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Application Notes for Configuring SIP Trunking between the COLT VoIP Access SIP Service and an Avaya IP Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure trunking using the Session Initiation Protocol (SIP) between the COLT VoIP Access SIP Service and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya IP Telephones.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the procedure for configuring SIP trunking between the COLT VoIP Access SIP trunking network and Avaya Communication Manager. The COLT VoIP Access service allows customers to connect their Avaya Communication Manager to the PSTN via an IP network.
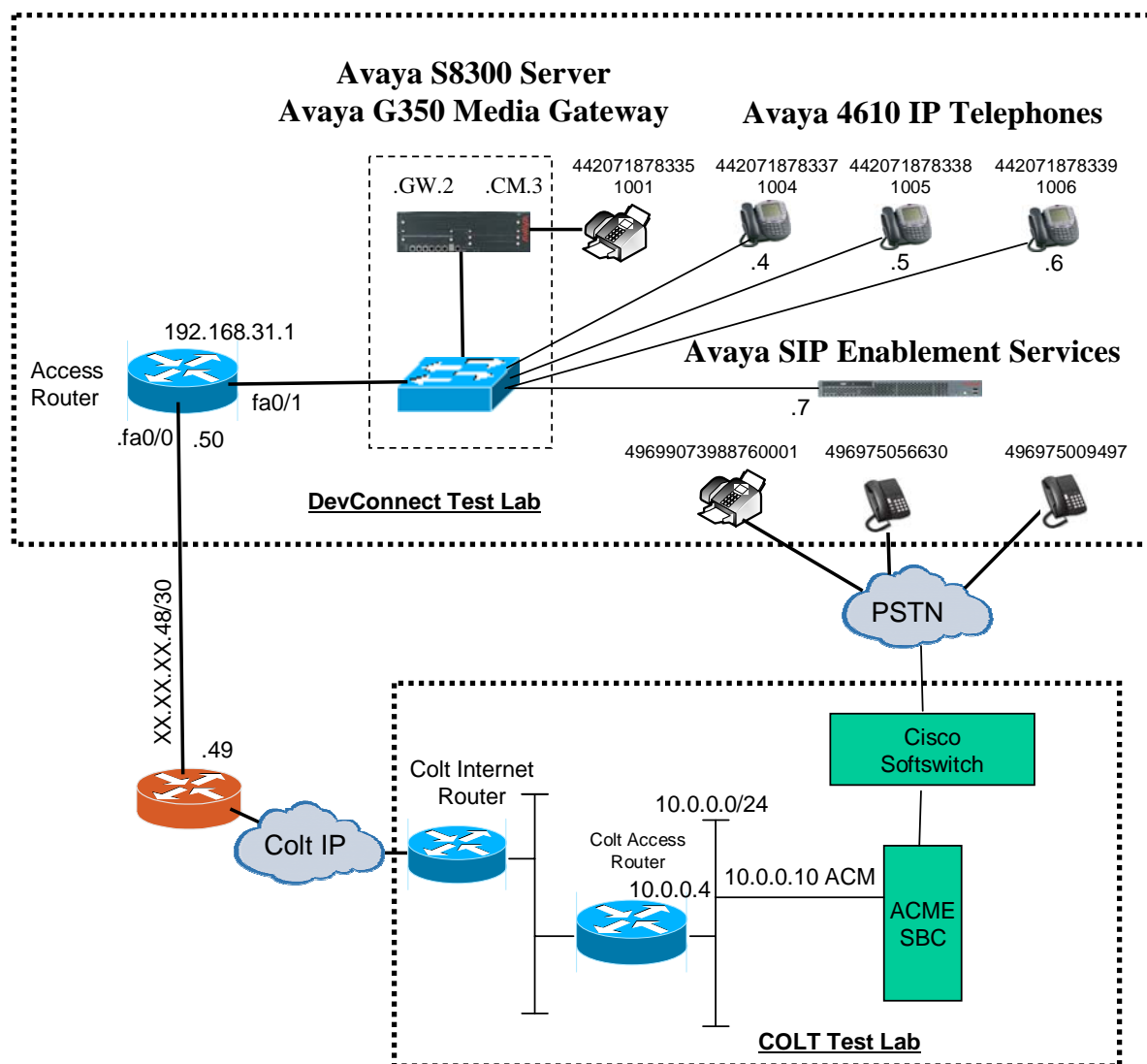
Figure 1: System Configuration

In the above diagram, Avaya IP Telephones and other telephone endpoints are attached to an Avaya S8300 Server running Avaya Communication Manager via an Avaya G350 Media Gateway. The Avaya SIP Enablement Services (SES) server serves as a registrar for local SIP endpoints and serves as the interface to the COLT SIP trunk (note that public IP addresses are not shown, for security reasons). Avaya 4610 IP Telephones configured for H.323 were used for

testing. The same telephones configured for SIP operation were tested, but only to verify the correct operation of incoming and outgoing calls.

The FAX and each of the Avaya IP Telephones registered with Avaya Communication Manager is assigned a PSTN telephone number which can be for FAX or the telephones.

Avaya Communication Manager and the COLT SIP network are configured to support direct IP connections, thus avoiding the necessity to route voice streams thorough the Avaya G350 Media Gateway. Avaya Communication Manager and the COLT SIP network are configured to support T.38 FAX transmission.

2. Equipment and Software Validated

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

Equipment	Software
Avaya Communication Manager	R014x.00.1.731.2
Avaya G350 Media Gateway	26.36.0
Avaya SIP Enablement Services	SES-4.0.0.0-033.6
Avaya 4610 IP Telephones (SIP)	2.2.2
Avaya 4610 IP Telephones (H.323)	2.8.3
ACME Session Border Controller	4.1.4 p24
Cisco Soft Switch	PGW 2200 version 9.7.3

Table 1: Equipment and Software Validated

3. Configuration

3.1. Avaya Communication Manager

The Avaya Communication Manager configuration was performed using the System Access Terminal (SAT).

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Avaya Communication Manager is licensed to meet the minimum requirements to interoperate with the COLT SIP network. Those items shown in bold indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Verify that the parameters are set as shown in the following table:

Parameter	Usage
Maximum Concurrently Registered IP Stations (p.2)	This parameter must be large enough to support the number of IP stations to be attached.
Maximum Administered SIP Trunks (p.2)	This parameter must be large enough to support the number of SIP trunks to be attached.
Enhanced EC500? (p.4)	This parameter must be set to "y".
IP Trunks? (p.4)	This parameter must be set to "y".
ISDN-PRI? (p.4)	This parameter must be set to "y".

Table 2: System-Parameters Customer-Options Parameters

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

IP PORT CAPACITIES                                              USED
                 Maximum Administered H.323 Trunks: 30    5
          Maximum Concurrently Registered IP Stations: 10    3
            Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
             Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0    0
               Maximum Video Capable H.323 Stations: 0     0
               Maximum Video Capable IP Softphones: 0      0
                   Maximum Administered SIP Trunks: 10    3

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                         Maximum TN2501 VAL Boards: 0      0
                      Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
 Maximum Number of Expanded Meet-me Conference Ports: 0      0
```

Figure 2: System-Parameters Customers-Options Form, Page 2

```
change system-parameters customer-options                     Page   4 of  10
                             OPTIONAL FEATURES

   Emergency Access to Attendant? y                          IP Stations? y
         Enable 'dadmin' Login? y
         Enhanced Conferencing? n                     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? n                Malicious Call Trace? n
     External Device Alarm Admin? n            Media Encryption Over IP? n
 Five Port Networks Max Per MCC? n   Mode Code for Centralized Voice Mail? n
               Flexible Billing? n
   Forced Entry of Account Codes? n                Multifrequency Signaling? y
       Global Call Classification? n     Multimedia Call Handling (Basic)? n
            Hospitality (Basic)? y    Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n
                      IP Trunks? y

             IP Attendant Consoles? n
```

Figure 3: System-Parameters Customers-Options Form, Page 4

3.1.2. Set system-parameters features

Use the **change system-parameters features** command to set the parameters as shown in the following table:

Parameter	Usage
Trunk-to-Trunk Transfer	Set this value to "all".

Table 3: System-Parameters Features Parameters

```
change system-parameters features                            Page   1 of  17
                        FEATURE-RELATED SYSTEM PARAMETERS
                            Self Station Display Enabled? n
                                 Trunk-to-Trunk Transfer: all
       Automatic Callback - No Answer Timeout Interval (rings): 3
                         Call Park Timeout Interval (minutes): 10
           Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? y
                                    Music/Tone on Hold: none
               Music (or Silence) on Transferred Trunk Calls? no
                        DID/Tie/ISDN/SIP Intercept Treatment: attd
        Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                      Automatic Circuit Assurance (ACA) Enabled? n




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

Figure 4: System-Parameters Features Form, Page 1

3.1.3. Configure Media Gateway

Use the **add media-gateway next** command to allocate a media gateway to be used as an interface to the COLT VoIP Access SIP Service and locally-attached endpoints, using the following parameters:

Parameter	Usage
Name	Assign a name to identify this unit.
Serial No:	Enter the serial number of the gateway.
Network Region	Enter the network region to be used by the gateway. Region "1" was used for the tested configuration.

Table 4: Media Gateway Parameters

```
add media-gateway next                                    Page   1 of   1
                          MEDIA GATEWAY
        Number: 1                           Registered?  n
          Type: g350              FW Version/HW Vintage:
          Name: G350                          IP Address:
      Serial No: ########          Controller IP Address:
    Encrypt Link? y                           MAC Address:
  Network Region: 1
        Location: 1                             Site Data:
    Recovery Rule: none

        Slot    Module Type            Name
        V1:
        V2:
        V3:
        V4:
        V5:
        V6:
        V7:                                Max Survivable IP Ext: 8
        V8:
        V9:
```

Figure 5: Media Gateway Form

3.1.4. SIP Interface to SIP Enablement Services

Use the **change node-names ip** command to assign the name "ses" to the IP address of the SES server.

```
change node-names ip                                      Page   1 of   2
                          IP NODE NAMES
    Name            IP Address
default         0.0.0.0
procr           192.168.31.3
ses             192.168.31.7
```

Figure 6: Node-Names Ip Form

Use the **add signaling-group** command to allocate a signaling group for interface to SES using the following parameters:

Parameter	Usage
Group Type	Enter "sip".
Near-end Node Name	Enter "procr" to designate the G350 processor as the near end node name.
Far-end Node Name	Enter "ses" to assign the SES server as the far end node name.
Direct IP-IP Audio Connections	Enter "y" to allow direct IP-IP endpoint connections (shuffling).

Table 5: Signaling-Group Parameters

```
add signaling-group 1                                           Page   1 of   1
                             SIGNALING GROUP

 Group Number: 1                   Group Type: sip
                         Transport Method: tls




  Near-end Node Name: procr                   Far-end Node Name: ses
Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                           Far-end Network Region:
      Far-end Domain:

                                          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): 3
```

Figure 7: Signaling-Group Form

Use the **add trunk-group <n>** command, were <n> is an unused trunk number, to allocate a trunk group to be used as an interface to the COLT VoIP Access SIP Service. Use the parameters show in the following table.

Parameter	Usage
Group Type (p.1)	Enter "sip".
Group Name (p.1)	Assign a name for identification purposes.
TAC (p.1)	Enter the Trunk Access Code allocated in **Figure 10**.
Service Type (p.1)	Enter "tie".
Signaling Group (p.1)	Enter the number of the signaling group allocated in **Figure 7**.
Number of Members (p.1)	Enter a number large enough to support the maximum number of anticipated simultaneous calls to be made via the SIP trunk.
Preferred Minimum Session Refresh Interval (p.2)	Enter "900" seconds, as required for the COLT SIP trunk interface. This should be half of the Session Refresh Interval which is configured for the COLT VoIP Access SIP Service.

Table 6: Dial Plan Analysis Parameters

```
add trunk-group 1                                               Page   1 of  21
                               TRUNK GROUP

Group Number: 1                     Group Type: sip          CDR Reports: y
  Group Name: SIP                           COR: 1      TN: 1      TAC: *01
   Direction: two-way      Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n


                                             Signaling Group: 1
                                           Number of Members: 5
```

Figure 8: Trunk-Group Form, p.1

```
add trunk-group 1                                               Page   2 of  21
     Group Type: sip

TRUNK PARAMETERS

    Unicode Name? y


                                    Redirect On OPTIM Failure: 5000

        SCCAN? n                              Digital Loss Group: 18
             Preferred Minimum Session Refresh Interval(sec): 900
```

Figure 9: Trunk-Group Form, p.2

3.1.5. Dial Plan

Use the **change dialplan analysis** command to configure the dial plan as shown in the following table.

Parameter	Usage
Dialed string: "0"	Use a "0" as Facilities Access Code (FAC) to access external telephone numbers.
Dialed string: "1"	Four digits numbers starting with "1" are for local extensions.
Dialed string: "*01"	The dialed string "*01" is the Trunk Access Code (TAC) shown in **Figure 8**.

Table 7: Dial Plan Analysis Parameters

```
change dialplan analysis                                          Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                                      Percent Full:    0

      Dialed   Total  Call    Dialed   Total  Call    Dialed   Total  Call
      String   Length Type    String   Length Type    String   Length Type
       0         1     fac
       1         4     ext
      *01        3     dac
```

Figure 10: Dialplan Analysis Form

Use the **change feature-access-codes** command to assign dialed digit strings to feature access
codes. Use a "0" as the leading digit of ARS numbers which provide access to the SIP trunk.

```
change feature-access-codes                                       Page   1 of   5
                            FEATURE ACCESS CODE (FAC)
           Abbreviated Dialing List1 Access Code:
           Abbreviated Dialing List2 Access Code:
           Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                   Announcement Access Code:
                  Answer Back Access Code:
                    Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
      Auto Route Selection (ARS) - Access Code 1: 0      Access Code 2:
               Automatic Callback Activation:          Deactivation:
Call Forwarding Activation Busy/DA:        All:         Deactivation:
   Call Forwarding Enhanced Status:         Act:        Deactivation:
                        Call Park Access Code:
                      Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                 CDR Account Code Access Code:
                       Change COR Access Code:
                  Change Coverage Access Code:
                    Contact Closure   Open Code:          Close Code:
```

Figure 11: Feature-Access-Codes Form

3.1.6. Outgoing Call Routing

Use the **change ars analysis** command to designate that all ("x") ars numbers beginning with
"0", with a minimum length of "7" digits and a maximum length of "20" digits, be routed via
route pattern "1" using public numbering format ("pubu").

```
change ars analysis 0                                             Page   1 of   2
                            ARS DIGIT ANALYSIS TABLE
                                 Location:  all         Percent Full:    0

        Dialed              Total     Route    Call   Node  ANI
        String          Min   Max   Pattern    Type   Num   Reqd
  x                      7    20      1         pubu         n
                                                             n
                                                             n
```

Figure 12: Ars Analysis Form

Use the **change route-pattern** command to designate that calls routed via routing pattern "1" be routed via trunk group "1", and that the "0" digit, which was removed by the "Auto Route Selection Access Code 1", shown in **Figure 11**, should be restored as the leading digit of the called party number so that it has the format "00"<country code><number>.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1  Pattern Name: SIP
                            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: 1    0        1               0                                    n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
     0 1 2 M 4 W    Request                                     Dgts Format
                                                              Subaddress
 1: y y y y y n  n             rest                                       none
 2: y y y y y n  n             rest                                       none
 3: y y y y y n  n             rest                                       none
 4: y y y y y n  n             rest                                       none
 5: y y y y y n  n             rest                                       none
 6: y y y y y n  n             rest                                       none
```

Figure 13: Route Pattern Form

Use the **change public-unknown-numbering** command to designate that the local FAX and the three locally attached Avaya IP Telephones each be assigned public telephone numbers, as shown in **Figure 1**.

```
change public-unknown-numbering 4                               Page   1 of   2
                    NUMBERING - PUBLIC/UNKNOWN FORMAT
                                        Total
Ext Ext            Trk      CPN         CPN
Len Code           Grp(s)   Prefix      Len
                                                   Total Administered: 5
  4  1001          1        00442071878335   14       Maximum Entries: 240
  4  1004          1        00442071878337   14
  4  1005          1        00442071878338   14
  4  1006          1        00442071878339   14
```

Figure 14: Public-Unknown-Numbering Form

3.1.7. Incoming Call Routing

Use the **change inc-call-handling-trmt trunk-group** command to map calls arriving from trunk group "1" from public numbering format to the extensions of the locally attached Avaya IP Telephones and the FAX shown in **Figure 1**.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of   3
                      INCOMING CALL HANDLING TREATMENT
 Service/        Called    Called      Del Insert
 Feature         Len       Number
 tie            14 00442071878335   14  1001
 tie            14 00442071878337   14  1004
 tie            14 00442071878338   14  1005
 tie            14 00442071878339   14  1006
```

Figure 15: Inc-Call-Handling-Trmt Trunk-Group Form

3.1.8. Configure Codec Sets

Use the **change ip-codec-set** command to designate a codec set to be used for communication
with the COLT SIP trunk. Testing was done with both the G.711A and G.729B codecs, using
the default of 2 frames per packet and a packet size of 20ms in both cases.

Parameter	Usage
Audio Codec (p. 1)	Enter "G.711A" or "G.729B" as the codec to be used to communication with the COLT SIP trunk.
FAX Mode (p. 2)	Enter "t.38-standard" to specify that the T.38 standard should be used to transmit FAX documents via the COLT SIP trunk.
TDD/TTY Mode (p. 2)	Enter "off".

Table 8: IP-Codec-Set Parameters

```
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.711A             n           2        20
 2:
```

Figure 16: IP-Codec-Set Form, p.1

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

                       IP Codec Set

                           Allow Direct-IP Multimedia? n



                 Mode              Redundancy
       FAX       t.38-standard         0
       Modem     off                   0
       TDD/TTY   off                   3
       Clear-channel  n                0
```

Figure 17: IP-Codec-Set Form, p.2

3.1.9. Configure IP Network Region

Use the **change ip-network-region** command to designate a network region to be used for the
COLT SIP trunk using the parameters shown in the following table.

Parameter	Usage
Region	Enter an unassigned network region number (this must be the same number as was assigned in **Figure 5**.
Location	Enter "1".
Authoritative Domain	Enter the domain name assigned in **Figure 22**.
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in **Figure 16**.

Table 9: IP-Network-Region Parameters

```
change ip-network-region 1                                    Page   1 of  19
                            IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: ffm.com
    Name: FFM
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: 3329
DIFFSERV/TOS PARAMETERS                       RTCP Reporting Enabled? y
 Call Control PHB Value: 46     RTCP MONITOR SERVER PARAMETERS
      Audio PHB Value: 46        Use Default Server Parameters? y
      Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
      Audio 802.1p Priority: 6
      Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                              RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
          Keep-Alive Count: 5
```

Figure 18: IP-Network-Region Form, p.2

3.1.10.   Configure Telephone Stations

Use the **add station** command using the parameters shown in the following table. Repeat this for each of the locally attached stations shown in **Figure 1**.

Parameter	Usage
Type	Enter the type identifier of local telephone.
Security Code	Enter the security code to be assigned to the station for security purposes.
Name	Enter a name to identify the station or its user.

Table 10: Station Parameters for IP Telephones

```
add station 1004                                            Page   1 of   5
                               STATION

Extension: 1004                       Lock Messages? n
     Type: 4610                      Security Code: ####          TN: 1
     Port: S00000              Coverage Path 1:                  COR: 1
     Name: ext 1004           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: 1004
          Speakerphone: 2-way         Mute Button Enabled? y
      Display Language: english
 Survivable GK Node Name:
        Survivable COR: internal       Media Complex Ext:
  Survivable Trunk Dest? y                    IP SoftPhone? n




                                   Customizable Labels? y
```

Figure 19: Station Form for IP Telephones

3.1.11.   Configure FAX Devices

Use the **add station** command using the parameters shown in the following table.

Parameter	Usage
Type	Enter the "2500" to assign an analog device.
Port	Enter the identifier for the analog port to which the FAX is attached. In this case "V7" is the slot number, and "02" is the port number.
Name	Enter a name to identify the FAX or its user.

Table 11: Station Parameters for FAX Device

```
add station 1001                                              Page   1 of   5
                                  STATION

Extension: 1001                          Lock Messages? n
    Type: 2500                           Security Code:                 TN: 1
    Port: V702                        Coverage Path 1:                  COR: 1
    Name: FAX                         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: 1004
            Speakerphone: 2-way          Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
           Survivable COR: internal         Media Complex Ext:
    Survivable Trunk Dest? y                     IP SoftPhone? n



                                      Customizable Labels? y
```

Figure 20: Station Form for FAX Device

3.2. Avaya SIP Enablement Services

Configure SES by entering "<SES IP Address>/admin/" in a web browser. After entering the administrator name and password, the following screen content is displayed:

Figure 21: SES "Top" Configuration Screen

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3.2.1. Server Configuration

Select "System Properties" from the "Server Configuration" menu from the left frame of the screen. Enter values in this screen as shown in the following table:

Parameter	Usage
SIP Domain	Enter the same value as was used in **Figure 18**.
License Host	Enter the IP address of the license host, in this case the IP address of the SES server.

Table 12: Parameters for System Properties

Figure 22: SES System Properties Screen

3.2.2. Add Hosts

Select "Hosts" → "Add Host" from the left frame of the top level screen shown in **Figure 21**. Enter values in this screen as shown in the following table, accepting the default values for those parameters which are not listed.

Parameter	Usage
Host IP Address	Enter the IP address of the SES server.
DB Password	Enter the password which was entered from the initial_setup script when SES was installed.
Profile Service Password	Enter the password which was entered from the initial_setup script when SES was installed.
Host Type	Select "home/edge".

Table 13: "Add Host" Parameters

Figure 23: SES Add Host Screen

Select the "Map" menu point from the "List Media Servers" screen, and then click the "Add Another Map" control from the following screen (not shown).
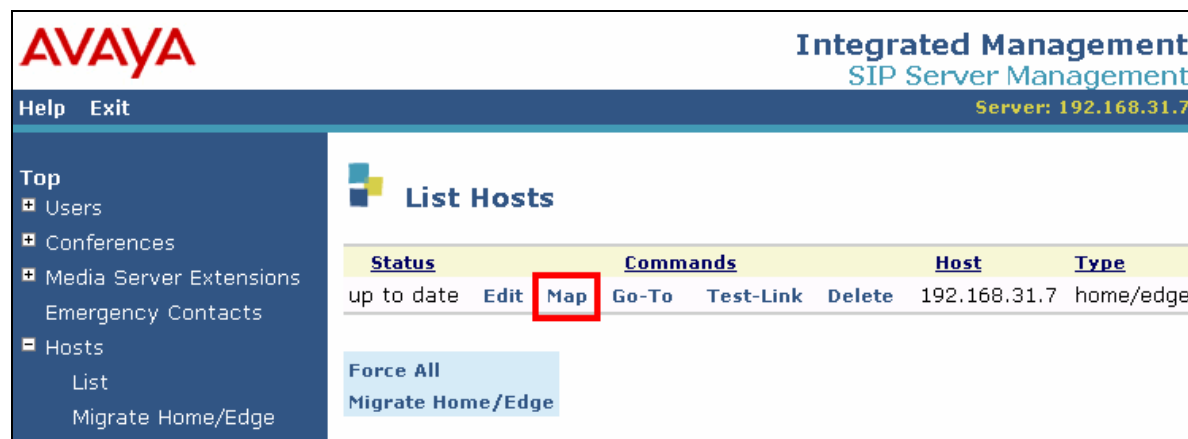
Figure 24: SES List Hosts Screen

Outbound calls are routed using Host Address Maps to select the destination host. Host Address Maps and Media Server Address Maps (see **Figure 30**) must be unique. This necessitates that the Host Address Maps be defined such that none of them conflict with the Media Server Address Map. Simply defining a host map with the value of "^sip:00*" to route PSTN calls to the COLT SIP network would cause external calls to local extensions to be routed back to the COLT SIP network. This can be avoided by defining the Host Address Maps shown in the following table. This causes calls to all numbers, except those which begin with "0044207187833" to be routed to the COLT SIP network. Thus, external calls to local extensions will not be rerouted to the COLT SIP network. The "National" map routes calls made to destinations within the national PSTN, i.e. those which have numbers beginning with a "0", followed by a city code and subscriber number.

Host Address Map Name	Host Address Map Pattern
PSTN-01	^sip:00[^4][0-9]*@
PSTN-02	^sip:004[^4][0-9]*@
PSTN-03	^sip:0044[^2][0-9]*@
PSTN-04	^sip:00442[^0][0-9]*@
PSTN-05	^sip:004420[^7][0-9]*@
PSTN-06	^sip:0044207[^1][0-9]*@
PSTN-07	^sip:00442071[^8][0-9]*@
PSTN-08	^sip:004420718[^7][0-9]*@
PSTN-09	^sip:0044207187[^8][0-9]*@
PSTN-10	^sip:00442071878[^3][0-9]*@
PSTN-11	^sip:004420718783[^3][0-9]*@
National	^sip:0[1-9][0-9]*@

Table 14: "Add Host Address Map" Parameters

The configuration for the first map in the above table is shown in the following figure. Repeat this procedure for each of the other entries in the table.

Figure 25: SES Add Host Address Map Screen

After all of the Host Address Maps have been configured, the content of the maps can be confirmed by selecting the "Address Map Priorities" menu item from the left frame of the top level screen.

Figure 26: SES Address Map Priorities

After the host address maps have been added, select "Add Another Contact" (not shown) and specify the contact parameters as shown in the following table.

Parameter	Usage
Contact	Enter "sip:$(user)@<SBC IP Address>:5060;transport=udp". The IP address of the SBC is shown in **Figure 1**. Note that the COLT VoIP Access SIP Service uses UDP as transport with port 5060, which is the standard UDP port used for SIP.

Table 15: "Add Host Contact" Parameters

Figure 27: SES Add Host Contact Screen

3.2.3. Add Media Server Interface

Select "Media Servers" → "Add" from the "Top" level menu shown in **Figure 21**, and specify the interface parameters as shown in the following table.

Parameter	Usage
Media Server Interface Name	Select a suitable name to identify this interface.
Host	Select the IP address of the SES server from the drop-down box.
SIP Trunk IP Address	Enter the IP address of the "procr" interface, as shown in **Figure 6**.

Table 16: "Add Media Server Interface" Parameters

Figure 28: SES Add Media Server Interface

Select the "Map" menu point from the "List Media Servers" screen, and then click the "Add Another Map" control from the following screen.

Figure 29: SES List Media Servers Screen

Enter the values shown in the following table in the "Add Media Server Address Map" screen.

Parameter	Usage
Name	Enter an appropriate name to identify the map.
Pattern	Enter "^sip:0044207187833[5-9]@ to specify that incoming numbers beginning with "0044207187833" followed by digit from "5" to "9" should be routed to the Avaya Communication Manager "procr" interface.

Table 17: "Add Media Server Address Map" Parameters
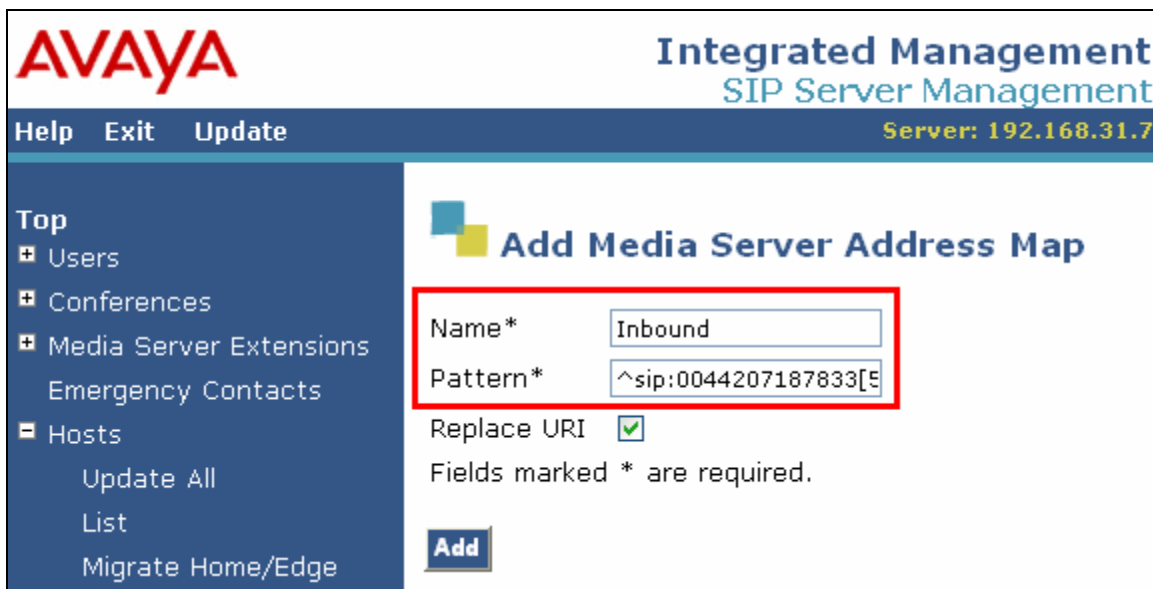
Figure 30: SES Add Media Server Address Map

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After the Media Server address map has been added, select "Add Another Contact" and specify the contact parameters as shown in the following table.

Parameter	Usage
Contact	Enter "sip:$(user)@<CM interface IP address>:5061;transport=tls". Note that the SIP connection to Avaya Communication Manager uses TLS as transport with port 5061, which is the standard TSL SIP port.

Table 18: "Add Host Contact" Parameters

Figure 31: SES Media Server Contact Screen

MRR; Reviewed:
SPOC 3/18/2008

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3.2.4. Configure Trusted Host

Select "Trusted Hosts" → "Add" from the "Top" level menu shown in **Figure 21**, and specify the parameters as shown in the following table.

Parameter	Usage
IP Address	Enter the IP address on the SBC which is allocated to SIP communications, as shown in **Figure 1**.
Host	Select the IP address of the SES server from the drop-down box.
Comment	Enter an appropriate name to identify the COLT VoIP Access SIP Service .

Table 19: "Add Trusted Host" Parameters

Figure 32: SES Trusted Host Screen

3.3. COLT 2611 Router

The 2611 router was configured with the instructions shown below. Note that although public IP addresses are not shown, for security reasons, the addresses shown below do correspond to those shown in **Figure 1**.

```
interface Tunnel0
 ip address 1.1.1.2 255.255.255.252
 tunnel source XX.XX.XX.50
 tunnel destination YY.YY.YY.54
 no shut
!
interface FastEthernet0/0
 description Link to Internet
 ip address XX.XX.XX.50 255.255.255.252
 duplex auto
 speed auto
 no shut
!
interface FastEthernet0/1
 description Link to Avaya
 ip address 192.168.31.1 255.255.255.0
 duplex auto
 speed auto
 no shut
!
ip classless
ip route 0.0.0.0 0.0.0.0 XX.XX.XX.49
ip route 10.0.0.0 255.255.255.0 Tunnel0
ip http server
ip pim bidir-enable
!
line con 0
line aux 0
line vty 0 4
 login
 password
!
end
```

Figure 33: Router Settings

4. Verification Steps

The correct configuration of the system can be verified by performing the following steps:

- Verify that the local Avaya IP Telephones can call each other.
- Verify that the Avaya S8300 Server and SES server can ping each other and the default gateway address of the 2611 router.
- Verify that the Avaya S8300 Server can ping the SBC port allocated to Avaya Communication Manager.
- Verify that locally attached Avaya IP Telephones and the telephones attached to the PSTN can call each other.
- Use the "status station" command from the SAT terminal to verify that calls between locally attached telephones and telephones attached to the PSTN are connected with one another without traversing the Avaya G350 Media Gateway.

▪ Verify that it is possible to send FAX messages between the locally attached FAX device and that which is attached to the PSTN.

5. Test Results

Long documents (25 pages) sent from the FAX attached to the Avaya G350 Media Gateway to the FAX attached to the PSTN exhibited sporadic pixel row dropouts. This problem is currently under investigation. Otherwise, all tests which were conducted were performed successfully.

6. Conclusion

These Application Notes contain instructions for configuring Avaya Communication Manager and Avaya SIP Enablement Services to connect to the COLT SIP network. A list of instructions is provided to enable the user to verify that the various components have been correctly configured.

7. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *Administrator Guide for Avaya Communication Manager*, February 2007, Issue 3, Document Number 03-300509.
[2] *Feature Description and Implementation for Avaya Communication Manager,* February 2007, Issue 5, Document Number 555-245-205.
[3] *Installing and Administering SIP Enablement Services,* May 2007, Issue 1.5, Document Number 03-600768.
[4] *Installing and Configuring the Avaya S8500 Media Server*, February 2007, Issue 6, Document Number 03-300143.
[5] *SIP Enablement Services (SES) Implementation Guide*, May 2007, Document Number 16-300140.
[6] *4600 Series IP Telephone R2.4 LAN Administrator Guide,* April 2006, Document Number 555-233-507.

Appendix A: Sample SIP INVITE Messages

These traces were made at a port which mirrored the connection between the Avaya G350 Media Server and the COLT 2611 router.

Incoming call:

```
Request-Line: INVITE sip:00442071878337@192.168.31.7:5060;user=phone SIP/2.0
    Message Header
        Via: SIP/2.0/UDP 10.0.0.10:5060;branch=z9hG4bKr6o37a1068jg4boph2g1.1
        From: 00496975050 <sip:00496975050@va-test.sip.colt.net:5060;user=phone>;tag=SDlqti301-
1075768590
        To: +442071878337 <sip:00442071878337@192.168.31.7:5060;user=phone>
        Call-ID: SDlqti301-eb02c0328bbf333c4735d2253a9cd208-v300g00
        CSeq: 1 INVITE
        Max-Forwards: 69
        Supported: timer
        Session-Expires: 1800
        Min-SE: 1800
        Contact: 00496975050 <sip:00496975050@10.0.0.10:5060;transport=udp>
        Allow: INVITE,ACK,PRACK,SUBSCRIBE,BYE,CANCEL,NOTIFY,INFO,REFER,UPDATE
        Content-Type: application/sdp
        Content-Length: 399
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 352 0 IN IP4 10.0.0.10
            Session Name (s): Cisco SDP 0
            Connection Information (c): IN IP4 10.0.0.10
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 20322 RTP/AVP 18 8 99 100
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): fmtp:18 annexb=no
            Media Attribute (a): rtpmap:99 telephone-event/8000
            Media Attribute (a): fmtp:99 0-15
            Media Attribute (a): rtpmap:100 X-NSE/8000
            Media Attribute (a): fmtp:100 192-194,200-202
            Media Attribute (a): X-sqn:0
            Media Attribute (a): X-cap: 1 audio RTP/AVP 100
            Media Attribute (a): X-cpar: a=rtpmap:100 X-NSE/8000
            Media Attribute (a): X-cpar: a=fmtp:100 192-194,200-202
            Media Attribute (a): X-cap: 2 image udptl t38
```

Outgoing call:

```
    Request-Line: INVITE sip:00496975056174@10.0.0.10:5060;transport=udp SIP/2.0
    Message Header
        Call-ID: 8010747b4afadc11f147eb2eab00
        CSeq: 1 INVITE
        From: "ext 1004" <sip:00442071878337@ffm.com:5061>;tag=8010747b4afadc11e147eb2eab00
        Record-Route: <sip:192.168.31.7:5060;lr>,<sip:192.168.31.3:5061;lr;transport=tls>
        To: "00496975056174" <sip:00496975056174@192.168.31.7>
        Via: SIP/2.0/UDP 192.168.31.7:5060;branch=z9hG4bK83838303030313131345bc.0,SIP/2.0/TLS
192.168.31.3;psrrposn=2;received=192.168.31.3;branch=z9hG4bK8010747b4afadc120147eb2eab00
        Content-Length: 154
        Content-Type: application/sdp
        Contact: "ext 1004" <sip:00442071878337@192.168.31.3:5061;transport=tls>
        Max-Forwards: 61
        User-Agent: Avaya CM/R014x.00.1.731.2
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
        History-Info: <sip:00496975056174@192.168.31.7>;index=1
        History-Info: "00496975056174" <sip:00496975056174@192.168.31.7>;index=1.1
        Supported: 100rel,timer,replaces,join,histinfo
        Min-SE: 1800
        Session-Expires: 1800;refresher=uac
        P-Asserted-Identity: "ext 1004" <sip:00442071878337@ffm.com:5061>
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1 1 IN IP4 192.168.31.3
            Session Name (s): -
            Connection Information (c): IN IP4 192.168.31.2
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 2052 RTP/AVP 8 127
            Media Attribute (a): rtpmap:8 PCMA/8000
            Media Attribute (a): rtpmap:127 telephone-event/8000
```

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