



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the COLT VoIP Access SIP Service and an Avaya Communication Manager 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 Avaya Communication Manager. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various 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.

Table of Contents

1. Introduction.....	3
1.1. System Configuration	4
1.2. Call Routing.....	5
1.2.1. Incoming Calls	5
1.2.2. Outgoing Calls	5
2. Equipment and Software Validated	6
3. Configuration	6
3.1. Avaya Communication Manager	6
3.1.1. Verify system-parameters customer-options	6
3.1.2. Set system-parameters features.....	8
3.1.3. Configure Media Gateway	9
3.1.4. SIP Interface to SIP Enablement Services	9
3.1.5. Dial Plan.....	12
3.1.6. Outgoing Call Routing.....	13
3.1.7. Incoming Call Routing.....	15
3.1.8. Configure Codec Sets	15
3.1.9. Configure IP Network Region	16
3.1.10. Configure Telephone Stations.....	17
3.1.11. Configure FAX Devices	20
3.2. Avaya SIP Enablement Services.....	21
3.2.1. Add Media Server Interface.....	29
3.2.2. Add Hosts.....	32
3.2.3. Add Trusted Host.....	37
4. Verification Steps.....	37
5. Test Results.....	38
6. Conclusion	38
7. Additional References.....	38
Appendix A: Sample SIP INVITE Messages	39

1. Introduction

These Application Notes describe the procedure for configuring Session Initiation Protocol (SIP) trunking between the COLT VoIP Access SIP trunking network and Avaya SIP telephony solution consisting of Avaya SIP Enablement Services (SES), Avaya Communication Manager, and Avaya IP telephones using the H.323 protocol as endpoints.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [7] is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by COLT. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

1.1. System Configuration

The following diagram illustrates the configuration which used for testing.

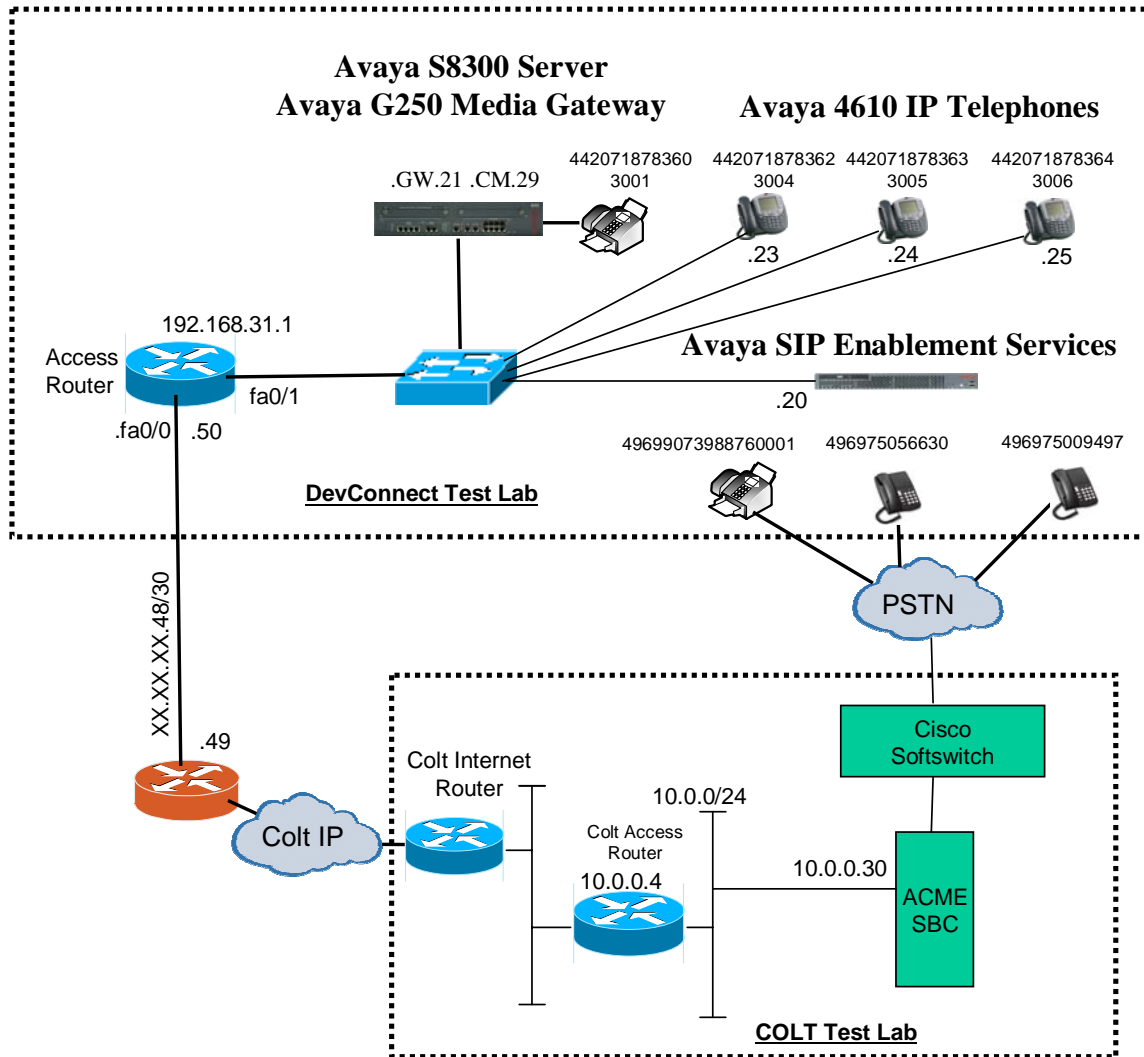


Figure 1: System Configuration

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 G250 Media Gateway. The Avaya SIP Enablement Services (SES) server runs on the Avaya S8300 Server together with Avaya Communication Manager. SES serves as a registrar for local SIP endpoints and 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 machines 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 through the Avaya G250 Media Gateway. Avaya Communication Manager and the COLT SIP network are configured to support T.38 fax transmission.

1.2. Call Routing

The telephone numbers received from the COLT VoIP Access SIP Service have the format 00<country code><national number> for both the called and the calling party, including numbers for those calls which originate in the local dialing area. The same number sequence can be used for dialing outgoing calls. Alternatively, calls to national destinations can be made by dialing 0<national number>.

1.2.1. Incoming Calls

An incoming call from “00496975056630” to “00442071878364” will be routed as follows:

- “00442071878364” will be matched by the Media Server Address Map shown in **Table 16** and be routed to Avaya Communication Manager SIP trunk by the associated Media Server Contact described in **Table 17**.
- The “00442071878364” entry in the “inc-call-handling-trmt” form shown in **Figure 16**, causes the number to be converted to “3005” and routed to the local telephone with that extension.
- Extension will ring showing the caller as “00496975056630”.

1.2.2. Outgoing Calls

An incoming call dialed from extension “3006” to “00496975056630” is routed as follows:

- “00496975056630” is matched by “Auto Route Selection Access Code 1” shown in **Figure 12**, which removed the leading “0” from the number yielding “0496975056630”.
- The “x” entry in the “ars analysis” form shown in **Figure 13** selects Route Pattern “1”.
- “route-pattern 1” shown in **Figure 14** inserts the leading digit “0” yielding “00496975056630”, and routes the call to trunk group 1. The Calling Party Number is changed to “442071878364” by the Public Unknown Numbering form shown in **Figure 15**.
- The call to “00496975056630” is sent via Trunk Group 1 to the SES server, which is matched by the Host Map “PSTN-02” (“^sip:004[^4][0-9]*@”) shown in **Table 19**.

- The Host Contact shown in **Figure 41** is used to route the call to the Session Border Controller.

2. Equipment and Software Validated

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

Equipment	Software
Avaya Communication Manager	5.0 (R015x.00.0.825.4 with patch 00.0.825.4-1310)
Avaya G250 Media Gateway	27.27.0
Avaya SIP Enablement Services	5.0 (SES-5.0.0.0-825.30)
Avaya 4610 IP Telephones	2.8.3
ACME Session Border Controller	4.1.4 p24
Cisco Soft Switch	PGW 2200 version 9.7.3
Cisco Voice Gateway AS5400	12.4(9)

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".
Extended Cvg/Fwd Admin? (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: 0                               0
    Maximum Concurrently Registered IP Stations: 12                   0
    Maximum Administered Remote Office Trunks: 0                       0
Maximum Concurrently Registered Remote Office Stations: 0             0
    Maximum Concurrently Registered IP eCons: 0                         0
    Max Concur Registered Unauthenticated H.323 Stations: 0            0
    Maximum Video Capable H.323 Stations: 0                            0
    Maximum Video Capable IP Softphones: 0                             0
    Maximum Administered SIP Trunks: 10                               0
    Maximum Administered Ad-hoc Video Conferencing Ports: 0            0
    Maximum Number of DS1 Boards with Echo Cancellation: 0            0
    Maximum TN2501 VAL Boards: 10                                       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

```

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? n                                           ISDN Feature Plus? n
    Enhanced EC500? y                                               ISDN/SIP 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? n
  External Device Alarm Admin? n                                       Media Encryption Over IP? n
Five Port Networks Max Per MCC? n                                     Mode Code for Centralized Voice Mail? n
  Flexible Billing? n
  Forced Entry of Account Codes? n                                     Multifrequency Signaling? y
  Global Call Classification? n                                       Multimedia Call Handling (Basic)? n
  Hospitality (Basic)? y                                             Multimedia Call Handling (Enhanced)? n
  Hospitality (G3V3 Enhancements)? n                                   Multimedia IP SIP Trunking? n
    IP Trunks? y

IP Attendant Consoles? n

```

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: g250                                           FW Version/HW Vintage:
      Name: G250                                           MGP 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:

                                Max Survivable IP Ext: 8

V9:
  
```

Figure 5: Media Gateway Form

3.1.4. SIP Interface to SIP Enablement Services

Use the **change node-names ip** command to configure to assign the name "ses" to the IP address of the SES server, which is co-resident with Avaya Communication Manager. Thus, it is assigned the same address as "procr", the local processor. "ses" is used in the SIP signaling group, as shown in **Figure 7**.

```

change node-names ip                                     Page 1 of 2
                                IP NODE NAMES
      Name          IP Address
default           0.0.0.0
procr             192.168.31.29
ses              192.168.31.20
  
```

Figure 6: Node-Names Ip Form

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

Parameter	Usage
Group Type	Enter "sip".
Near-end Node Name	Enter "procr" to designate the G250 processor as the near end node name.
Far-end Node Name	Enter "ses" to assign the SES server as the far end node name.
Near-end Listen Port	Assign this parameter to the value used in for the SIP Trunk Port in Figure 33 .
DTMF over IP	Enter "rtp-payload". This value used to have Avaya Communication Manager send DTMF transmissions using RFC 2833.
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: 1
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, where <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 shown 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 11 .
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 name via the SIP trunk.
Preferred Minimum Session Refresh Interval (p.2)	Enter "900" seconds, as required by the COLT SIP trunk interface. This should be half of the Session Refresh Interval which is configured for the COLT VoIP Access SIP Service.
Send Transferring Party Information (p.4)	Enter "y".

Table 6: Trunk-Group 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: *03
  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

```

change trunk-group 1
                                PROTOCOL VARIATIONS
                                Mark Users as Phone? n
                                Prepend '+' to Calling Number? n
                                Send Transferring Party Information? y
                                Telephone Event Payload Type: 127

```

Figure 10: Trunk-Group Form, p.4

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: "3"	Four digit numbers starting with "3" are for local extensions.
Dialed string: "*03"	The dialed string "*03" is the Trunk Access Code (TAC) shown in Figure 8 .

Table 7: Dial Plan Analysis Parameters

```

change dialplan analysis
                                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    0       1     fac    0       1     fac
                                3       4     ext    3       4     ext    3       4     ext
                                *03    3     dac    *03    3     dac    *03    3     dac

```

Figure 11: Dialplan Analysis Form

3.1.6. Outgoing Call Routing

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. Although this causes the leading “0” to be removed from the called party number, the “0” specified for the “Inserted Digits” parameter in the routing pattern (see **Figure 14**) restores it.

```

change feature-access-codes                                     Page 1 of 6
                                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 12: Feature-Access-Codes Form

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
  
```

Figure 13: Ars Analysis Form

Use the **change route-pattern** command to designate that calls routed via route 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 12**, 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
                        Secure SIP? n
      Grp FRL NPA Pfx Hop Toll No.   Inserted           DCS/ IXC
      No   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 14: 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
      Ext Ext      Trk      CPN      Total
      Len Code     Grp(s)   Prefix   CPN
                                         Len
                                         Total Administered: 0
                                         Maximum Entries: 240
4  3001           1           00442071878360 14
4  3004           1           00442071878362 14
4  3005           1           00442071878363 14
4  3006           1           00442071878364 14

```

Figure 15: 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 show 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 00442071878360    14 3001
tie           14 00442071878362    14 3004
tie           14 00442071878363    14 3005
tie           14 00442071878364    14 3006
  
```

Figure 16: 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” and “G.729A” as the codec to be used to communication with the COLT SIP trunk. The COLT network supports G729 and G729A but G729B is not supported.
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 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: G.729A      n                2          20

```

Figure 17: 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 18: Ip-Codec-Set Form, p.2

3.1.9. Configure IP Network Region

Use the **change ip-network-region <x>** command to designate a network region to be used for the COLT SIP trunk using the parameters shown in the following table, where <x> is the network region assigned to the procr IP interface. In this case “1” is used, as the procr IP interface is assigned to a default network region of “1”.

Parameter	Usage
Location	Enter “1”.
Authoritative Domain	Enter the domain name assigned in Figure 32
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in Figure 17 .

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 19: Ip-Network-Region Form, p.2

3.1.10. Configure Telephone Stations

Use the **add station** command to 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 3004                                     Page 1 of 5
                                                    STATION
Extension: 3004                                     Lock Messages? n
  Type: 4610                                       Security Code: ####          TN: 1
  Port: S00000                                    Coverage Path 1:           COR: 1
  Name: ext 3004                                  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: 3004
  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 IP Telephones

Use the **change cor 1** command to allow local stations to make external calls by setting “calling party restriction” to “none”. This Class of Restriction is assigned to the stations which have access to the COLT SIP network, as shown in **Figure 20**.

Parameter	Usage
Calling Party Restriction	Enter “none” to allow local stations to make external calls.

Table 11: Class of Restriction Parameters

```

change cor 1
                                     Page 1 of 23
                                     CLASS OF RESTRICTION

COR Number: 1
COR Description:

FRL: 0                               APLT? y
Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n         Called Party Restriction: none
Partitioned Group Number: 1          Forced Entry of Account Codes? n
Priority Queuing? n                   Direct Agent Calling? n
Restriction Override: none           Facility Access Trunk Test? n
Restricted Call List? n              Can Change Coverage? n

Access to MCT? y                      Fully Restricted Service? n
Group II Category For MFC: 7
Send ANI for MFE? n
MF ANI Prefix:                        Automatic Charge Display? n
Hear System Music on Hold? y         PASTE (Display PBX Data on Phone)? n
Can Be Picked Up By Directed Call Pickup? n
Can Use Directed Call Pickup? n
Group Controlled Restriction: inactive

```

Figure 21: Class of Restriction Form

Use the **change cos** command with the parameters shown in the following table for service class “1”, which is assigned to the stations which forward calls via the SIP trunk. This Class of Service is assigned to the stations which have access to the COLT SIP network, as shown in **Figure 20**.

Parameter	Usage
Restrict Call Fwd-Off Net	Enter “n” to allow calls to be forwarded via the SIP trunk.

Table 12: Class of Service Parameters

change cos		Page 1 of 2														
CLASS OF SERVICE																
	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Auto Callback	n	y	y	n	y	n	y	n	y	n	y	n	y	n	y	n
Call Fwd-All Calls	n	y	n	y	y	n	n	y	y	n	n	y	y	n	n	y
Data Privacy	n	y	n	n	n	y	y	y	y	n	n	n	n	y	y	y
Priority Calling	n	y	n	n	n	n	n	n	n	y	y	y	y	y	y	y
Console Permissions	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Off-hook Alert	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Client Room	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Restrict Call Fwd-Off Net	y	n	y	y	y	y	y	y	y	y	y	y	y	y	y	y
Call Forwarding Busy/DA	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Personal Station Access (PSA)	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding All	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Extended Forwarding B/DA	n	y	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Trk-to-Trk Transfer Override	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
QSIG Call Offer Originations	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n
Contact Closure Activation	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n	n

Figure 22: Class of Service Form

3.1.11. Configure FAX Devices

Use the **add station** command to 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 13: Station Parameters for FAX Device

```

add station 3001                                     Page 1 of 5
                                                    STATION
Extension: 3001                                     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 23: 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. Select “Launch Administration Web Interface”.

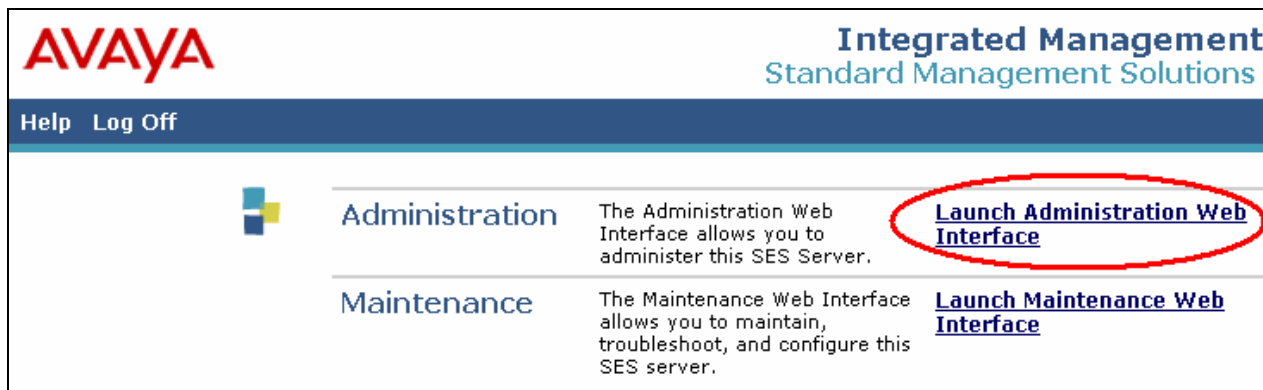


Figure 24: Launch Maintenance Web Interface Screen

From the “Top” menu click “Server Configuration” -> “License”.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top left features the Avaya logo. The top right displays the title "Integrated Management SIP Server Management" and the server IP address "Server: 192.168.31.20". Below the title bar, there are "Help" and "Exit" links. The main interface is divided into a left sidebar menu and a main content area. The sidebar menu includes "Top", "Setup", "Users", "Address Map Priorities", "Adjunct Systems", "Conferences", "Emergency Contacts", "Export/Import to ProVision", "Hosts", "IM logs", "Media Servers", "Media Server Extensions", "Server Configuration" (with sub-items "Admin Setup", "IM Log Settings", "License" (circled in red), and "System Properties"), "SIP Phone Settings", and "Survivable Call Processors". The main content area is titled "Top" and contains a list of management tasks:

Task Name	Description
Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Media Servers	Add and delete Media Servers.
Manage Media Server Extensions	Add and delete Media Server Extensions.

Figure 25: Select License from Top SES Screen

Click “Access WebLM”.

The screenshot shows a web interface with a blue sidebar on the left and a main content area on the right. The sidebar contains a 'Top' section with a list of menu items: Setup, Users, Address Map Priorities, Adjunct Systems, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, IM logs, Media Servers, Media Server Extensions, and Server Configuration. Under 'Server Configuration', there are sub-items: Admin Setup, IM Log Settings, and License. The main content area is titled 'List Licenses' and features a table with three columns: 'Proxy Name', 'Name', and 'Message'. The table contains three rows of data. A red circle highlights the text 'Access WebLM' located to the left of the table, between the 'Export/Import to ProVision' and 'Hosts' menu items in the sidebar.

	<u>Proxy Name</u>	<u>Name</u>	<u>Message</u>
Show	sipserver	Basic Proxy	
Show	sipserver	Edge Proxy	
Show	sipserver	Home Seats	

Figure 26: Select WebLM from License Screen

Log in to WebLM with the “admin” user name and corresponding password. Read the license and file when instructed by WebLM. Exit WebLM. Re-enter the URL “http:<S8300 IP address>/admin” into the Web browser.

AVAYA Web License Manager (WebLM)

You are here: Login

Login

User Name: admin

Password: ●●●●●●●●

Login

Figure 27: WebLM Login Screen

Click “Setup” on the Top SES menu.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. At the top left is the Avaya logo. At the top right, it says "Integrated Management SIP Server Management" and "Server: 192.168.31.20". Below the logo is a "Help Exit" link. On the left is a dark blue sidebar menu with "Top" and "Setup" (circled in red) as the main categories. Below "Top" are various sub-items like "Users", "Adjunct Systems", etc. The main content area on the right has a "Top" header and a list of management tasks with descriptions.

Top	
Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Media Servers	Add and delete Media Servers.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Server Configuration	View Properties of the system.
Manage SIP Phone Settings	Add/Delete Phone Settings
Manage Survivable Call Processors	Add and delete Survivable Call Processors.
System Status	View System Status.
Trace Logger	Manage SIP Trace Logs.
Manage Trusted Hosts	Add and delete Trusted Hosts.

Figure 28: Initiate “Setup” from Top SES Configuration Screen

Click “Setup Dataservice”.

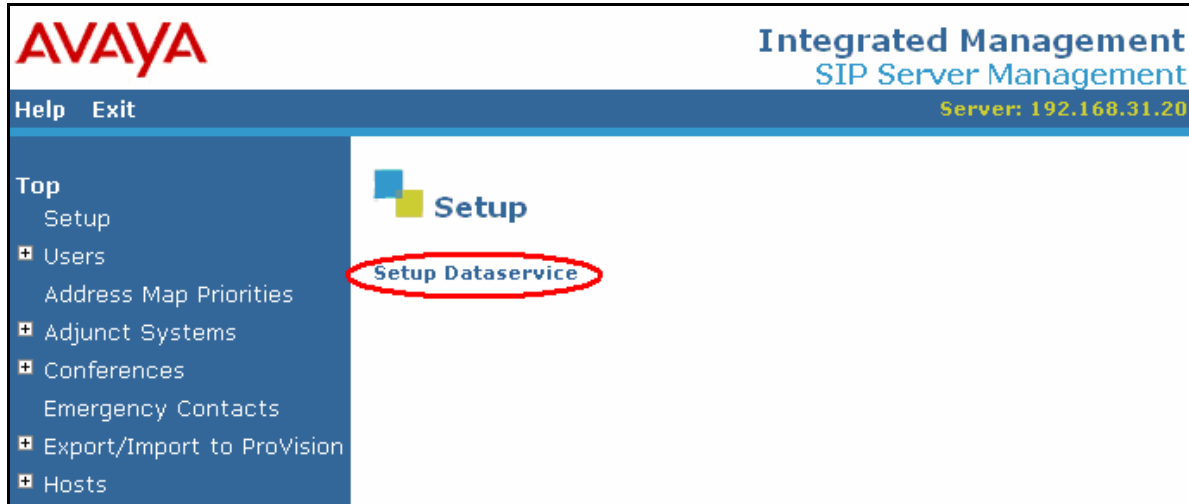


Figure 29: Initiate “Setup Dataservice” from Top Setup Screen

Select “This server is the SES Master Administration System for the SES Network”, and click “Setup”, and “Continue” for the screen that follows.

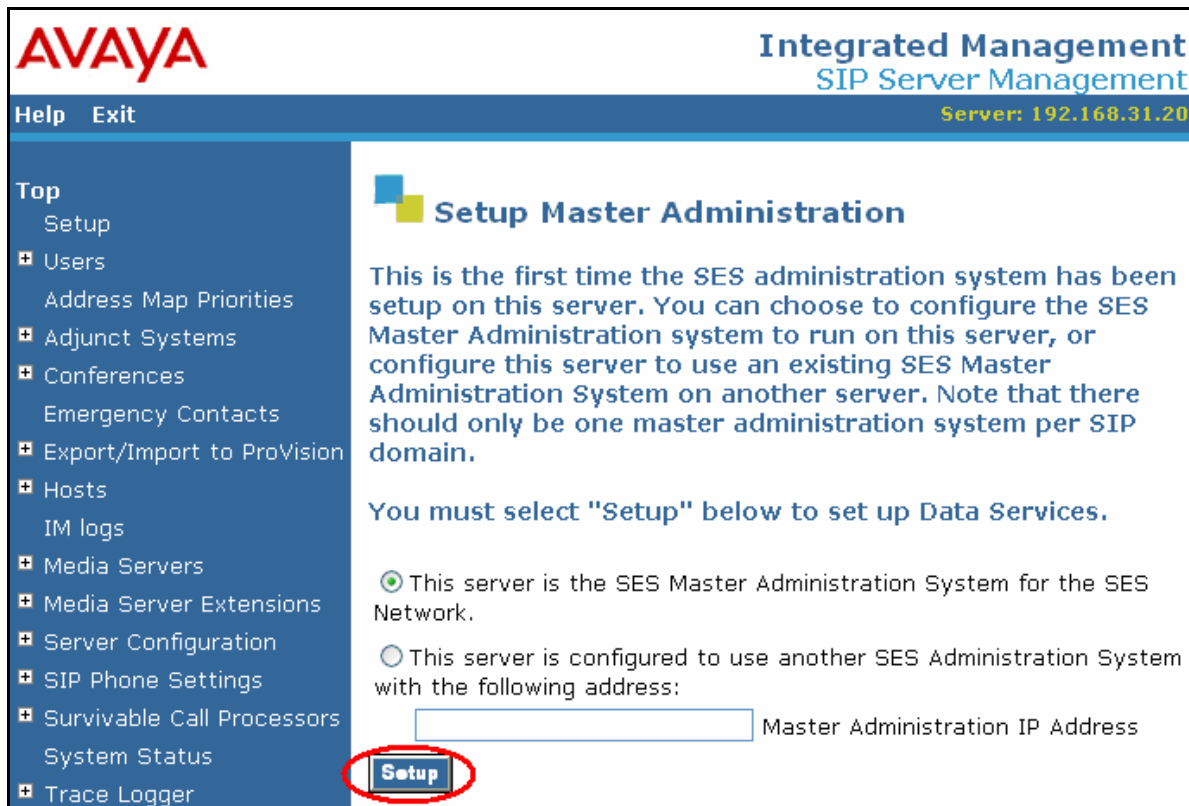


Figure 30: “Setup Master Administration” Screen

Click “Setup SIP domain”.



Figure 31: “Setup SIP Domain” Screen

Enter values in this screen as shown in the following table, and click “Update”, followed by “Ok” for the following screen.

Parameter	Usage
SIP Domain	Enter the same value as was used for “Authoritative Domain” in Figure 19 .
License Host	Enter the IP address of the license host, in this case the IP address of the SES server.

Table 14: Parameters for System Properties

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo and the text "Integrated Management SIP Server Management" with the server IP "192.168.31.20". A navigation menu on the left lists various system settings. The main content area is titled "View System Properties" and displays the following configuration details:

- SES Version: SES-5.0.0.0-825.30
- System Configuration: simplex
- Host Type: CM combined home-edge
- SIP Domain*: (highlighted with a red box)

Below the SIP Domain field, there is a note: "Note that the DNS domain is unknown". A detailed instruction follows: "If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com".

At the bottom of the configuration list, the SIP License Host* field is also highlighted with a red box, containing the value "192.168.31.20".

Figure 32: System Properties Screen

3.2.1. Add Media Server Interface

Select “Media Servers” → “Add” from the “Top” level menu shown in **Figure 25**, and specify the interface parameters as shown in the following table.

Parameter	Usage
Media Server Interface Name	Enter a descriptive name for this interface.
SIP Trunk IP Address	Enter the IP address of the S8300 Server, or the address of the C-LAN interface if an G650 gateway is used.
Media Server Admin Address	Enter the IP address of the S8300 Server administration interface.
Media Server Admin Login	Enter an administrator user ID for the media server.
Media Server Admin Password	Enter the password for the above user.

Table 15: “Add Media Server Interface” Parameters

The screenshot shows the 'Edit Media Server Interface' configuration page in the Avaya Integrated Management SIP Server Management interface. The page title is 'AVAYA Integrated Management SIP Server Management' with the server IP '192.168.31.20'. The left navigation menu includes 'Top', 'Users', 'Address Map Priorities', 'Adjunct Systems', 'Certificate Management', 'Conferences', 'Emergency Contacts', 'Export/Import to ProVision', 'Hosts', 'IM logs', 'Media Servers' (with sub-items 'Add' and 'List'), 'Media Server Extensions', 'Server Configuration', 'SIP Phone Settings', 'Survivable Call Processors', and 'System Status'. The main configuration area is titled 'Edit Media Server Interface' and contains the following fields:

- Media Server Interface Name***: G250
- Host**: 192.168.31.20
- SIP Trunk Link Type**: TCP TLS
- SIP Trunk IP Address***: 192.168.31.29
- Media Server Admin Address (see Help)**: 192.168.31.29
- Media Server Admin Port**: 5023
- Media Server Admin Login**: admin
- Media Server Admin Password**: [masked]
- Media Server Admin Password Confirm**: [masked]
- SMS Connection Type**: SSH Telnet Not Available

Figure 33: SES Add Media Server Interface Screen

Select the “Map” menu point from the “List Media Servers” screen, and then click the “Add Map In New Group” control from the following screen.

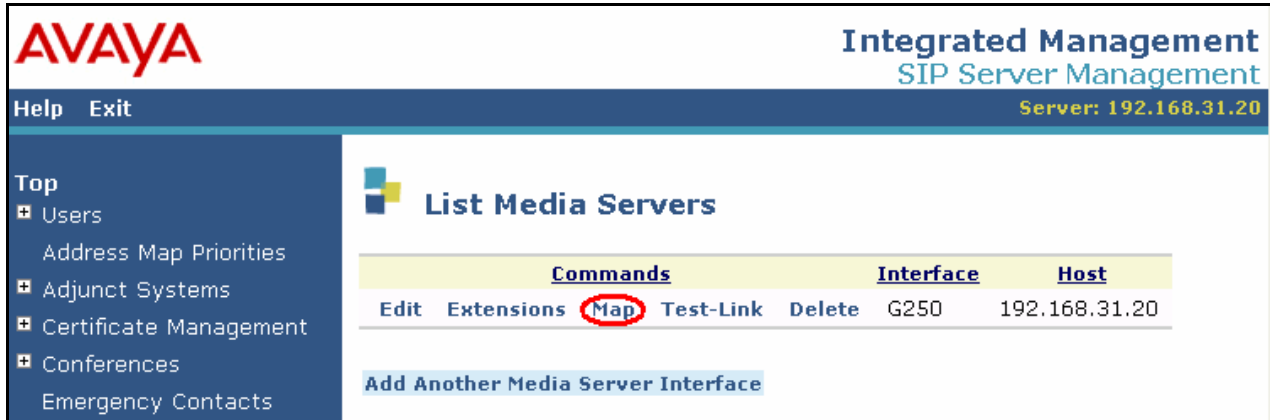


Figure 34: SES List Media Servers Screen

Media Server Address Maps are used to route incoming calls to Avaya Communication Manager. 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:0044207187836[0234]@" to specify that incoming numbers beginning with “0044207187836” followed by a digit “0”, “2”, “3”, or “4” should be routed to the Avaya Communication Manager “.

Table 16: “Add Media Server Address Map” Parameters

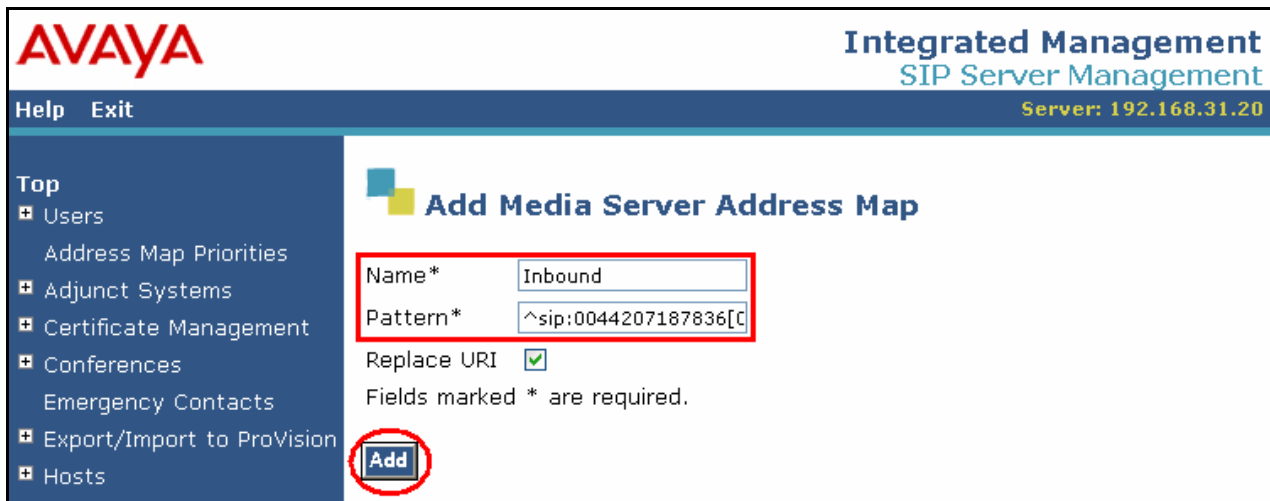


Figure 35: SES Add Media Server Address Map

After the Media Server address map has been added, the following contact is added automatically:

Parameter	Usage
Contact	Enter “sip:\$(user)@<CM interface IP address>:5061;transport=tls”.

Table 17: “Media Server Contact” Parameters

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The header includes the Avaya logo and the text "Integrated Management SIP Server Management" with the server IP "192.168.31.20". The navigation menu on the left lists various options like Users, Address Map Priorities, Adjunct Systems, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, and IM logs. The main content area is titled "List Media Server Address Map" and contains a table with the following data:

Commands	Name	Commands	Contact
Edit Delete	Inbound	Edit Delete	sip:\$(user) @192.168.31.29:5061;transport=tls
Add Another Map		Add Another Contact	
			Delete Group

Below the table, there is a button labeled "Add Map In New Group".

Figure 36: SES Media Server Contact Screen

3.2.2. Add Hosts

Select “Hosts” → “Add Host” from the left frame of the top level screen shown in **Figure 25**. Enter values in this screen as shown in the following table, accepting the default values for those parameters which are not listed. Click the “Add” button upon completion and the “Continue” button when the following screen is displayed.

Parameter	Usage
Host IP Address	Enter the IP address of the SES server.
Profile Service Password	Enter the password which was entered from the initial setup script when SES was installed.

Table 18: “Add Host” Parameters

Figure 37: SES Add Host Screen

Select the “Map” menu point from the “List Hosts” screen, and then click the “Add Another Map” control from the following screen.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The header includes the Avaya logo, the title 'Integrated Management SIP Server Management', and the server IP address '192.168.31.20'. A navigation menu on the left lists various management options. The main content area is titled 'List Hosts' and contains a table with the following data:

Commands		Host	Type	SES Version
Edit	Map To	192.168.31.20	SES combined home-edge	SES-5.0.0.0-825.31

Below the table, there is a button labeled 'Migrate Home/Edge'.

Figure 38: 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 35**) must be unique. This necessitates that the Host Address Maps be defined such that none of them conflict with the Media Server Address Map defined in **Figure 35**. 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, as this would conflict with the Media Server Address Map shown in **Figure 35**. 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 “0044207187836” to be routed to the COLT SIP network. Thus, external incoming 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 via the COLT SIP network, 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[^6][0-9]*@
National	^sip:0[1-9][0-9]*@

Table 19: “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.

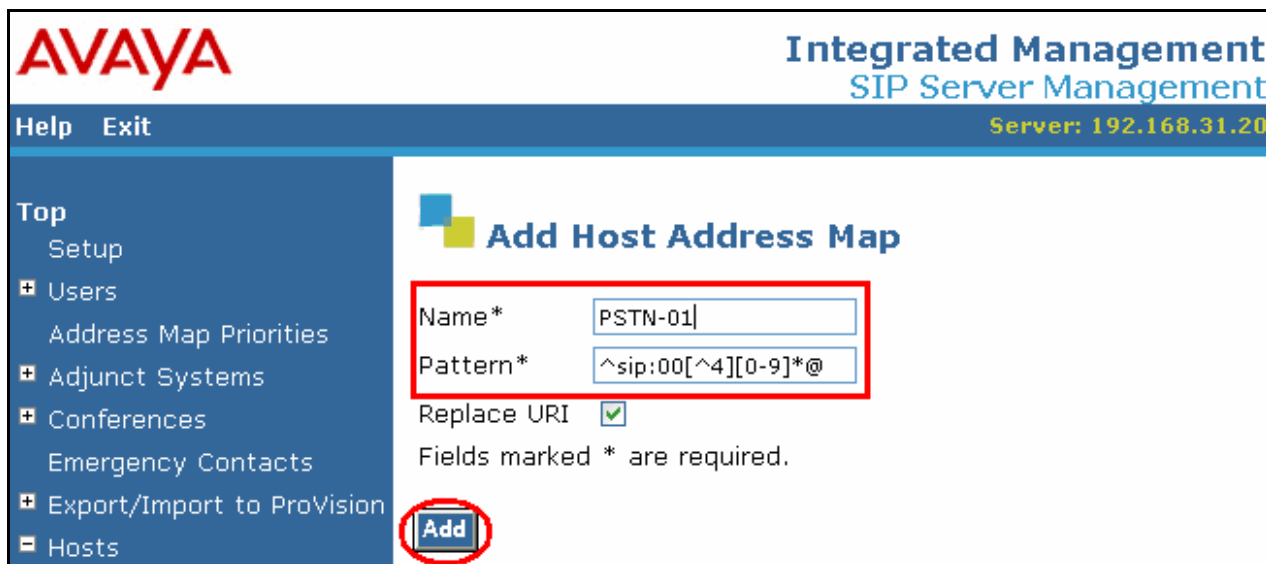


Figure 39: SES Add Host Address Map Screen

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.

Map Handle	Pattern	Map Type	Map Owner	Host	Priority* Highest Priority
Inbound	^sip:0044207187836[0234]@	media server	G250	192.168.31.20	1
NationalCall	^sip:0[1-9][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-01	^sip:00[^4][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-02	^sip:004[^4][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-03	^sip:0044[^2][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-04	^sip:00442[^0][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-05	^sip:004420[^7][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-06	^sip:0044207[^1][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-07	^sip:00442071[^8][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-08	^sip:004420718[^7][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-09	^sip:0044207187[^8][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-10	^sip:00442071878[^3][0-9]*@	host	192.168.31.20	192.168.31.20	1
PSTN-11	^sip:004420718783[^6][0-9]*@	host	192.168.31.20	192.168.31.20	1

Figure 40: SES Address Map Priorities

After the host address maps have been added, select “Add Another Contact” and specify the contact parameters as shown in the following table, and click “Add”.

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 20: “Add Host Contact” Parameters



Figure 41: SES Add Host Contact Screen

3.2.3. Add Trusted Host

Select “Hosts” → “Add Host” from the left frame of the top level screen shown in **Figure 25**. Enter values in this screen as shown in the following table, accepting the default values for those parameters which are not listed. Click the “Add” button upon completion and the “Continue” button when the following screen is displayed.

Parameter	Usage
IP Address	Enter the IP address of the COLT Session Border Controller, as shown in Figure 1 .
Comment	Enter an appropriate comment for this host.

Table 21: “Add Trusted Host” Parameters

The screenshot shows the AVAYA Integrated Management SIP Server Management interface. The top header includes the AVAYA logo and the text "Integrated Management SIP Server Management" with the server IP "192.168.31.20". A navigation menu on the left lists "Top", "Users", "Address Map Priorities", "Adjunct Systems", "Conferences", "Emergency Contacts", "Export/Import to ProVision", "Hosts", and "List". The main content area is titled "Add Trusted Host" and contains three form fields: "IP Address*" with the value "10.0.0.30", "Host*" with a dropdown menu showing "192.168.31.20", and "Comment:" with the value "COLT SBC". Below the fields is a note: "Fields marked * are required." and an "Add" button, which is circled in red.

Figure 42: SES Add Trusted Host Screen

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 can ping the SBC port allocated 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 another without traversing the Avaya G250 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

Avaya Communication Manager experienced problems with making calls to GSM telephones which were doing international roaming due to timeout prior to connection. This problem is will be fixed by making the timeout an administrable value.

Avaya Communication Manager was also unable to route calls via the SIP trunk for Extended Call Forwarding. A temporary fix for this problem is to terminate telephone numbers configured as destinations for Extended Call Forwarding with a “#” character.

Calls arriving from the COLT network via the SIP trunk are signaled with one ring, although the System Features Distinctive Audible Ringing parameter for external calls was set to two rings.

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*, January 2008, Issue 4.0, Document Number 03-300509.
- [2] *Feature Description and Implementation for Avaya Communication Manager*, January 2008, Issue 6, Document Number 555-245-205.
- [3] *Installing and Administering SIP Enablement Services*, January 2008, Issue 5.0, Document Number 03-600768.
- [4] *Installing and Configuring the Avaya S8500 Media Server*, January 2008, Issue 6, Document Number 03-300143.
- [5] *SIP Enablement Services (SES) Implementation Guide*, January 2008, Issue 5, Document Number 16-300140.
- [6] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Issue 7, Document Number 555-233-507

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: <http://www.rfc-editor.org/rfcsearch.html>.

- [7] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
- [8] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

Appendix A: Sample SIP INVITE Messages

These traces were made at a port which mirrored the connection between the Avaya G250 Media Gateway and the COLT 2611 router.

Incoming call:

```
Request-Line: INVITE sip:00442071878362@192.168.31.20:5060;user=phone SIP/2.0
Message Header
  Via: SIP/2.0/UDP 10.0.0.30:5060;branch=z9hG4bK40jen4002061hacajlcl.1
  From: 00491772600205 <sip:00491772600205@va.sip.colt.net:5060;user=phone>;tag=SDj9p0901-1502827391
  To: +442071878362 <sip:00442071878362@192.168.31.20:5060;user=phone>
  Call-ID: SDj9p0901-938776341e6f6eabb26d7d81dd37753d-v300g00
  CSeq: 1 INVITE
  Max-Forwards: 69
  Supported: timer
  Session-Expires: 1800
  Min-SE: 1800
  Contact: 00491772600205 <sip:00491772600205@10.0.0.30:5060;transport=udp>
  Allow: INVITE,ACK,PRACK,SUBSCRIBE,BYE,CANCEL,NOTIFY,INFO,REFER,UPDATE
  Content-Type: application/sdp
  Content-Length: 356
Message body
  Session Description Protocol
    Session Description Protocol Version (v): 0
    Owner/Creator, Session Id (o): - 2099 0 IN IP4 10.0.0.30
    Session Name (s): Cisco SDP 0
    Connection Information (c): IN IP4 10.0.0.30
    Time Description, active time (t): 0 0
    Media Description, name and address (m): audio 20346 RTP/AVP 18 8 99 100
    Media Attribute (a): rtpmap:99 telephone-event/8000
    Media Attribute (a): fmp:99 0-15
    Media Attribute (a): rtpmap:100 X-NSE/8000
    Media Attribute (a): fmp: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=fmp:100 192-194,200-202
    Media Attribute (a): X-cap: 2 image udpt1 t38
```

Outgoing call:

```
Request-Line: INVITE sip:00496975056630@10.0.0.30:5060;lr SIP/2.0
Message Header
  Accept-Language: en
  Call-ID: 0b6171e53fbdc195147eb9cd300
  CSeq: 1 INVITE
  From: "extn 3004" <sip:442071878362@ffm.com:5061>;tag=0b6171e53fbdc194147eb9cd300
  Record-Route: <sip:192.168.31.20:5060;lr>,<sip:192.168.31.29:5061;lr;transport=tls>
  To: "00496975056630" <sip:00496975056630@192.168.31.20>
  Via: SIP/2.0/UDP 192.168.31.20:5060;branch=z9hG4bK0303032626266363632b6b.0,SIP/2.0/TLS
192.168.31.29;psrrposn=2;received=192.168.31.29;branch=z9hG4bK0b6171e53fbdc196147eb9cd300
  Content-Length: 165
  Content-Type: application/sdp
  Contact: "extn 3004" <sip:442071878362@192.168.31.29:5061;transport=tls>
  Max-Forwards: 61
  User-Agent: Avaya CM/R015x.00.0.825.4
  Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH
  Supported: 100rel,timer,replaces,join,histinfo
  Alert-Info: <cid:internal@invalid.unknown.domain>;avaya-cm-alert-type=internal
  Min-SE: 1800
  Session-Expires: 1800;refresher=uac
  P-Asserted-Identity: "extn 3004" <sip:442071878362@ffm.com:5061>
  History-Info: <sip:00496975056630@192.168.31.20>;index=1,"00496975056630"
<sip:00496975056630@192.168.31.20>;index=1.1
  Message body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): - 1 1 IN IP4 192.168.31.29
      Session Name (s): -
      Connection Information (c): IN IP4 192.168.31.21
      Bandwidth Information (b): AS:64
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 2050 RTP/AVP 8 127
      Media Attribute (a): rtpmap:8 PCMA/8000
      Media Attribute (a): rtpmap:127 telephone-event/8000
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