is “ip-tdm” from the local 4610 IP Telephone at 2.2.35.201 to the Avaya G350 Media Gateway at 2.2.35.87.

```

status station 58002                                     Page 4 of 8
                                AUDIO CHANNEL Port: S00011
G.711MU          Switch-End Audio Location: 1
                IP Address                               Port  Node Name      Rgn
Other-End:      2. 2. 35. 87                            2056
Set-End:        2. 2. 35.201                            2212
Audio Connection Type: ip-tdm

```

Like the Avaya 9620 IP Telephone using extension 58020 illustrated previously, the Avaya 4610 IP Telephone using extension 58002 is registered to the S8300 Media Server and uses Annex H.

```

status station 58002                                     Page 5 of 8
                                IP ENDPOINT DATA
                                Port: S00011
Product ID-Release: IP_Phone 2.700          H.245 Tunneler? does not apply
Registration Status: registered-authenticated  MAC Address: 00:09:6e:01:bf:4d
Authentication Type: Avaya AnxH             Dependency Mode: main
Native NAT Address: not applicable
ALG NAT WAN Address: not applicable

```

The following screen shows the end-to-end media path. The Avaya 4610 IP Telephone uses G.711MU and AES to communicate with the local G350 Media Gateway, which in turn uses G.711MU and the “1-srtp-aescm128-hmac80” profile to communicate with the remote 9620 IP Telephone across the H.323 trunk.

```

status station 58002                                     Page 7 of 8
                                SRC PORT TO DEST PORT TALKPATH
src port: S00011
S00011:TX:2.2.35.201:2212/g711u/20ms/aes
001V035:RX:2.2.35.87:2056/g711u/20ms/aes:TX:ctxID:166
001V036:RX:ctxID:166:TX:2.2.35.87:2054/g711u/20ms/1-srtp-aescm128-hmac80
T00038:RX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00038

```

5.4. Extension 58022 (IP Softphone) Calls Extension 52020 (9620)

This section illustrates the behavior when an Avaya IP Softphone (x58022) calls an Avaya 9600-Series IP Telephone (x52020), and the call is routed over the H.323 IP Trunk. This section is included for completeness, but no new concept is illustrated. With respect to media encryption, the IP Softphone and Avaya 4600-Series IP Telephones will result in the same connection topology. Since the local Avaya IP Softphone does not support SRTP, but the configured inter-system codec set has been configured to require SRTP, a media processor from the G350 Media Gateway is utilized at the near-end to ensure SRTP is used over the IP Trunk. Since the G350 Media Gateway media processor does not support G.722-64K, the G.711MU codec is used. No media processor is required from the far-end in the final connection, since the answering 9620 IP Telephone can use SRTP to communicate directly with the G350 Media Gateway media processor.

The following screen shows a subset of the information available from the “status station” command for the originating IP Softphone. The intra-region codec used is G.711MU, and the connection is “ip-tdm” from the local IP Softphone at 2.2.35.222 to the Avaya G350 Media Gateway at 2.2.35.87, using G.711MU and AES. The connection across the trunk from the Avaya G350 Media Gateway to the far-end Avaya 9620 IP Telephone uses G.711MU and SRTP. This connection topology is the same as when the call was originated from an Avaya 4600-Series IP Telephone rather than the Avaya IP Softphone.

```

status station 58022                                     Page 7 of 8
                SRC PORT TO DEST PORT TALKPATH
src port: S00043
S00043:TX:2.2.35.222:2048/g711u/20ms/aes
001V035:RX:2.2.35.87:2056/g711u/20ms/aes:TX:ctxID:220
001V038:RX:ctxID:220:TX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80
T00040:RX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00040

```

5.5. Extension 58003 (2420) Calls Extension 52020 (9620)

This section illustrates the behavior when a TDM device, such as an Avaya 2400-Series Digital Telephone (x58003) calls an Avaya 9600-Series IP Telephone (x52020), and the call is routed over the H.323 IP Trunk. A TDM device such as the Avaya 2420 will require the services of a media processor. Since the Avaya G350 Media Processor does not support G.722-64K but does support SRTP, the G.711MU codec is negotiated, and SRTP is used across the H.323 trunk. No media processor is required from the far-end in the final connection, since the answering 9620 IP Telephone can use SRTP to communicate with the G350 media processor.

Since this case is not substantially different than those previously presented, only a “status station” screen showing the end-end media path is included. The Avaya G350 Media Gateway serving the Avaya 2420 Digital Telephone connected to port 1V401 uses G.711MU and SRTP to communicate with the remote Avaya 9620 IP Telephone.

```

status station 58003                                     Page 6 of 7
                SRC PORT TO DEST PORT TALKPATH
src port: 001V401
001V401:TX:ctxID:177
001V038:RX:ctxID:177:TX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80
T00037:RX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00037

```

5.6. Examples of Local Calls on the S8300 Media Server System

Although these Application Notes focus on the inter-system SRTP connectivity over the H.323 IP Trunk, local intra-system calls are also secured with this configuration. Per the sample objectives outlined in Section 1.1, recall that local calls have been secured with the strongest encryption natively supported by the devices, to minimize the use of media gateway VoIP resources. The following sub-sections illustrate the connection details for a representative sample of local calls.

5.6.1. Extension 58020 (9620) Calls Extension 58030 (9630)

For a local call between two Avaya 9600 Series IP Telephones, the final media connection will be “ip-direct” between the two IP Telephones using G.722 and SRTP. The following screen illustrates an example call trace.

```
list trace station 58020                                     Page 1
LIST TRACE
time      data
14:42:52  active station      58020 cid 0x97
14:42:52  G711MU ss:off ps:20 rn:1/1 2.2.35.203:2178 2.2.35.87:2052
14:42:52  xoip: fax:Relay modem:off tty:US 2.2.35.87:2052 uid:0x954
14:42:55  dial 58030
14:42:55  ring station      58030 cid 0x97
14:42:55  G711MU ss:off ps:20 rn:1/1 2.2.35.209:2950 2.2.35.87:2050
14:42:55  xoip: fax:Relay modem:off tty:US 2.2.35.87:2050 uid:0x952
14:42:56  active station      58030 cid 0x97
14:42:56  G72264K ss:off ps:20 rn:1/1 2.2.35.203:2178 2.2.35.209:2950
14:42:56  G72264K ss:off ps:20 rn:1/1 2.2.35.209:2950 2.2.35.203:2178
```

The following screen shows the connection details using the “status station” command.

```
status station 58020                                       Page 6 of 7
SRC PORT TO DEST PORT TALKPATH
src port: S00007
S00007:TX:2.2.35.203:2178/g722-64/20ms/1-srtp-aescm128-hmac80
S00009:RX:2.2.35.209:2950/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: S00009
```

5.6.2. Extension 58002 (4610) Calls Extension 58020 (9620)

For a local call between an Avaya 4600-Series IP Telephone and Avaya 9600-Series IP Telephone, the final media connection will be “ip-direct” between the two IP Telephones using G.711MU and AES media encryption. The following screen illustrates an example call trace.

```
list trace station 58002                                     Page 1
LIST TRACE
time      data
14:45:48  active station      58002 cid 0x9a
14:45:48  G711MU ss:off ps:20 rn:1/1 2.2.35.201:2212 2.2.35.87:2056
14:45:48  xoip: fax:Relay modem:off tty:US 2.2.35.87:2056 uid:0x958
          VOIP data from: 2.2.35.87:2056
14:45:50  dial 58020
14:45:50  ring station      58020 cid 0x9a
14:45:50  G711MU ss:off ps:20 rn:1/1 2.2.35.203:2178 2.2.35.87:2058
14:45:50  xoip: fax:Relay modem:off tty:US 2.2.35.87:2058 uid:0x954
14:45:51  active station      58020 cid 0x9a
14:45:51  G711MU ss:off ps:20 rn:1/1 2.2.35.201:2212 2.2.35.203:2178
14:45:51  G711MU ss:off ps:20 rn:1/1 2.2.35.203:2178 2.2.35.201:2212
```

The following screen shows the connection details using the “status station” command.

```
status station 58002                               Page 6 of 7
SRC PORT TO DEST PORT TALKPATH
src port: S00011
S00011:TX:2.2.35.201:2212/g711u/20ms/aes
S00007:RX:2.2.35.203:2178/g711u/20ms/aes
dst port: S00007
```

5.6.3. Extension 58030 (9630) Calls Extension 58003 (2420)

For a local call between an Avaya 9600-Series IP Telephone and a non-IP device such as an Avaya 2400-Series Digital Telephone, the IP leg of the connection will be between the 9600-Series IP Telephone and the G350 Media Gateway VoIP using G.711MU and SRTP. The following screen shows the connection details for such a call using the “status station” command.

```
status station 58030                               Page 7 of 8
SRC PORT TO DEST PORT TALKPATH
src port: S00009
S00009:TX:2.2.35.209:2950/g711u/20ms/1-srtp-aescm128-hmac80
001V038:RX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:132
001V401:RX:ctxID:132
dst port: 001V401
```

5.6.4. Extension 58002 (4610) Calls Extension 58003 (2420)

For a local call between an Avaya 4600-Series IP Telephone and a non-IP device such as an Avaya 2400-Series Digital Telephone, the IP leg of the connection will be between the 4600-Series IP Telephone and the G350 Media Gateway VoIP using G.711MU and AES media encryption. The following screen shows the connection details for such a call using the “status station” command.

```
status station 58002                               Page 6 of 7
SRC PORT TO DEST PORT TALKPATH
src port: S00011
S00011:TX:2.2.35.201:2212/g711u/20ms/aes
S00007:RX:2.2.35.203:2178/g711u/20ms/aes
dst port: S00007
```

5.7. Conference Call Example

The preceding sections illustrated various point-to-point calls between two end user telephones. For conference calls, Avaya Communication Manager can apply the configured encryption to each IP leg of the overall conference topology. In the example presented in this section, an Avaya Communication Manager meet-me conference on the Avaya S8300 Media Server was used to enable six telephones to dial-in to join a conference. Four of the telephones (x58002, x58003, x58020, x58030) were local to the S8300 Media Server. The other two telephones (x51010, x52020) dialed in over the H.323 Trunk Group to the conference.

The following screen illustrates the status of the meet-me conference vector directory number 58081 (configuration not shown), showing the four local telephones and two trunks representing the two remote telephones.

```

status meet-me-vidn 58081
                                GENERAL STATUS
                                Service State: active
Extension: 58081
Conferee Ports
001V401      S00007      S00011      S00009      T00037      T00038

```

The following screens are included with little elaboration to illustrate that the various G.711MU IP legs in the final conference topology are secured in the same fashion as illustrated previously for point-to-point calls. Since these Application Notes focus on the trunk connectivity, the “status trunk” command is used. In this case, trunk member 1 is associated with the call from remote station 52020 (9620, IP 2.2.1.103). Similar information would be available using the “status station” command for a conference participant on the Avaya S8300 Media Server.

```

status trunk 9/1                                     Page 1 of 7
                                TRUNK STATUS
Trunk Group/Member: 0009/001      Service State: in-service/active
Port: T00037                      Maintenance Busy? no
Signaling Group ID:              CA-TSC state: none
IGAR Connection? no
  Connected Ports: 001V401      S00007
                   S00011      S00009
                   T00038

Port      Near-end IP Addr : Port      Far-end IP Addr : Port
Signaling: 01A0017      2. 2. 35. 88 : 1720      2. 2.185. 2 : 19940
H.245: 01A0017      2. 2. 35. 88 : 1720      2. 2.185. 2 : 19940
G.711MU   Audio: 1      2. 2. 35. 87 : 2050      2. 2. 1.103 : 2884
Video:
Video Codec:
H.245 Tunneler in Q.931? yes      Authentication Type: Avaya AnxH
Audio Connection Type: ip-tdm

```

Paging forward, the following screen shows a leg of the call involving digital station 58003 on port 1V401. Observe the use of G.711MU and SRTP across the trunk.

```

status trunk 9/1                                     Page 3 of 7
                                SRC PORT TO DEST PORT TALKPATH
src port: T00037
T00037:TX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
001V038:RX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:163
001V401:RX:ctxID:163
dst port: 001V401

```

Paging forward, the following screen shows a leg of the call involving extension 58020. Observe the use of G.711MU and SRTP across the trunk and also locally to the Avaya 9620 IP Telephone (2.2.35.203)

```
status trunk 9/1 Page 4 of 7
                SRC PORT TO DEST PORT TALKPATH
src port: T00037
T00037:TX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
001V038:RX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:163
001V035:RX:ctxID:163:TX:2.2.35.87:2056/g711u/20ms/1-srtp-aescm128-hmac80
S00007:RX:2.2.35.203:2178/g711u/20ms/1-srtp-aescm128-hmac80
dst port: S00007
```

Paging forward, the following screen shows a leg of the call involving extension 58002. Observe the use of G.711MU and SRTP across the trunk, and the use of AES locally to the Avaya 4610 IP Telephone (2.2.35.201)

```
status trunk 9/1 Page 5 of 7
                SRC PORT TO DEST PORT TALKPATH
src port: T00037
T00037:TX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
001V038:RX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:163
001V034:RX:ctxID:163:TX:2.2.35.87:2058/g711u/20ms/aes
S00011:RX:2.2.35.201:2212/g711u/20ms/aes
dst port: S00011
```

Paging forward, the following screen shows a leg of the call involving extension 58030. Observe the use of G.711MU and SRTP across the trunk as well as locally to the Avaya 9630 IP Telephone (2.2.35.209)

```
status trunk 9/1 Page 6 of 7
                SRC PORT TO DEST PORT TALKPATH
src port: T00037
T00037:TX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
001V038:RX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:163
001V037:RX:ctxID:163:TX:2.2.35.87:2052/g711u/20ms/1-srtp-aescm128-hmac80
S00009:RX:2.2.35.209:2950/g711u/20ms/1-srtp-aescm128-hmac80
dst port: S00009
```

Paging forward, the following screen shows a leg of the call involving the second trunk member in the conference. Observe the use of G.711MU and SRTP across both trunks. Trunk port T00038 connects to the remote TN2602 Media Processor (2.2.185.4), which had been inserted by the S8720 Media Server when the Avaya 4610 IP Telephone (extension 51010) called into the conference, to satisfy the configured requirement to use SRTP across the trunk.

```
status trunk 9/1 Page 7 of 7
                SRC PORT TO DEST PORT TALKPATH
src port: T00037
T00037:TX:2.2.1.103:2884/g711u/20ms/1-srtp-aescm128-hmac80
001V038:RX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:163
001V033:RX:ctxID:163:TX:2.2.35.87:2060/g711u/20ms/1-srtp-aescm128-hmac80
T00038:RX:2.2.185.4:2600/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00038
```

6. Conclusion

As illustrated in these Application Notes, the Secure Real-Time Transport Protocol (SRTP) introduced in Avaya Communication Manager Release 4.0 can be used to secure local connections as well as connections over H.323 IP Trunks. Customers can satisfy varying security objectives for different parts of a network using a flexible configuration approach. A network can logically be partitioned into different “network regions”, and codec and security decisions can be separately controlled for connections within a given region, and for connections from a given region to any other region. As illustrated in these Application Notes, Avaya Communication Manager evaluates the capabilities of the devices in a connection and makes intelligent choices, inserting media gateway resources if required, to honor the configured security requirements.

7. References

References [1] and [2] are among the Application Notes available at <http://www.avaya.com>.

The first reference uses the same Avaya S8720 Media Server configuration used in these Application Notes.

[1] “Configuring Avaya Communication Manager Release 4.0 for Dial Plan Transparency and Inter-Gateway Alternate Routing”, Issue 1.0.

The following Application Notes were written coincident with Avaya Communication Manager Release 2.0 (prior to the support of SRTP) and illustrate the use of both AES and AEA media encryption:

[2] “Configuring Avaya Communication Manager for Media Encryption”, Issue 1.0.

<http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/media-encrypt.pdf>

Reference [3] below includes information related to H.235.5 (Annex H).

[3] “What’s New in Avaya Communication Manager 3.1.X, Document Number 03-300682, Issue 1.2, July 2006”

http://support.avaya.com/elmodocs2/comm_mgr/r3_1/pdfs/03_300682_1_2.pdf

Reference [4] is a link to The Secure Real-Time Transport Protocol, RFC 3711.

[4] <http://www.ietf.org/rfc/rfc3711.txt>

Reference [5] is an example of relevant Avaya Communication Manager Release 4.0 product documentation available at <http://support.avaya.com>

[5] Administration for Network Connectivity for Avaya Communication Manager, Document Number 555-233-504, Issue 12, February 2007

Reference [6] is a tutorial on Avaya Communication Manager network regions.

[6] Avaya Communication Manager Network Region Configuration Guide, October 2005

http://support.avaya.com/elmodocs2/comm_mgr/r3/netw-region-tutorial-cm30-1005.pdf

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