Avaya Solution & Interoperability Test Lab

Configuring Secure Real-Time Transport Protocol (SRTP) and G.722 Audio using Avaya 9600-Series IP Telephones running SIP and H.323 Firmware – Issue 1.0

Abstract

These Application Notes illustrate the configuration and operation of the Secure Real-Time Transport Protocol (SRTP) and the G.722 wideband audio codec, in conjunction with Avaya 9600-Series IP Telephones running SIP and H.323 firmware. Avaya 9600-Series IP Telephones running SIP firmware can obtain SRTP and G.722 parameters from a settings file. These Application Notes supplement previously published Application Notes that illustrated SRTP and G.722 using Avaya 9600-Series IP Telephones running H.323 firmware. A sample Avaya Communication Manager configuration for secure connectivity to Avaya 9600-Series IP Telephones using SRTP and G.722 Audio is presented.

1. Introduction and Scope

These Application Notes supplement previously published Application Notes that illustrated SRTP and G.722 using Avaya 9600-Series IP Telephones running H.323 firmware. **Figure 1** provides an overview of the network used to verify the previously published Application Notes. The **Figure 1** configuration is described in reference [1].

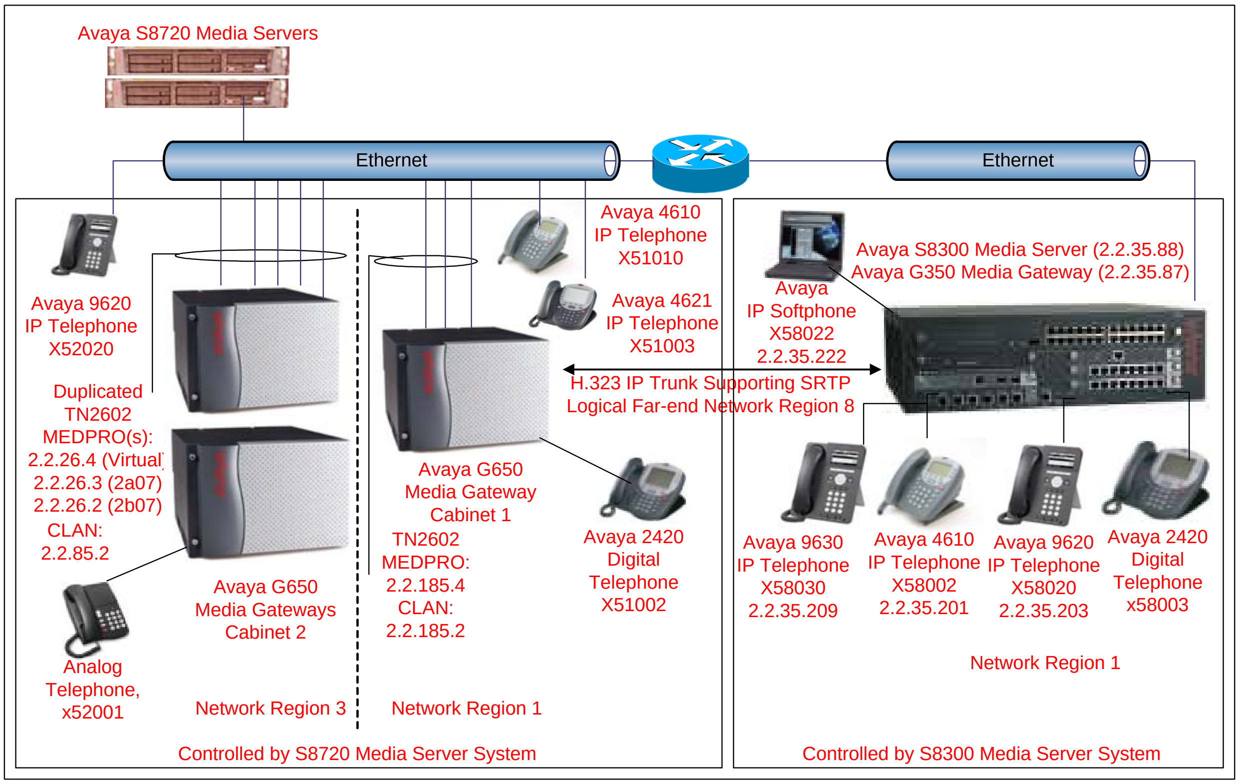


Figure 1: Network Overview Illustrated in Reference [1]

In these Application Notes, the same Avaya S8300 Server shown in **Figure 1** is used as the feature server for Avaya 9600-Series IP Telephones running SIP firmware that support SRTP and G.722, as shown in **Figure 2**. These Avaya one-X™ Deskphone SIP telephones are registered to an Avaya SIP Enablement Services Server (SES) redundant configuration. The SES duplication is not required for SRTP or G.722, and the configuration of the SES is outside the scope of these Application Notes. Although not shown in **Figure 2**, the sample configuration was also verified with devices that do not support SRTP or G.722. For information on Avaya 1600-Series IP Telephones, see Section 5.8. For information on Avaya IP Softphone, see Section 5.9. For information on Avaya 4600-Series IP Telephones running SIP firmware, see Section 5.10.

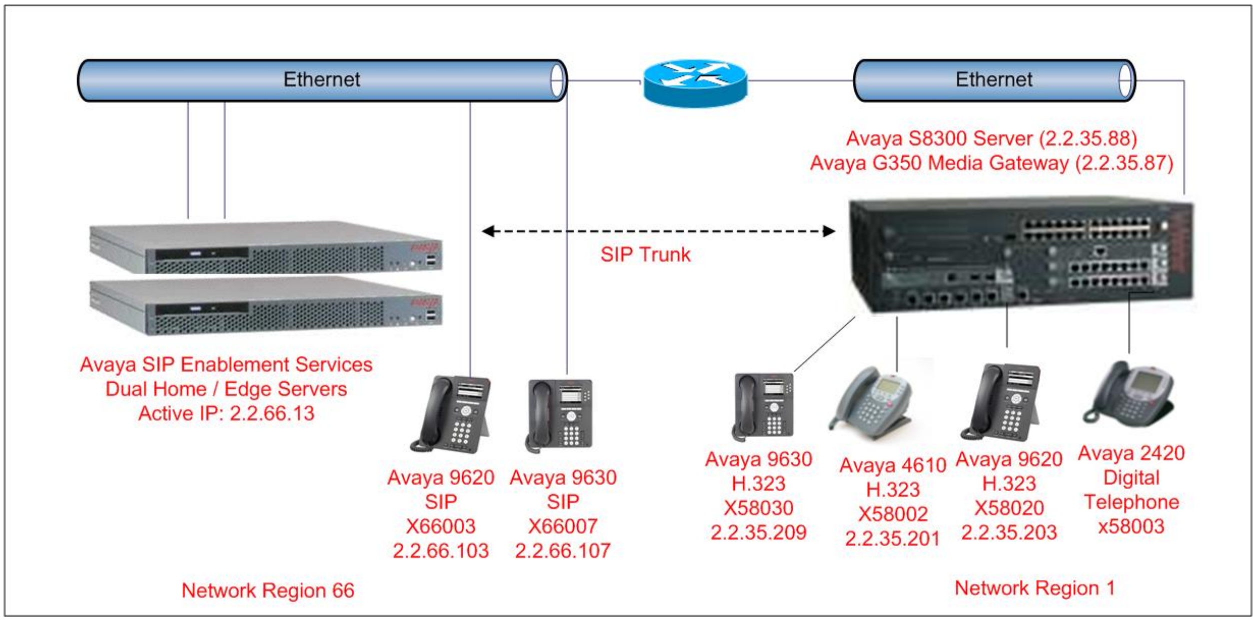


Figure 2: Network Configuration for SRTP using Avaya one-X™ Deskphone SIP

1.1. Example Codec / Security Objectives for the Figure 2 Network

These Application Notes assume the following high-level objectives for the **Figure 2** network. Each customer's objectives will vary. Section 5 shows a representative sample of calls fulfilling these objectives:

- Media connections within a network region 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-region connections will utilize the fewest resources from media gateway VoIP processors. These intra-region 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 regions will again be optimized for the highest quality audio codec, but SRTP will be required for inter-region connections, even if satisfying the SRTP requirement means that media gateway resources must be utilized. This objective assumes that inter-region calls require the strongest security procedures, independent of the native capabilities of the end devices. These inter-region objectives will be satisfied by configuring a codec set with the same audio codec preferences used for intra-region calls, but with a requirement to use SRTP for the inter-region media connection. End-user devices incapable of native SRTP will have gateway media processing resources assigned automatically to honor the requirement that SRTP be used for the inter-region connection.

These objectives mirror the objectives presented in reference [1] for the **Figure 1** network.

A five digit dial plan is used so that any user may dial a five-digit number to reach any other user. Extensions in the range 58XXX are used for all devices controlled by the Avaya S8300 Server that are not SIP (i.e., the H.323 and digital stations). Extensions in the range 660XX are used for the SIP telephones registered with the Avaya SES. The Avaya S8300 Server and Avaya SES are connected via SIP. A procedural example of configuring SIP connectivity between Avaya Communication Manager and the Avaya SES can be found in reference [3]. In these Application Notes, sample screens capture the relevant configuration, but detailed procedural steps are omitted.

Section 5 illustrates detailed status and call trace verifications to reinforce the behavior achieved using the configuration presented in Section 4. Although the focus of these Application Notes is the connectivity for calls using SIP, note that the procedures also result in encrypted connections among the H.323 IP Telephones and digital telephones controlled by Avaya Communication Manager, as detailed in reference [1].

1.2. Avaya Communication Manager SRTP Summary

SRTP is a media encryption standard, defined in RFC 3711 (Reference [5]) as a profile of RTP. The Avaya Communication Manager 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. With HMAC-SHA1-80, an authentication value included in the packet results in 10 additional bytes of overhead per RTP 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 (e.g., direct SRTP media between two Avaya 9600-Series IP Telephones, whether H.323 or SIP).
- Capability harmonization among parties in a call, and insertion of gateway media processing resources where required.

In addition to the Avaya G350 Media Gateway, TN2602 Media Processor, and Avaya 9600-Series IP Telephones illustrated in these Application Notes and reference [1], the Avaya G250 Media Gateway, Avaya G700 Media Gateway, and Avaya IG550 Integrated Gateway also support SRTP. At time of writing, the TN2302 Media Processor, and the Avaya 4600-Series IP

Telephones running H.323 firmware support AES media encryption, but not SRTP. The Avaya 4600-Series IP Telephones running SIP firmware support neither SRTP nor AES media encryption. For more information on including Avaya 4600-Series IP Telephones running SIP firmware, see Section 5.10.

2. Equipment and Software Validated

Table 1 shows the equipment and version information used in the sample configuration shown in **Figure 2**. All versions used are generally available.

Network Component	Version Information
Avaya SIP Enablement Services Server (SES)	SES 4.0 (build 33.6)
Avaya S8300 Server	Avaya Communication Manager 4.0.1 Load 731.2
Avaya G350 Media Gateway	26.33.0
Avaya 4600-Series IP Telephones (H.323)	2.8
Avaya 9600-Series IP Telephones (H.323)	1.5
Avaya 9600-Series IP Telephones (SIP)	1.0 (1.0.13.1 (5))

Table 1 – Equipment Version Information

3. Conventions, Assumptions, and Licensing

In these Application Notes, Avaya Communication Manager screen captures 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 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 Off-PBX Telephone capacity is available for the SIP Telephones.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V14
       Location: 1                          RFA System ID (SID): 1
       Platform: 13                         RFA Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 900   91
                                      Maximum Stations: 450   33
                              Maximum XMOBILE Stations: 10    0
                    Maximum Off-PBX Telephones - EC500: 10    2
                    Maximum Off-PBX Telephones -   OPS: 10    8
```

On Page 2, verify sufficient station and trunk capacity are available.

```
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: 40    8
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 1     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 2     0
                   Maximum Video Capable IP Softphones: 2     0
                       Maximum Administered SIP Trunks: 50    10

   Maximum Number of DS1 Boards with Echo Cancellation: 1     0
                             Maximum TN2501 VAL Boards: 0     0
                     Maximum Media Gateway VAL Sources: 1     1
```

The bolded fields shown on Page 4 are features that must be enabled in the license to achieve the configuration presented in reference [1] and 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
```

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

4. Configuration

This section illustrates the configuration of Avaya Communication Manager, Avaya SES, Avaya 9600-Series telephone and 46xxsettings.txt parameters relevant to achieving the objectives in Section 1.1.

4.1. Avaya Communication Manager Configuration

This section illustrates the relevant Avaya Communication Manager configuration for the S8300 Server shown in **Figure 2**. For detailed product documentation on Avaya Communication Manager, consult references [6] and [10].

4.1.1. 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 a SIP signaling group. The node name "procr" refers to the Avaya S8300 Server processor Ethernet. The node name "SES-act" refers to the logical IP Address of the duplicated Avaya SES system, as shown in **Figure 2.**

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SES-act             2.2.66.13
procr               2.2.35.88
```

4.1.2. SIP Signaling Group and Trunk Group

Configure the SIP signaling group using the "add signaling group" command. In this case, signaling group 66 is created. The parameters relevant to these Application Notes are shown in bold below.

The **Group Type** is set to "sip". The **Near-end Node Name** is set to "procr", the S8300 Server processor Ethernet. The **Far-end Node Name** is set to "SES-act", the node name assigned to the logical IP address of the duplicated Avaya SES system. The **Near-end Listen Port** and **Far-end Listen Port** are 5061, consistent with the **Transport Method** of "tls". The **Far-end Network Region** parameter is set to 66, a network region that is different from the near-end devices on the Avaya S8300 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 SIP signaling group and trunk group. The **Far-end Domain** is set to a domain consistent with the overall configuration. In this case, the domain is "demoroom.com". The **Direct IP-IP Audio Connections** field must be left at the default value of "y" to allow direct media paths (e.g., between Avaya 9600-Series IP Telephones) for calls using this signaling group.

```
add signaling-group 66
                                SIGNALING GROUP

 Group Number: 66              Group Type: sip
                        Transport Method: tls

   Near-end Node Name: procr                  Far-end Node Name: SES-act
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 66
       Far-end Domain: demoroom.com
                                            Bypass If IP Threshold Exceeded? n
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
 Enable Layer 3 Test? n
 Session Establishment Timer(min): 120
```

The command "add trunk-group" can be used to add a trunk group to be associated with the SIP signaling group. The use of SRTP does not affect the trunk group configuration. In the screen capture that follows, the **Group Type** is set to "sip", the **Signaling Group** is set to 66, the

Service Type is set to "tie", and an appropriate **TAC** and **Number of Members** have been specified. Each connection to a SIP telephone will occupy one SIP trunk member.

```
add trunk-group 66                                                  Page   1 of  21
                                TRUNK GROUP
Group Number: 66                   Group Type: sip           CDR Reports: y
  Group Name: SIP-SES                     COR: 3        TN: 1        TAC: 166
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                       Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                        Signaling Group: 66
                                                      Number of Members: 10
```

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 T00049 – T00058. These trunk identifiers are shown below, and can also be observed in the verification screens shown in Section 5.

```
display trunk-group 66                                                  Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/10
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  10
       Port             Name
  1: T00049             SIP-SES
  2: T00050             SIP-SES
  3: T00051             SIP-SES
  4: T00052             SIP-SES
  5: T00053             SIP-SES
  6: T00054             SIP-SES
  7: T00055             SIP-SES
  8: T00056             SIP-SES
  9: T00057             SIP-SES
 10: T00058             SIP-SES
```

4.1.3. Association of Endpoints with Network Regions

IP telephones can be assigned a network region based on an IP address mapping. 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 H.323 IP Telephones registered to the Avaya S8300 Server are assigned to network region 1. The Avaya 9600-Series SIP Telephones are mapped to network region 66. (In the configuration presented in these Application Notes, the IP Network Map configuration shown below is not strictly necessary. The far-end of the SIP signaling group has already been set to 66, and the H.323 IP Telephones are mapped to default region 1).

```
change ip-network-map                                                   Page   1 of  32
                         IP ADDRESS MAPPING
                                                                   Emergency
                                            Subnet                 Location
 From IP Address  (To IP Address   or Mask)  Region    VLAN   Extension
 2  .2  .35 .200  2  .2  .35 .209              1        n
 2  .2  .66 .100  2  .2  .66 .110              66       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 the SIP Trunk Group, a media processing resource (e.g., the integrated VoIP processing of the 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 (by default), and that the gateway is configured in network region 1 (also the default).

```
display media-gateway 1
                              MEDIA GATEWAY
         Number: 1                                Registered?  y
           Type: g350                FW Version/HW Vintage: 26 .33 .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
```

4.1.4. Network Region Configuration

Reference [7] is a tutorial on Avaya Communication Manager network regions. The following screens illustrate aspects of the configuration for network region 1. As shown in the previous section, all the H.323 and digital endpoints served by the Avaya S8300 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". The **Authoritative Domain** matches the "demoroom.com" domain also configured for the SIP signaling group. The **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** fields are left at the default "yes" values to allow direct media path connections where appropriate.

```
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                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
```

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 H.323 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 screens illustrate aspects of the configuration for network region 66. From Page 1, observe that the **Codec Set** to be used for connections within region 66 is set to "1". The **Authoritative Domain** matches the "demoroom.com" domain also configured for the SIP signaling group. The **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** fields are left at the default "yes" values to allow direct media path connections where appropriate.

```
change ip-network-region 66                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 66
Location:         Authoritative Domain: demoroom.com
    Name: SIP-Phones
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3029
```

The connectivity between network region 66 (SIP telephones) and network region 1 (H.323 IP Telephones, digital telephones) is specified via the Inter Network Region Connection Management table beginning on Page 3. These connections will use codec set 3 as shown below.

```
change ip-network-region 66                                     Page   3 of  19

                   Inter Network Region Connection Management
 src dst codec direct   WAN-BW-limits   Video                          Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Intervening-regions  CAC IGAR
 66  1    3     y    NoLimit                                                 n
```

The assignment of the codec set 3 interconnectivity between network region 66 and 1 automatically creates a symmetrical configuration for region 1 (to region 66).

4.1.5. 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                   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 SIP signaling group are configured to require SRTP while still using the best quality codec supported by the SRTP-communicating devices. Therefore, codec set 3, used for connections between network region 1 and network region 66, 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. For considerations when using Avaya 4600-series IP Telephones running SIP firmware, see Section 5.10.

```
change ip-codec-set 3                                            Page   1 of   2
                          IP Codec Set
    Codec Set: 3

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

4.1.6. Station for SIP Telephone Extension

The following screen illustrates the configuration for one of the SIP telephone extensions. There is no unique configuration for SRTP or G.722 audio. Similar station configuration is performed for each of the extensions to be associated with the SIP Telephones.

```
display station 66003                                          Page   1 of   5
                                     STATION
Extension: 66003                         Lock Messages? n             BCC: 0
     Type: 9620                          Security Code: 123456         TN: 1
     Port: S00015                      Coverage Path 1:               COR: 1
     Name: Juan Deraround              Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 66003
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
```

The default configuration to allow **Direct IP-IP Audio Connections** is retained.

```
display station 66003                                          Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
            CDR Privacy? n                                Data Restriction? n
   Redirect Notification? y                      Idle Appearance Preference? n
 Per Button Ring Control? n                    Bridged Idle Line Preference? n
   Bridged Call Alerting? n                        Restrict Last Appearance? y
  Active Station Ringing: single
                                                               EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced                  Audible Message Waiting? n
     MWI Served User Type:                       Display Client Redirection? n
              AUDIX Name:                       Select Last Used Appearance? n
                                                  Coverage After Forwarding? S

                                                Direct IP-IP Audio Connections? y
  Emergency Location Ext: 66003            Always Use? n IP Audio Hairpinning? n
```

In the sample configuration, the Avaya Environment parameter for the 9600-Series SIP telephones was set to “yes”. The Avaya Environment enables Avaya SIP telephones to obtain feature button programming automatically, based on the button assignments translated in the Avaya Communication Manager station record. Note that the Avaya Environment is not required for SRTP or G.722.

```
display station 66003                                          Page   4 of   5
                                     STATION
BUTTON ASSIGNMENTS
 1: call-appr                               4: call-appr
 2: call-appr                               5: no-hld-cnf
 3: call-appr                               6: call-pkup
```

4.1.7. Off-PBX Telephone Mapping for SIP Telephones

Although there is nothing unique about the off-pbx telephone mapping due to the use of SRTP, the following screens are included to illustrate the configuration. Station extension 66003 is mapped to SIP trunk group 66. The Avaya SES also uses 66003 for this user. The default configuration set is retained.

```
change off-pbx-telephone station-mapping 66003                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 66003             OPS          -      66003              66         1
```

The following screen shows Page 2 of the off-pbx-telephone station-mapping for extension 66003.

```
change off-pbx-telephone station-mapping 66003                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 66003          3       both           all         both
```

Off-pbx mappings are performed for each SIP telephone. The following screen shows a list command illustrating the off-pbx-telephone station-mapping used in the configuration, including the subset of the telephone extensions shown in **Figure 2**.

```
list off-pbx-telephone station-mapping
                    STATION TO OFF-PBX TELEPHONE MAPPING

Station        Appl   CC  Phone Number     Config Trunk   Mapping      Calls
Extension                                  Set    Select  Mode         Allowed
66000          OPS        66000            1  /   66      both         all
66001          OPS        66001            1  /   66      both         all
66002          OPS        66002            1  /   66      both         all
66003          OPS        66003            1  /   66      both         all
66004          OPS        66004            1  /   66      both         all
66005          OPS        66005            1  /   66      both         all
66006          OPS        66006            1  /   66      both         all
66007          OPS        66007            1  /   66      both         all
```

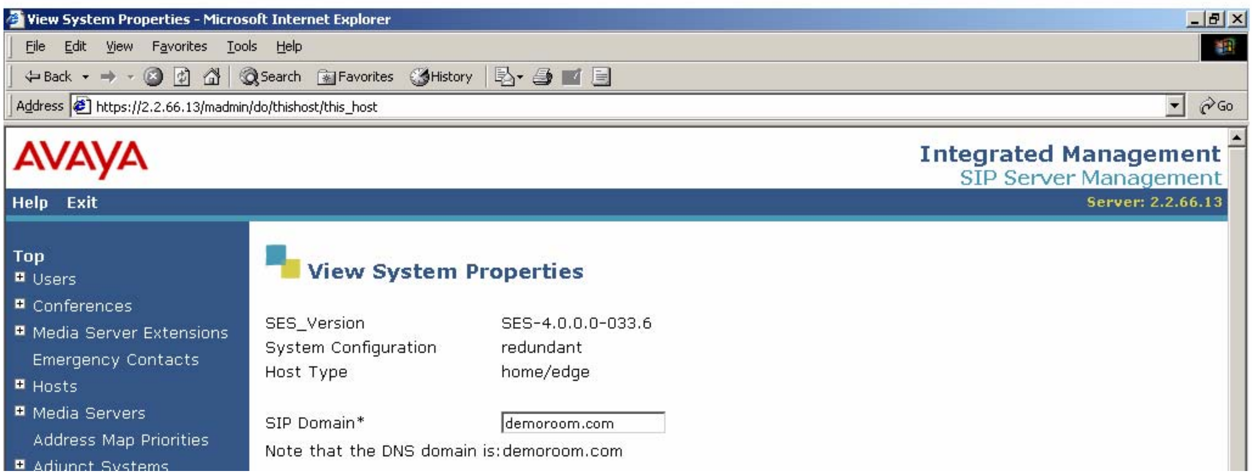
4.1.8. Saving Configuration Changes

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

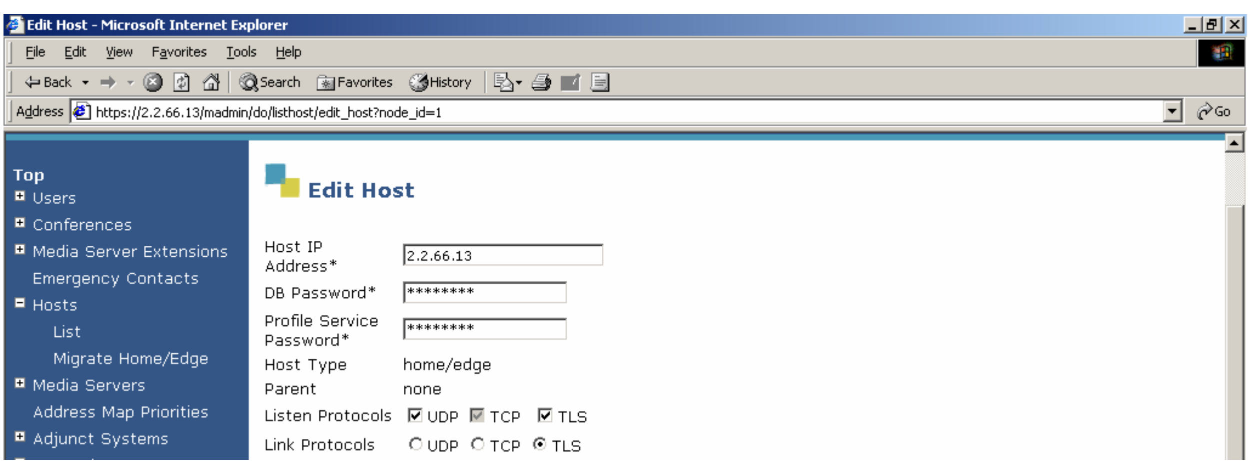
4.2. Avaya SES Configuration

Although there is nothing unique about the Avaya SES configuration due to the use of SRTP or G.722, the following screens illustrate aspects of the SES configuration used in these Application Notes. Consult reference [3] and reference [11] for additional information. All screen captures in this section can be obtained from the Avaya SES Administrative Web interface.

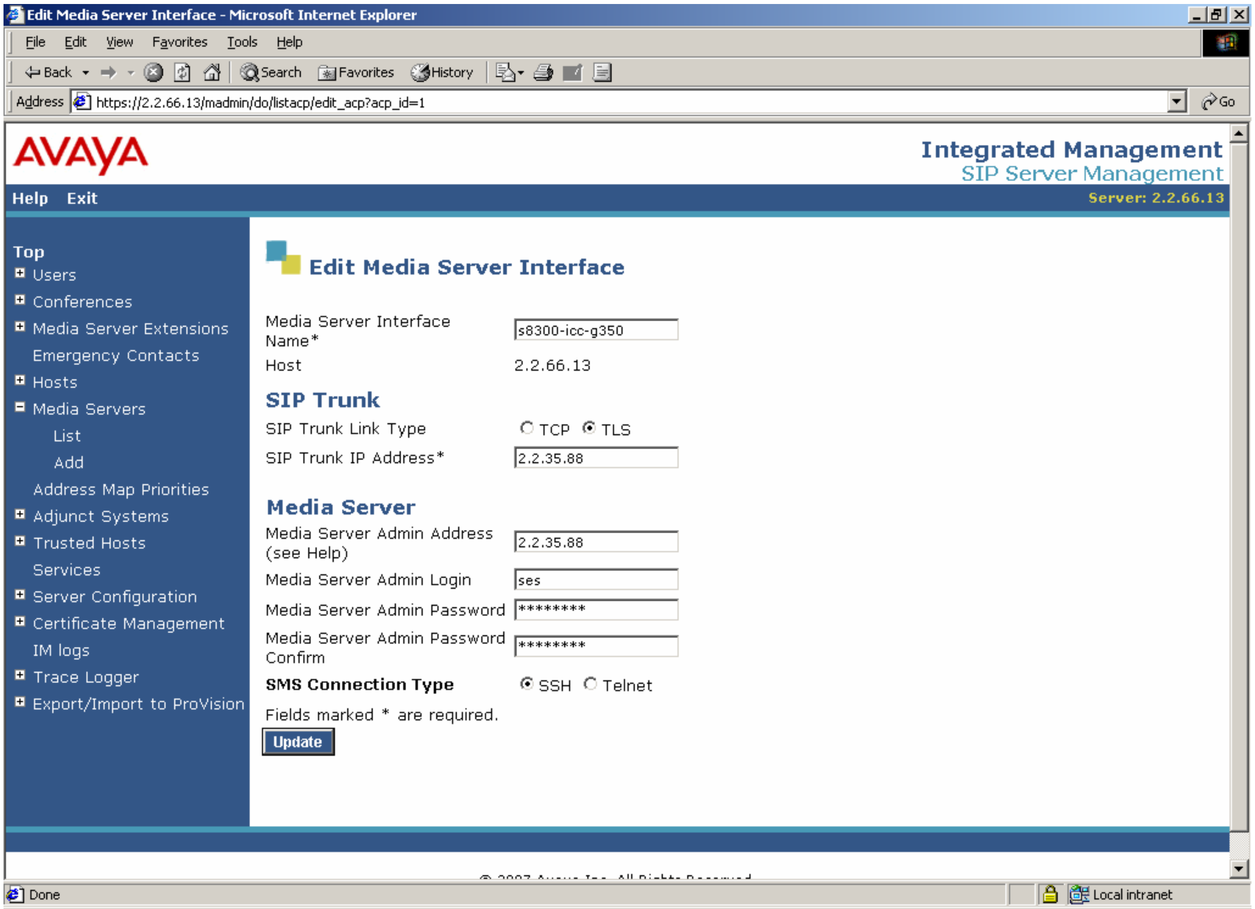
The following screen illustrates a portion of the System Properties, accessible via **Server Configuration → System Properties**. The SIP Domain "demoroom.com" and SES Version shown in **Table 1** can be observed.



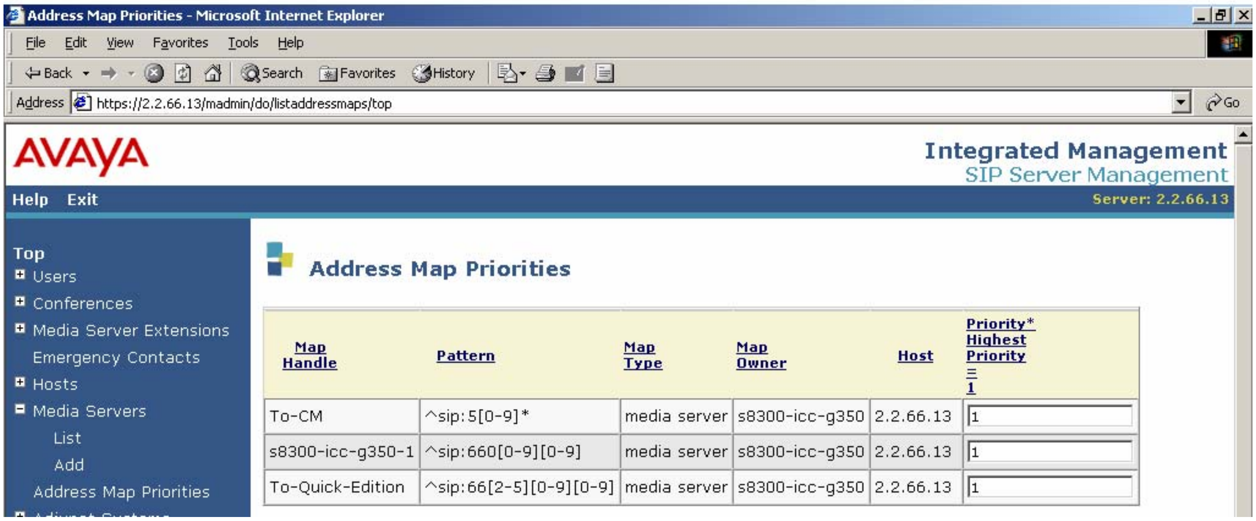
The following screen illustrates the Host configuration, which can be accessed via **Hosts → List**. The IP Address 2.2.66.13 that is used as the SIP Proxy Server for the SIP Telephones, and the far-end of the Avaya Communication Manager SIP signaling group can be observed.



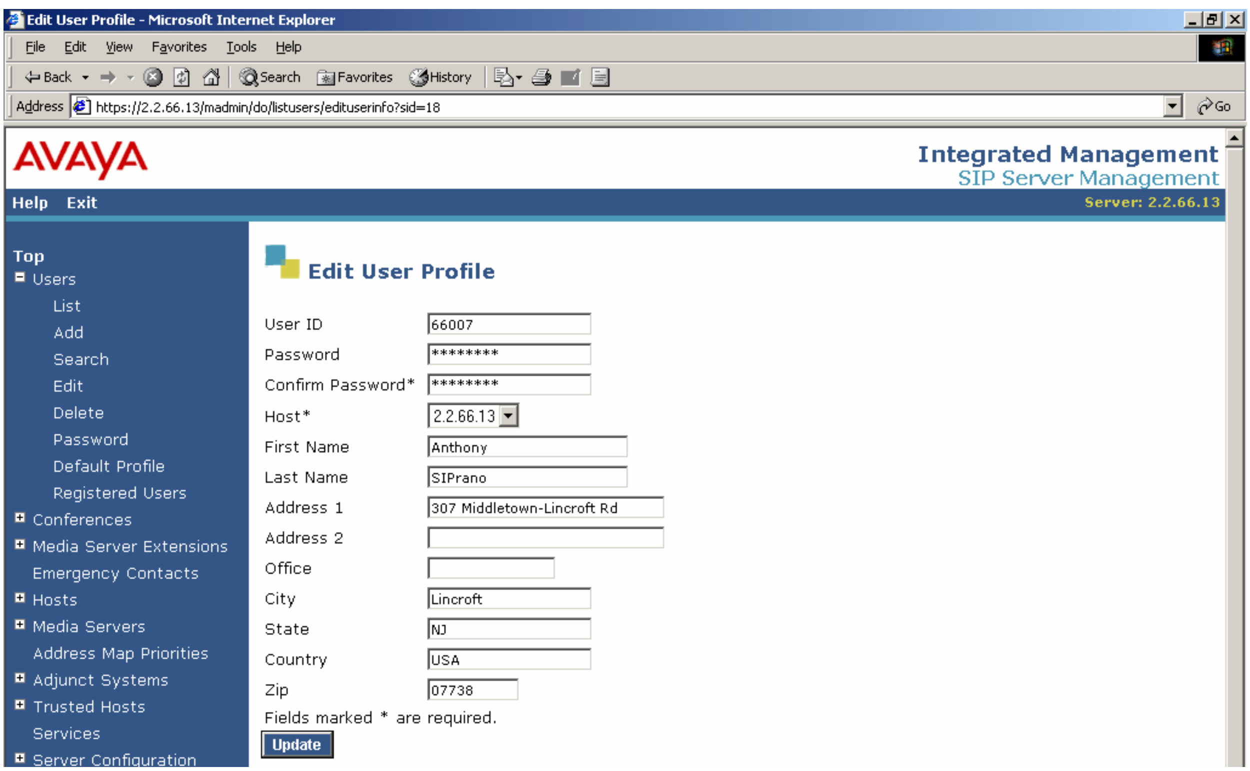
The following screen illustrates the Media Server configuration for the S8300 Server, accessible via **Media Servers → List**. The SIP Trunk using TLS to the IP Address of the S8300 Server at 2.2.35.88 can be observed.



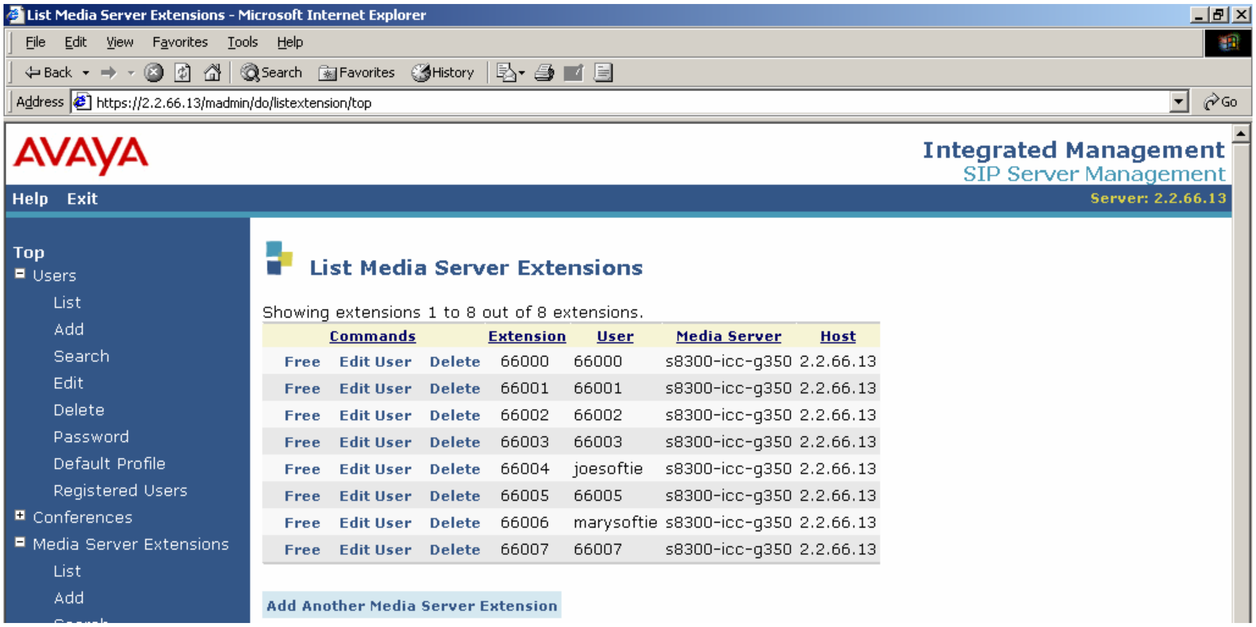
The following screen illustrates a simple Address Map configuration which includes the dial strings such as 5XXXX and 660XX used in these Application Notes. This screen is accessible via **Media Servers → Address Map Priorities**.



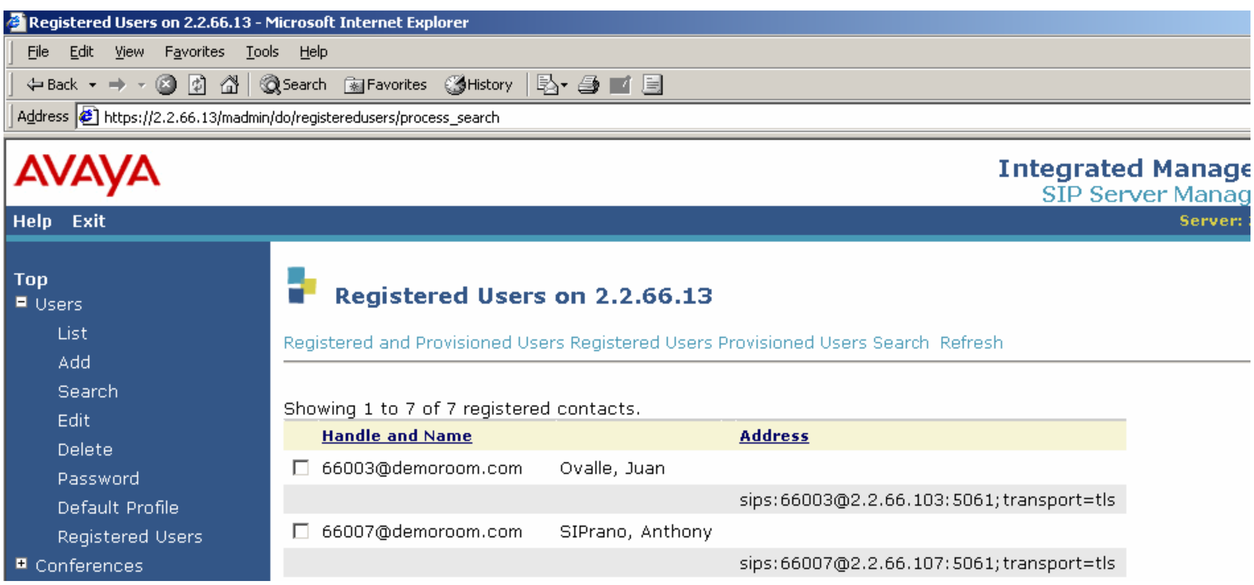
The following screen illustrates a sample SES user, accessible via **Users → Edit**.



The following screen illustrates the media server extensions and the corresponding users. This screen is accessible via **Media Server Extensions → List.**



The following screen illustrates a subset of the users registered with the Avaya SES. Note the use of transport type TLS, which is required for SRTP. This screen is accessible via **Users → Registered Users**.



4.3. 9600-Series SIP Telephones and 46xxsettings.txt Parameters

This section presents information specific to SRTP and G.722 for the Avaya 9600-Series SIP Telephones in the sample configuration. For additional information, please consult the Avaya one-X Deskphone SIP Administrators Guide in reference [12].

The 9600-Series SIP Telephones must use TLS for SRTP to be used for a call (i.e., the parameter SIPSIGNAL is at the default value "2" for TLS over TCP). If not prohibited by security configuration, the telephone's craft procedures can be used to verify the use of TLS. Check that the "Transport Type" parameter is set to TLS under the craft SIP menu. Alternatively, the transport type can be checked using the Avaya SES Administrative interface via the list of registered users. An example screen showing that the registered telephones are using TLS is shown as the last screen capture in Section 4.2.

The following screen shows a portion of the 46xxsettings.txt file acquired by the IP Telephones in the sample configuration. Note that G.722 must be enabled via the settings file as shown below.

```
## SIP Domain is “demoroom.com” consistent with Communication Manager and SES
SET SIPDOMAIN demoroom.com

## SIP Proxy address and MWI Server is the logical IP address for the duplicated SES
SET SIPPROXYSRVR 2.2.66.13
SET MWISRVR 2.2.66.13

####################  SRTP Settings #######################
## 1 = aescm128-hmac80
## 2 = aescm128-hmac32
## 3 = aescm128-hmac80-unauth
## 4 = aescm128-hmac32-unauth
## 5 = aescm128-hmac80-unenc
## 6 = aescm128-hmac32-unenc
## 7 = aescm128-hmac80-unenc-unauth
## 8 = aescm128-hmac32-unenc-unauth
## 9 = none (default)

## The following enables profile 1 in the comment above: aescm128-hmac80
## Other profiles can be allowed using a comma-separated list such as “1,9”
## The default value for MEDIAENCRYPTION is “1,9”

SET MEDIAENCRYPTION "1"

## The following enables G.722.  By default, ENABLE_G722 is 0, meaning disabled.
SET ENABLE_G722 1

## Although not required, the S8300 Server was used to serve the 46xxsettings.txt
## and binaries in the sample configuration.  The S8300 expected https on port 411.
SET TLSPORT 411
```

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 Server.

The following screen illustrates a subset of the registered H.323 IP stations, versions, and network regions. The extensions depicted in **Figure 2** are in bold.

```
list registered-ip-stations
                            REGISTERED IP STATIONS

Station Ext/  Set     Product    Prod  Station         Net Gatekeeper     TCP
 Orig Port    Type    ID         Rel   IP Address      Rgn IP Address     Skt
58002         4610    IP_Phone   2.800 2.2.35.201      1   2.2.35.88       y
58020         9620#   IP_Phone   1.500 2.2.35.203      1   2.2.35.88       y
58030         9630    IP_Phone   1.500 2.2.35.209      1   2.2.35.88       y
58050         9650    IP_Phone   1.500 2.2.35.202      1   2.2.35.88       y
```

The following screen shows that the Avaya G350 Media Gateway (2.2.35.87) is registered to the S8300 Server (2.2.35.88), and is in network region 1. The IP Address of the Avaya G350 Media Gateway will feature prominently in the verifications.

```
list media-gateway
                             MEDIA-GATEWAY REPORT

Num Name                     Serial No/       IP Address/       Type    NetRgn Reg?
                             FW Ver/HW Vint   Cntrl IP Addr             RecRule

1   G350-Right               03IS07589449     2  .2  .35 .87   g350      1       y
                             26 .33 .0  /1    2  .2  .35 .88             none
```

More detail is presented for the first few example calls to show the procedures to obtain the information.

5.1. Extension 58020 (9620 H.323) Calls Extension 66007 (9630 SIP)

This section illustrates the behavior when an Avaya 9600-Series IP Telephone running H.323 firmware (x58020) calls an Avaya 9600-Series IP Telephone (x66007) running SIP firmware. The call is routed from Avaya Communication Manager over the SIP Trunk to the Avaya SES. Since the Avaya 9600-Series IP Telephones support G.722-64K, this preferred codec is used. Since the Avaya 9600-Series IP Telephones support SRTP, the media path can "shuffle" to "ip-direct", meaning that media processing resources from the gateway are not required in the final connection path. The following screen shows the annotated trace output for such a call, taken from the Avaya S8300 Server.

```
list trace station 58020                                           Page   1
                                LIST TRACE
time            data
17:00:36     active station    58020 cid 0x270
17:00:36     G711MU ss:off ps:20 rn:1/1 2.2.35.203:2712 2.2.35.87:2056
17:00:36     xoip: fax:Relay modem:off tty:US 2.2.35.87:2056 uid:0x954
17:00:39     dial 66007
17:00:39     term station    66007 cid 0x270
17:00:42     active station    66007 cid 0x270
17:00:42     G72264K ss:off ps:20 rn:66/1 2.2.66.107:5004 2.2.35.203:2712
17:00:42     G72264K ss:off ps:20 rn:1/66 2.2.35.203:2712 2.2.66.107:5004
```

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: S00008         Switch-End IP Signaling Loc: PROCR    H.245 Port:
            IP Address                                Port  Node Name       Rgn
Switch-End:   2.  2. 35. 88                           1720                  1
   Set End:   2.  2. 35.203                           2349                  1
```

The codec used is G.722-64K and the connection is "ip-direct" between the two 9600-Series IP Telephones. Note that the region of the "Other-End" is region 66, the far-end network region configured for the SIP signaling group used for the call.

```
status station 58020                                        Page   4 of   7
                          AUDIO CHANNEL Port: S00008
G.722-64K     Switch-End Audio Location:
            IP Address                                Port  Node Name       Rgn
Other-End:   2.  2. 66.107                            5004                  66
  Set-End:   2.  2. 35.203                            2712                  1
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: S00008
 Product ID-Release: IP_Phone   1.500         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
```

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 9600-Series IP Telephones, one H.323, and the other at the far-end of the SIP Trunk.

```
status station 58020                                            Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: S00008
S00008:TX:2.2.35.203:2712/g722-64/20ms/1-srtp-aescm128-hmac80
T00053:RX:2.2.66.107:5004/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: T00053
```

The "status trunk 66" command can be used to see which trunk member is handing the call.

```
status trunk 66
                             TRUNK GROUP STATUS
Member   Port     Service State      Mtce Connected Ports
                                     Busy
0066/001 T00049   in-service/idle    no
0066/002 T00050   in-service/idle    no
0066/003 T00051   in-service/idle    no
0066/004 T00052   in-service/idle    no
0066/005 T00053   in-service/active  no   S00008
0066/006 T00054   in-service/idle    no
0066/007 T00055   in-service/idle    no
0066/008 T00056   in-service/idle    no
0066/009 T00057   in-service/idle    no
0066/010 T00058   in-service/idle    no
```

The "status trunk 66/5" command shows similar information for this same call obtained using the "status trunk" command for trunk member 5 handling the call. The signaling is between the S8300 Server at 2.2.35.88 (near-end of signaling group 66) and the active Avaya SES at 2.2.66.13 (far-end of signaling group 66). The media path is ip-direct between the two telephones.

```
status trunk 66/5                                               Page   1 of   2
                              TRUNK STATUS
 Trunk Group/Member: 0066/005              Service State: in-service/active
              Port: T00053             Maintenance Busy? no
    Connected Ports: S00008

                  Port     Near-end IP Addr : Port    Far-end IP Addr : Port
        Signaling: 01A0017    2.  2. 35. 88  : 5061      2.  2. 66. 13 : 5061
G.722-64K   Audio:            2.  2. 35.203  : 2712      2.  2. 66.107 : 5004
```

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

```
status trunk 66/5                                              Page   2 of   2
                      SRC PORT TO DEST PORT TALKPATH
src port: T00053
T00053:TX:2.2.66.107:5004/g722-64/20ms/1-srtp-aescm128-hmac80
S00008:RX:2.2.35.203:2712/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: S00008
```

5.2. Extension 66007 (9630 SIP) Calls Extension 58020 (9620 H.323)

This section illustrates the behavior when an Avaya 9600-Series Telephone running SIP firmware (66007) calls an Avaya 9600-Series IP Telephone running H.323 firmware (x58020). The call arrives from SIP trunk 66, which can be traced using the “list trace tac” command as shown below.

```
list trace tac 166                                                    Page   1
                                LIST TRACE
time            data
17:08:35     Calling party station    66007 cid 0x273
17:08:35     Calling Number & Name 66007 Anthony SIPra
17:08:35     active station    66007 cid 0x273
17:08:37     G711MU ss:off ps:20 rn:66/1 2.2.66.107:5004 2.2.35.87:2056
17:08:37     xoip: fax:Relay modem:off tty:US 2.2.35.87:2056 uid:0x50031
17:08:37     dial 58020
17:08:37     ring station    58020 cid 0x273
17:08:37     G711MU ss:off ps:20 rn:1/1 2.2.35.203:2712 2.2.35.87:2052
17:08:37     xoip: fax:Relay modem:off tty:US 2.2.35.87:2052 uid:0x954
             VOIP data from: 2.2.35.87:2052
17:08:40     active station    58020 cid 0x273
17:08:41     G72264K ss:off ps:20 rn:66/1 2.2.66.107:5004 2.2.35.203:2712
17:08:41     G72264K ss:off ps:20 rn:1/66 2.2.35.203:2712 2.2.66.107:5004
```

The following screen shows a subset of the information available from the “status station” command for the called station. The connection is G.722 “ip-direct” between the two 9600-Series IP Telephones, and the “1-srtp-aescm128-hmac80” profile is in evidence.

```
status station 58020                                           Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: S00008
S00008:TX:2.2.35.203:2712/g722-64/20ms/1-srtp-aescm128-hmac80
T00049:RX:2.2.66.107:5004/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: T00049
```

Similar information is available using the “status trunk” command as shown in the following screen.

```
status trunk 66/1                                              Page   2 of   2
                      SRC PORT TO DEST PORT TALKPATH
src port: T00049
T00049:TX:2.2.66.107:5004/g722-64/20ms/1-srtp-aescm128-hmac80
S00008:RX:2.2.35.203:2712/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: S00008
```

5.3. Extension 66007 (9630 SIP) Calls Extension 66003 (9620 SIP)

This section illustrates the behavior when an Avaya 9600-Series Telephone running SIP firmware (x66007) calls another Avaya 9600-Series IP Telephone (x66003) running SIP firmware. The call arrives from SIP trunk group 66 and can be traced as in the previous section.

```
list trace tac 166                                                  Page   1
                                LIST TRACE
time            data
17:13:35     Calling party station    66007 cid 0x274
17:13:35     Calling Number & Name 66007 Anthony SIPra
17:13:35     active station    66007 cid 0x274
17:13:37     G711MU ss:off ps:20 rn:66/1 2.2.66.107:5004 2.2.35.87:2050
17:13:37     xoip: fax:Relay modem:off tty:US 2.2.35.87:2050 uid:0x50031
17:13:37     dial 66003
17:13:37     term station    66003 cid 0x274
17:13:43     active station    66003 cid 0x274
17:13:44     G72264K ss:off ps:20 rn:66/66 2.2.66.103:5004 2.2.66.107:5004
17:13:44     G72264K ss:off ps:20 rn:66/66 2.2.66.107:5004 2.2.66.103:5004
```

The following screen shows a subset of the information available from the “status trunk” command. Observe that two trunk members are used, one for each SIP telephone.

```
status trunk 66
                             TRUNK GROUP STATUS
Member   Port     Service State      Mtce Connected Ports
                                     Busy
0066/001 T00049   in-service/active  no   T00054
0066/002 T00050   in-service/idle    no
0066/003 T00051   in-service/idle    no
0066/004 T00052   in-service/idle    no
0066/005 T00053   in-service/idle    no
0066/006 T00054   in-service/active  no   T00049
0066/007 T00055   in-service/idle    no
0066/008 T00056   in-service/idle    no
0066/009 T00057   in-service/idle    no
0066/010 T00058   in-service/idle    no
```

The following status screen shows the ip-direct media path between the two telephones, which uses G.722 and the “1-srtp-aescm128-hmac80” encryption profile.

```
status trunk 66/1                                            Page   2 of   2
                      SRC PORT TO DEST PORT TALKPATH
src port: T00049
T00049:TX:2.2.66.107:5004/g722-64/20ms/1-srtp-aescm128-hmac80
T00054:RX:2.2.66.103:5004/g722-64/20ms/1-srtp-aescm128-hmac80
dst port: T00054
```

5.4. Extension 58002 (4610 H.323) Calls Extension 66007 (9630 SIP)

This section illustrates the behavior when an Avaya 4600-Series IP Telephone running H.323 firmware (x58002) calls an Avaya 9600-Series IP Telephone running SIP firmware (x66007).

Since the Avaya 4600-Series Telephone does not support SRTP, but the configured inter-region codec set has been configured to require SRTP, a media processor from the G350 Media Gateway is utilized to ensure SRTP is used over the SIP Trunk. Since the G350 Media Gateway media processor does not support G.722-64K, the G.711MU codec is used. The answering 9630 SIP 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 H.323 telephone. The intra-region codec used is G.711MU, and the connection is "ip-tdm" from the telephone at 2.2.35.201 to the Avaya G350 Media Gateway at 2.2.35.87, using G.711MU and AES. The connection across the SIP trunk from the Avaya G350 Media Gateway to the Avaya 9630 SIP Telephone uses G.711MU and SRTP.

```
status station 58002                                        Page   7 of   8
                      SRC PORT TO DEST PORT TALKPATH
src port: S00054
S00054:TX:2.2.35.201:2946/g711u/20ms/aes
001V037:RX:2.2.35.87:2052/g711u/20ms/aes:TX:ctxID:9
001V035:RX:ctxID:9:TX:2.2.35.87:2056/g711u/20ms/1-srtp-aescm128-hmac80
T00055:RX:2.2.66.107:5004/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00055
```

The following screen shows similar information available from the "status trunk" command for the SIP trunk handling the call. The connection is "ip-tdm" from the telephone at 2.2.35.201 to the Avaya G350 Media Gateway at 2.2.35.87, using G.711MU and AES. The connection across the SIP trunk from the Avaya G350 Media Gateway to the Avaya 9630 SIP Telephone uses G.711MU and SRTP.

```
status trunk 66/7                                           Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00055
T00055:TX:2.2.66.107:5004/g711u/20ms/1-srtp-aescm128-hmac80
001V035:RX:2.2.35.87:2056/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:9
001V037:RX:ctxID:9:TX:2.2.35.87:2052/g711u/20ms/aes
S00054:RX:2.2.35.201:2946/g711u/20ms/aes
dst port: S00054
```

5.5. Extension 66007 (9630 SIP) Calls Extension 58002 (4610 H.323)

This case is not substantially different than those previously presented. A "status station" screen showing the end-end media path is shown below. The connection topology is the same as in the prior section where the same end-user devices were used for a call in the opposite call direction.

```
status station 58002                                          Page   7 of   8
                      SRC PORT TO DEST PORT TALKPATH
src port: S00054
S00054:TX:2.2.35.201:2946/g711u/20ms/aes
001V034:RX:2.2.35.87:2058/g711u/20ms/aes:TX:ctxID:5
001V038:RX:ctxID:5:TX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80
T00049:RX:2.2.66.107:5004/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00049
```

5.6. Examples of Local Calls For the S8300 Server

Examples of calls among H.323 extensions 58002, 58003, 58020, and 58030, on the Avaya S8300 Server are shown in detail in reference [1]. There is no change to the behavior of these calls.

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 Server was used to enable six telephones to dial-in to join a conference. Four of the telephones (x58002, x58003, x58020, x58030) were digital or H.323 IP Telephones controlled by the S8300 Server. The other two telephones are Avaya 9600-Series IP Telephones running SIP firmware (x66003, x66007) dialed in over the SIP trunk group to the conference.

To aid in understanding the status screens in this section, the order of the telephones joining the meet-me conference were 66003 (9600-Series SIP), 58020 (9600-Series H.323), 66007 (9600-Series SIP), 58030 (9600-Series H.323), 58003 (digital), 58002 (4600-Series H.323). 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 SIP telephones.

```
status meet-me-vdn 58081
                                          Service State: active
 Extension: 58081
 Conferee Ports
 T00049        S00008      T00050      S00056      001V401     S00054
```

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 SIP station 66003. Observe the use of SRTP between the G350 Media Gateway VoIP and the

9600-Series SIP Telephone, and the use of SRTP between the G350 Media Gateway VoIP and the Avaya 9600-Series H.323 telephone (x58020, 2.2.35.203).

```
status trunk 66/1                                              Page   3 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: T00049
T00049:TX:2.2.66.103:5004/g711u/20ms/1-srtp-aescm128-hmac80
001V037:RX:2.2.35.87:2052/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:15
001V038:RX:ctxID:15:TX:2.2.35.87:2050/g711u/20ms/1-srtp-aescm128-hmac80
S00008:RX:2.2.35.203:2712/g711u/20ms/1-srtp-aescm128-hmac80
dst port: S00008
```

Paging forward, the following screen shows a leg of the conference connection involving SIP station 66007 at the far-end of the SIP trunk (T00050). Observe the use of G.711MU and SRTP across the trunk.

```
status trunk 66/1                                              Page   4 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: T00049
T00049:TX:2.2.66.103:5004/g711u/20ms/1-srtp-aescm128-hmac80
001V037:RX:2.2.35.87:2052/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:15
001V035:RX:ctxID:15:TX:2.2.35.87:2056/g711u/20ms/1-srtp-aescm128-hmac80
T00050:RX:2.2.66.107:5004/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00050
```

Paging forward, the following screen shows a leg of the conference connection involving H.323 station 58030 (S00056, 2.2.35.209). Observe the use of G.711MU and SRTP to the 9600-Series IP Telephone running H.323 firmware.

```
status trunk 66/1                                              Page   5 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: T00049
T00049:TX:2.2.66.103:5004/g711u/20ms/1-srtp-aescm128-hmac80
001V037:RX:2.2.35.87:2052/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:15
001V034:RX:ctxID:15:TX:2.2.35.87:2058/g711u/20ms/1-srtp-aescm128-hmac80
S00056:RX:2.2.35.209:2370/g711u/20ms/1-srtp-aescm128-hmac80
dst port: S00056
```

Paging forward, the following screen shows a leg of the conference connection involving digital station 58003 (001V401). Observe the use of G.711MU and SRTP over the SIP trunk to the gateway serving the digital phone.

```
status trunk 66/1                                              Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: T00049
T00049:TX:2.2.66.103:5004/g711u/20ms/1-srtp-aescm128-hmac80
001V037:RX:2.2.35.87:2052/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:15
001V401:RX:ctxID:15
dst port: 001V401
```

Paging forward, the following screen shows a leg of the conference connection involving Avaya 4600-Series H.323 IP station 58002 (S00054, 2.2.35.201). Observe the use of G.711MU and

SRTP over the SIP trunk to the gateway, and the use of AES from the gateway to the Avaya 4600-Series IP Telephone running H.323 firmware.

```
status trunk 66/1                                              Page   7 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: T00049
T00049:TX:2.2.66.103:5004/g711u/20ms/1-srtp-aescm128-hmac80
001V037:RX:2.2.35.87:2052/g711u/20ms/1-srtp-aescm128-hmac80:TX:ctxID:15
001V033:RX:ctxID:15:TX:2.2.35.87:2060/g711u/20ms/aes
S00054:RX:2.2.35.201:2946/g711u/20ms/aes
dst port: S00054
```

5.8. Using Avaya 1600-Series IP Telephones

The Avaya 1600-Series IP Telephones Release 1, also known as one-X Deskphone Value Edition, support Avaya media encryption using AES, but do not support G.722 or SRTP. Consult reference [13] for additional information. With the Avaya Communication Manager codec set configuration presented in these Application Notes, calls involving an Avaya 1600-Series IP Telephone added to IP Network Region 1 use the same codec and encryption as would an Avaya 4600-Series IP Telephone. A representative sample of calls involving Avaya 1600-Series IP Telephones is presented below to reinforce the expected behavior.

The following screen shows how the Avaya 1600-Series IP Telephones may be aliased as 4600-Series IP Telephones.

```
change alias station                                           Page   1 of   1
                              ALIAS STATION
                Alias Set Type            Supported Set Type
                   1603                        4610
                   1608                        4610
                   1616                        4620
```

The bolded rows in the following screen show that several Avaya 1600-Series IP Telephones were registered with Avaya Communication Manager in network region 1.

```
list registered-ip-stations

                            REGISTERED IP STATIONS
Station Ext/  Set     Product    Prod  Station         Net Gatekeeper     TCP
 Orig Port    Type    ID         Rel   IP Address      Rgn IP Address     Skt
58002         4610    IP_Phone   2.800 2.2.35.201      1   2.2.35.88       y
58020         9620#   IP_Phone   1.500 2.2.35.203      1   2.2.35.88       y
58030         9630    IP_Phone   1.500 2.2.35.209      1   2.2.35.88       y
58050         9650    IP_Phone   1.500 2.2.35.202      1   2.2.35.88       y
58161         1616    IP_Phone   1.  0 2.2.66.166      1   2.2.35.88       y
58163         1603    IP_Phone   1.  0 2.2.66.163      1   2.2.35.88       y
58166         1616    IP_Phone   1.  0 2.2.66.161      1   2.2.35.88       y
58168         1608    IP_Phone   1.  0 2.2.66.160      1   2.2.35.88       y
```

Like the Avaya 4600-Series and Avaya 9600-Series IP Telephones running H.323 firmware, the Avaya 1600-Series IP Telephones Release 1.0 support the “Annex H” (H.235.5) secure registration procedures. No change is required to the IP network region 1 configuration shown in Section 4.1.4. The following screen shows the registration status for station 58168.

```
status station 58168                                           Page   5 of   8
                            IP ENDPOINT DATA
               Port: S00063
 Product ID-Release: IP_Phone   1.  0         H.245 Tunneled? does not apply
Registration Status: registered-authenticated    MAC Address: 00:04:0d:f5:e5:aa
Authentication Type: Avaya AnxH              Dependency Mode: main
```

The following screen shows the final connection status for a call from 1600-Series telephone extension 58168 to 9600-Series SIP telephone extension 66007. Observe that the connection uses G.711MU and AES media encryption from the 1600-Series telephone (2.2.66.160) to the Avaya G350 Media Gateway, and SRTP from the G350 Media Gateway to the 9600-Series telephone (2.2.66.107). This connection topology is the same as shown in Section 5.4, which illustrated a 4600-Series H.323 telephone calling a 9600-Series SIP telephone.

```
status station 58168                                           Page   7 of   8
                      SRC PORT TO DEST PORT TALKPATH
src port: S00063
S00063:TX:2.2.66.160:2126/g711u/20ms/aes
001V032:RX:2.2.35.87:2062/g711u/20ms/aes:TX:ctxID:282
001V034:RX:ctxID:282:TX:2.2.35.87:2058/g711u/20ms/1-srtp-aescm128-hmac80
T00058:RX:2.2.66.107:5004/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00058
```

The following screen shows the final connection status for a call from 1600-Series telephone extension 58168 to 9600-Series H.323 telephone extension 58020. Observe that the connection is “ip-direct” using G.711MU and AES media encryption from the 1600-Series telephone (2.2.66.160) to the 9600-Series H.323 telephone (2.2.35.203). This connection topology is the same as shown in reference [1] for local calls between 4600-Series H.323 and 9600-Series H.323 IP Telephones.

```
status station 58168                                           Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: S00063
S00063:TX:2.2.66.160:2126/g711u/20ms/aes
S00008:RX:2.2.35.203:2712/g711u/20ms/aes
dst port: S00008
```

This same connection type (G.711 MU and AES) would apply to a call between a 1600-Series telephone and a 4600-Series H.323 telephone, or a call between two 1600-Series IP Telephones. The following screen shows the final connection status for a call from 1600-Series telephone extension 58168 to 1600-Series telephone extension 58166. Observe that the connection is “ip-direct” using G.711MU and AES media encryption between the two 1600-Series IP Telephones.

```
status station 58168                                           Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: S00063
S00063:TX:2.2.66.160:2126/g711u/20ms/aes
S00068:RX:2.2.66.161:2502/g711u/20ms/aes
dst port: S00068
```

5.9. Using Avaya IP Softphone in Road Warrior Mode

When using Avaya IP Softphone in "road warrior" mode (e.g., with a headset), the IP Softphone processes the media for the call. Like the Avaya 4600-Series and 1600-Series IP Telephones, the IP Softphone supports AES media encryption, but does not support SRTP or G.722. With the Avaya Communication Manager codec set configuration presented in these Application Notes, calls involving an Avaya IP Softphone in "road warrior" mode in IP network region 1 will use the same codec and encryption as would an Avaya 4600-Series IP Telephone.

Like the Avaya 1600-Series, 4600-Series, and 9600-Series IP Telephones running H.323 firmware, the Avaya IP Softphone Release 6 supports the "Annex H" (H.235.5) secure registration procedures. No change is required to the IP network region 1 configuration shown in Section 4.1.4. The following screen shows the registration status for an IP Softphone Release 6 registered as extension 58022.

```
status station 58022                                            Page   5 of   6
                            IP ENDPOINT DATA
               Port: S00043
 Product ID-Release: IP_Soft    5.242         H.245 Tunneled? does not apply
Registration Status: registered-authenticated    MAC Address: 00:0f:1f:23:eb:30
Authentication Type: Avaya AnxH              Dependency Mode: main
```

The following screen shows the final connection status for a call from an Avaya IP Softphone Release 6 to 9600-Series SIP telephone extension 66007. Observe that the connection uses G.711MU and AES media encryption from the Avaya IP Softphone (2.2.35.200) to the Avaya G350 Media Gateway, and SRTP from the G350 Media Gateway to the 9600-Series telephone (2.2.66.107). This connection topology is the same as shown in Section 5.4, which illustrated a 4600-Series H.323 IP Telephone calling a 9600-Series SIP IP Telephone.

```
status station 58022                                            Page   7 of   8
                      SRC PORT TO DEST PORT TALKPATH
src port: S00043
S00043:TX:2.2.35.200:2048/g711u/20ms/aes
001V037:RX:2.2.35.87:2052/g711u/20ms/aes:TX:ctxID:98
001V032:RX:ctxID:98:TX:2.2.35.87:2062/g711u/20ms/1-srtp-aescm128-hmac80
T00051:RX:2.2.66.107:5004/g711u/20ms/1-srtp-aescm128-hmac80
dst port: T00051
```

The following screen shows the final connection status for a call from this same Avaya IP Softphone to 9600-Series H.323 telephone extension 58020. Observe that the connection is "ip-direct" using G.711MU and AES media encryption from the Avaya IP Softphone (2.2.35.200) to the 9600-Series H.323 telephone (2.2.35.203).

```
status station 58022                                          Page   6 of   7
                      SRC PORT TO DEST PORT TALKPATH
src port: S00043
S00043:TX:2.2.35.200:2048/g711u/20ms/aes
S00008:RX:2.2.35.203:2712/g711u/20ms/aes
dst port: S00008
```

Reference [1] includes additional call verifications using the Avaya IP Softphone shown in **Figure 1**.

5.10. Using Avaya 4600-Series IP Telephones Running SIP Firmware

When using SRTP in conjunction with Avaya Communication Manager, any Avaya 4600-Series SIP Telephones must be upgraded to Release 2.225. See Product Support Notice PSN 1245U for additional information: http://support.avaya.com/elmodocs2/PSN/PSN1245u.pdf

The configuration and verifications presented to this point in these Application Notes have required SRTP across the SIP trunk to illustrate the sample objectives listed in Section 1.1. However, at time of writing, Avaya 4600-Series IP Telephones running SIP firmware support neither SRTP nor AES media encryption. If an Avaya 4600-Series IP Telephone running SIP firmware were configured in the same fashion (network region 66, OPS using trunk group 66, etc.) as the 9600-Series IP Telephones and no changes were made to the configuration presented in these Application Notes, calls to and from the Avaya 4600-Series SIP Telephones would be denied due to a codec mismatch (i.e., codec requires SRTP, but 4600-Series SIP Telephone does not support SRTP). If it is desired to introduce Avaya 4600-Series IP Telephones into the configuration, the simplest change to the configuration presented in these Application Notes would be to allow “none” in ip-codec-set 3 as shown in the following screen.

```
change ip-codec-set 3                                         Page   1 of   2
                          IP Codec Set
    Codec Set: 3

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.722-64K                   2        20
 2: G.711MU          n          2        20
 3:
 4:
 5:
 6:
 7:
     Media Encryption
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
```

With this one change, all call verifications previously presented would remain the same. Calls to and from the newly introduced Avaya 4600-Series SIP Telephones would be allowed and media paths achieved, but recognize that any connection legs involving the 4600-Series SIP Telephones would not use media encryption.

6. Conclusion

As illustrated in these Application Notes, the Secure Real-Time Transport Protocol (SRTP) available in Avaya Communication Manager can be used to secure connections over SIP Trunks to Avaya 9600-Series SIP telephones registered with Avaya SES. 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 - 3] are among the Application Notes available at http://www.avaya.com. Click on the Application Notes link from the Resource Library available here:

http://www.avaya.com/gcm/master-usa/en-us/resource/index.htm

The first reference illustrates the use of SRTP and G.722 using H.323 IP Telephones and trunks.

[1] Configuring Avaya Communication Manager Release 4.0 to Use Secure Real-Time Transport Protocol (SRTP) over H.323 IP Trunks, Issue 1.0.

http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/srtp-iptrunk.pdf

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

The following Application Notes provide a procedural example for configuring the Avaya SES with Avaya Communication Manager in support of SIP Telephones:

[3] Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Snom 190/220/360 SIP Telephones, Issue 1.0

http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/snomsip.pdf

Reference [4] includes information related to H.235.5 (Annex H).

[4] 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 [5] is a link to The Secure Real-Time Transport Protocol, RFC 3711.

[5] http://www.ietf.org/rfc/rfc3711.txt

Reference [6] is an example of relevant Avaya Communication Manager Release 4 product documentation available at http://support.avaya.com

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

Reference [7] is a tutorial on Avaya Communication Manager network regions.

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

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

References [8-13] are general guides to the solutions included in these Application Notes.

[8] Avaya Extension to Cellular and OPS Installation and Administration Guide, Version 6.0 Issue 9, DocID 210-100-500, June 2005, available at http://support.avaya.com

[9] SIP Support in Avaya Communication Manager Running on the S8300, S8400, S8500 Series, and S8700 Series Media Server, Issue 1.6, Doc ID 555-245-206, March, 2007, available at http://support.avaya.com

[10] Administrator Guide for Avaya Communication Manager, Issue 3.1, Doc ID 03-300509, February 2007, available at http://support.avaya.com

[11] Installing and Administering SIP Enablement Services, Issue 4, Doc ID 03-600768, May, 2007, available at http://support.avaya.com

[12] Avaya one-X Deskphone SIP for 9600-Series IP Telephones Administrator Guide, Release 1.0, Doc ID 16-601944, May 2007, available at http://support.avaya.com

[13] Avaya one-X Value Edition 1600-Series IP Telephones Administrators Guide, Release 1.0, Doc ID 16-601443, Issue 1, June 2007

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