



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 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.

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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 Avaya Communication Manager, and Avaya IP telephones using the H.323 protocol as endpoints. The communication between Avaya Communication Manager and COLT VoIP Access SIP trunking network is via the TCP protocol.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [4] 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.

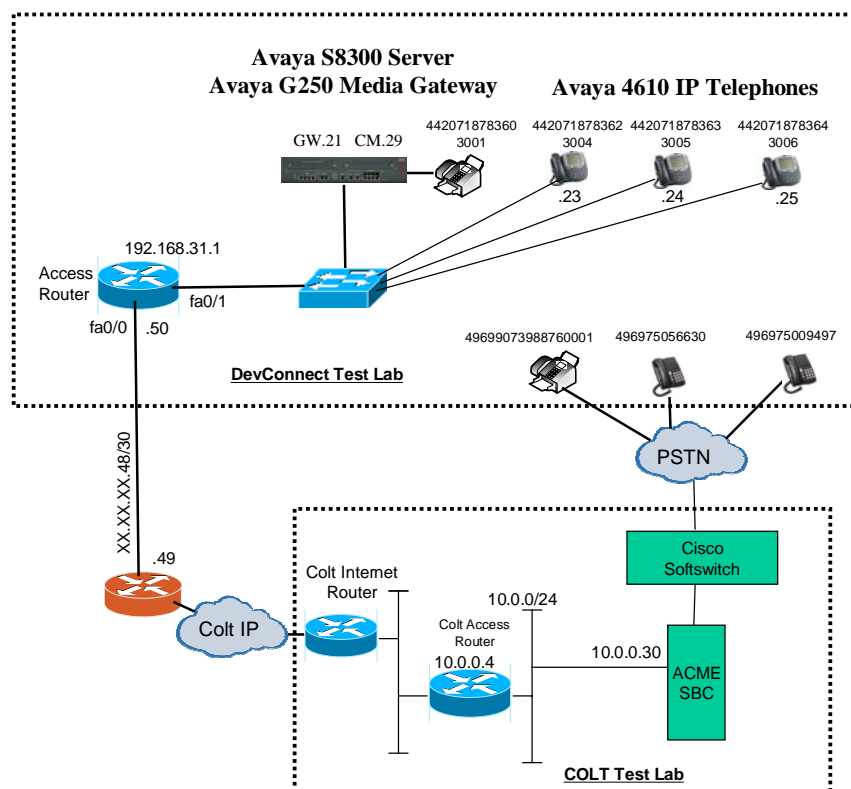


Figure 1: System Configuration

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. 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>.

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 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 G250 Gateway

The Avaya G250 Gateway can be configured via an ASCII terminal attached to the serial “Console” port of the G250. Log in with an appropriate user name and password, and enter the commands shown in **Figure 2** using the parameter values shown in **Table 2**.

Parameter	Usage
<media gateway ip address>	For the test configuration, this address is “192.168.31.21”
<ip network mask>	For the test configuration, this address is “255.255.255.0”
<default ip router>	For the test configuration, this address is “192.168.31.1”
<avaya 8300 server ip address>	For the test configuration, this address is “192.168.31.29”

Table 2: G250 Gateway Configuration Parameters

```
interface Vlan 1
ip address <media gateway ip address> <ip network mask>
exit
ip default-gateway <default ip router>
set mgc list <avaya 8300 server ip address>
```

Figure 2: G250 Gateway Configuration

3.2. Avaya Communication Manager

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

3.2.1. 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 3: Optional Features 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 3: Optional Features 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 4: Optional Features Form, Page 4

3.2.2. 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 4: Feature-Related System 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 5: Feature-Related System Parameters Form, Page 1

3.2.3. 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 5: 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 6: Media Gateway Form

3.2.4. SIP Interface to COLT VoIP Access SIP Service

Use the **change node-names ip** command to assign the name "colt" to the IP address of COLT VoIP Access SIP Service, which is used in the SIP signaling group, as shown in **Figure 8**.

```

change node-names ip                                     Page 1 of 2
                IP NODE NAMES
    Name          IP Address
    default       0.0.0.0
    procr         192.168.31.29
    colt          10.0.0.30

```

Figure 7: IP Node Names Form

Use the **add signaling-group** command to allocate a signaling group for the interface to the COLT VoIP Access SIP Service using the following parameters:

Parameter	Usage
Group Type	Enter "sip".
Transport Method	Enter "tcp".
Near-end Node Name	Enter "procr" to designate the S8300 processor as the near end node name.
Far-end Node Name	Enter "colt".
DTMF over IP	Enter "rtp-payload". This value is 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 6: Signaling-Group Parameters

```

add signaling-group 1                               Page 1 of 1
                SIGNALING GROUP

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

Near-end Node Name: procr          Far-end Node Name: colt
Near-end Listen Port: 5060        Far-end Listen Port: 5060
Far-end Domain:                Far-end Network Region: 1

                                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 8: 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 12 .
Service Type (p.1)	Enter "tie".
Signaling Group (p.1)	Enter the number of the signaling group allocated in Figure 8 .
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 7: 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 9: 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 10: 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 11: Trunk Group Form, p.4

3.2.5. 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 9 .

Table 8: 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 12: Dial Plan Analysis Table Form

3.2.6. 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 15**) 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 13: Feature Access Code 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 14: ARS Digit Analysis Table 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           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 15: 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 16: Public Unknown Numbering Form

3.2.7. Incoming Call Routing

Use the **change inc-call-handling-trmt trunk-group 1** 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/ Feature	Called Len	Called Number	Del	Insert	
tie	14	00442071878360	14	3001	
tie	14	00442071878362	14	3004	
tie	14	00442071878363	14	3005	
tie	14	00442071878364	14	3006	

Figure 17: Incoming Call Handling Treatment Form

3.2.8. 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 9: 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 18: 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 19: IP Codec Set Form, p.2

3.2.9. 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 an appropriate domain name to be assigned to the SIP trunk.
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in Figure 18 .

Table 10: 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
  Codec Set: 1           Intra-region IP-IP Direct Audio: yes
                        Inter-region IP-IP Direct Audio: yes
                        UDP Port Min: 2048           IP Audio Hairpinning? n
                        UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46           RTCP Reporting Enabled? y
  Audio PHB Value: 46                 RTCP MONITOR SERVER PARAMETERS
  Video PHB Value: 26                 Use Default Server Parameters? y
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
  H.323 Link Bounce Recovery? y           RSVP Enabled? n
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5

```

Figure 20: IP Network Region Form, p.2

3.2.10. 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 11: 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 21: 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 21**.

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

Table 12: 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 22: 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 21**.

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

Table 13: 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 23: Class of Service Form

3.2.11. Configure FAX Devices

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

Parameter	Usage
Type	Enter "2500" to assign an analog device.
Port	Enter the identifier for the analog port to which the FAX is attached. In this case "1V305" denotes gateway "1" module "3", which denotes the on-board interfaces, and "5" is the analog interface.
Name	Enter a name to identify the FAX or its user.

Table 14: Station Parameters for FAX Device

```

add station 3001                                     Page 1 of 4
                                                    STATION
Extension: 3001                                     Lock Messages? n          BCC: 0
  Type: 2500                                         Security Code:           TN: 1
  Port: 1V305                                       Coverage Path 1:        COR: 1
  Name: FAX                                           Coverage Path 2:        COS: 1
                                                    Hunt-to Station:        Tests? y

STATION OPTIONS
  XOIP Endpoint type: auto                          Time of Day Lock Table:
  Loss Group: 1                                       Message Waiting Indicator: none
  Off Premises Station? n

  Survivable COR: internal
  Survivable Trunk Dest? y
  
```

Figure 24: Station Form for FAX Device

3.3. Avaya IP Telephones

All Avaya IP Telephones must be configured such that the default gateway is assigned to the IP address of the Access Router which provides access to the COLT SIP network. For the test configuration, a value of "192.168.31.1" was used, as shown in **Figure 1**. This value can either be assigned manually to each telephone, or automatically via DHCP.

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 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 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 will be fixed in a future release 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 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] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Issue 7, Document Number 555-233-507

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

- [4] *RFC 3261 - SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
- [5] *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 captured using a port which mirrored the connection between the Avaya G250 Media Gateway and the COLT 2611 router.

Incoming call:

```
Ethernet II, Src: 192.168.31.1 (00:06:28:e2:36:a1), Dst: Tenovis_c5:87:e4 (00:07:3b:c5:87:e4)
Internet Protocol, Src: 10.0.0.30 (10.0.0.30), Dst: 192.168.31.29 (192.168.31.29)
Transmission Control Protocol, Src Port: 18130 (18130), Dst Port: 5060 (5060), Seq: 0, Ack: 0,
Len: 1041
Session Initiation Protocol
Request-Line: INVITE sip:00442071878362@192.168.31.29:5060;user=phone SIP/2.0
Message Header
  Via: SIP/2.0/TCP 10.0.0.30:5060;branch=z9hG4bK4mm9pt30dojhic0kc701.1
  From: 00496975050 <sip:00496975050@va.sip.colt.net:5060;user=phone>;tag=SD9idd601-884771343
  To: +442071878362 <sip:00442071878362@192.168.31.29:5060;user=phone>
  Call-ID: SD9idd601-5906715d9711b421a642397999d26807-v300g00
  CSeq: 1 INVITE
  Max-Forwards: 69
  Supported: timer
  Session-Expires: 1800
  Min-SE: 1800
  Contact: 00496975050 <sip:00496975050@10.0.0.30:5060;transport=tcp>
  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): - 231 0 IN IP4 10.0.0.30
    Owner Username: -
    Session ID: 231
    Session Version: 0
    Owner Network Type: IN
    Owner Address Type: IP4
    Owner Address: 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 20106 RTP/AVP 18 8 99 100
  Media Attribute (a): rtpmap:18 G729/8000
  Media Attribute (a): fmp:18 annexb=no
  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 udptl t38
```

Outgoing call:

```
Ethernet II, Src: Tenovis_c5:87:e4 (00:07:3b:c5:87:e4), Dst: 192.168.31.1 (00:06:28:e2:36:a1)
Internet Protocol, Src: 192.168.31.29 (192.168.31.29), Dst: 10.0.0.30 (10.0.0.30)
Transmission Control Protocol, Src Port: 10014 (10014), Dst Port: 5060 (5060), Seq: 0, Ack: 0,
Len: 1274
Session Initiation Protocol
Request-Line: INVITE sip:00496975056174@10.0.0.30 SIP/2.0
Message Header
  From: "extn 3004" <sip:00442071878362@ffm.com>;tag=0ae8eae6f43dd12e04866109600
  To: "00496975056174" <sip:00496975056174@10.0.0.30>
  Call-ID: 0ae8eae6f43dd12f04866109600
  CSeq: 1 INVITE
  Max-Forwards: 62
  Route: <sip:10.0.0.30;lr;phase=terminating;transport=tcp>
  Record-Route: <sip:192.168.31.29;lr;transport=tcp>
  Via: SIP/2.0/TCP 192.168.31.29;branch=z9hG4bK0ae8eae6f43dd13004866109600
  User-Agent: Avaya CM/R015x.00.0.825.4
  Supported: 100rel, timer, replaces, join, histinfo
  Allow: INVITE, CANCEL, BYE, ACK, PRACK, SUBSCRIBE, NOTIFY, REFER, OPTIONS, INFO, PUBLISH
  Contact: "extn 3004" <sip:00442071878362@192.168.31.29;transport=tcp>
  Session-Expires: 1800;refresher=uac
  Min-SE: 1800
  P-Asserted-Identity: "extn 3004" <sip:00442071878362@ffm.com>
  Accept-Language: en
  Content-Type: application/sdp
  History-Info: <sip:00496975056174@10.0.0.30>;index=1
  History-Info: "00496975056174" <sip:00496975056174@10.0.0.30>;index=1.1
  Alert-Info: <cid:internal@invalid.unknown.domain>;avaya-cm-alert-type=internal
  Content-Length: 191
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 2056 RTP/AVP 8 18 127
    Media Attribute (a): rtpmap:8 PCMA/8000
    Media Attribute (a): rtpmap:18 G729/8000
    Media Attribute (a): rtpmap:127 telephone-event/8000
```

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