Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the COLT VoIP Access SIP Service and an Avaya Aura™ 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 Aura™ Communication Manager. The Avaya solution consists of Avaya Aura™ 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.

1. Introduction

These Application Notes describe the procedure for configuring Session Initiation Protocol (SIP) trunking between the COLT Office Voice SIP trunking service and Avaya SIP telephony solution consisting Avaya Aura™ Communication Manager, and Avaya IP telephones. The communication between Avaya Aura™ Communication Manager and COLT Office Voice 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 Communication Manager and the trunking 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 Number, etc.

1.1. Interoperability Compliance Testing

The following features were tested:

- Incoming & outgoing basic calls, including busy, no answer, calling party hang-up, called party hang-up
- Outbound calls to domestic and international PSTN and GSM national and international endpoints
- Codec support and priority selection
- DTMF tone generation and recognition using RFC 2833
- Calling Party Number and Called Party Number presentation and restriction for incoming and outgoing calls
- Call Forwarding unrestricted / Busy / No Answer to local extension, PSTN, and GSM endpoint.
- Supervised Call Transfer / Blind Call Transfer to local extension, PSTN, and GSM endpoint.
- Conference Call with both local and PSTN endpoints, also with mixed codecs.
- Short and long Fax Send / Receive using T.38, using both the G.711 and G.729 codecs.
- Simultaneous Calls
- Long Calls
- Calls with both ends muted
- Recovery from both trunk and phone connection failure

1.2. Support

Telephone support is available on a national basis as shown in the following table, which shows the hotline number for each country where support is available, as well as a toll-free number if available.

Country	Hot Line	Toll-Free
Austria	(+43) 1 20 500 500	0800 880 990
Belgium	(+32) 2 790 16 29	0800 50701
Switzerland	(+41) 44 560 0720	0800 560 560
Denmark	(+45) 70 27 35 59	
France	(+33) 1 70995600	0800 948 888
Germany	(+49) 69 56606 3115	0800 855 4444
Ireland	(+34) 9355 02568	1800 944040
Italy	(+39) 0230 329 550	0800 909 377
Netherlands	(+31) 20 888 2433	0800 265 8023
Portugal	(+351) 211 200 222	808 780 222
Spain	(+34) 913 206018	901 888400
Sweden	(+46) 8781 8333	
UK	(+44) 203 140 2023	0800 136 166

Reference Configuration

The following diagram illustrates the configuration used for testing:

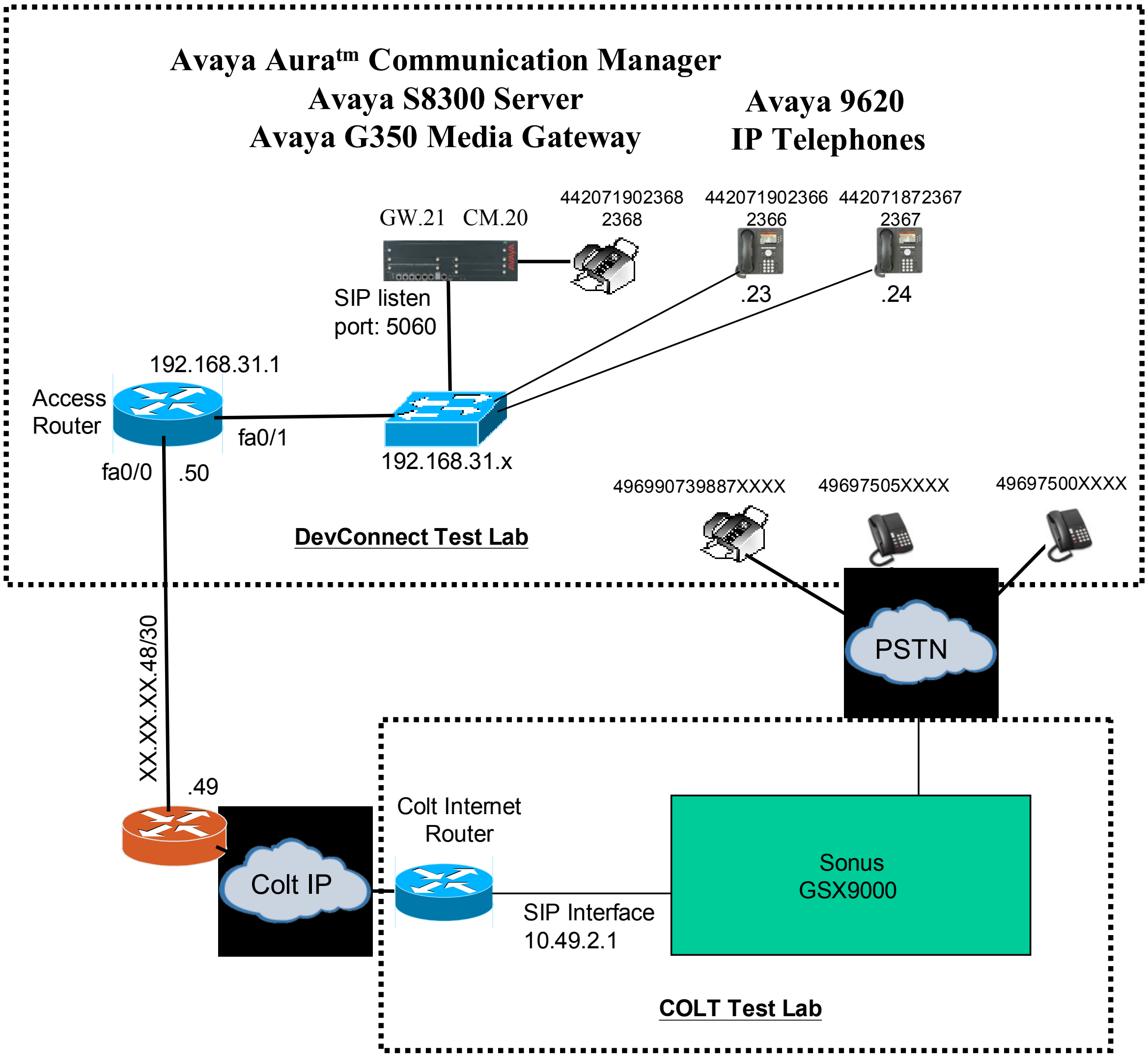

Figure 1: System Configuration

Communication Manager and the COLT Office Voice SIP trunk are configured to support T.38 fax transmission.

2. Equipment and Software Validated

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

Equipment	Firmware/Software Version
Communication Manager	R015x.02.0.947.3 Update: 02.0.947.3-17534
Avaya G350 Media Gateway	26.36.0
Avaya G350 Integrated Analog Interface	HW05/FW071
Avaya 9620 IP Telephones (H.323)	3.002
Sonus PSX	7.2.4R1
Sonus GSX	7.2.4R1
Sonus DSI	7.2.3R1
Sonus EMS	7.2.4R0

Table 1: Equipment and Software Validated

3. Configuration

3.1. Avaya Aura™ Communication Manager

The Communication Manager configuration was performed using the System Access Terminal (SAT) and the Web interface to Communication Manager

3.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Communication Manager is licensed to meet the minimum requirements to interoperate with the COLT Office Voice SIP trunk. 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.
ARS (p.3)	This parameter must be set to "y".
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: Optional Features Parameters

```
display system-parameters customer-options                       Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 1000  70
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 10    0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                        Maximum Video Capable Stations: 0     0
                   Maximum Video Capable IP Softphones: 1000  0
                       Maximum Administered SIP Trunks: 1000  285
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 10    0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   1
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 2: Optional Features Form, Page 2

```
display system-parameters customer-options                       Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n          Audible Message Waiting? n
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? n                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? n                         CAS Main? n
Answer Supervision by Call Classifier? n                Change COR by FAC? n
                                 ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? n   Cvg Of Calls Redirected Off-net? n
          ARS/AAR Dialing without FAC? y                      DCS (Basic)? n
          ASAI Link Core Capabilities? y                DCS Call Coverage? n
          ASAI Link Plus Capabilities? y               DCS with Rerouting? n
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? n
              ATM WAN Spare Processor? n                          DS1 MSP? n
                                 ATMS? n             DS1 Echo Cancellation? y
                  Attendant Vectoring? n
```

Figure 3: Optional Features Form, Page 3

```
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

3.1.2. 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: Feature-Related System Parameters

```
change system-parameters features                               Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                   Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    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: music Type:
              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

3.1.3. 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: "2"	Four digit numbers starting with "2" are for local extensions.
Dialed string: "*01"	The dialed string "*01" is the Trunk Access Code (TAC) used in **Figure 8**.

Table 4: Dial Plan Analysis Parameters

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
        0          1   fac
        2          4   ext
        *01        3   dac
```

Figure 6: Dialplan Analysis Table Form

3.1.4. SIP Interface to COLT Office Voice

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 processor interface as the near end node name.
Far-end Node Name	Enter "`colt-sonus-sip`".
Near-end Listen Port	Enter "5060".
Far-end Listen Port	Enter "5060".
DTMF over IP	Enter "rtp-payload". This value is used to have 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
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n





   Near-end Node Name: procr                Far-end Node Name: colt-sonus-sip
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       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
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? n                      Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Figure 7: Signaling Group Form

Use the **add trunk-group** <**n**> command, were **n** is an unused trunk number, to allocate a trunk group to be used as an interface to the COLT Office Voice SIP trunk. 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 6**
Service Type (p.1)	Enter "public-ntwrk".
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 handled by the SIP trunk.
Preferred Minimum Session Refresh Interval (p.2)	Enter "900" seconds, as required by the COLT Office Voice SIP trunk interface. This should be half of the Session Refresh Interval which is configured for the COLT Office Voice SIP trunk.
Send Transferring Party Information (p.4)	Enter "y".
Send Diversion Header (p.4)	Enter "y".
Telephone Event Payload Type (p.4)	Enter a value which matches that which is configured for the Sonus SBC.

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: *01
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          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: yes

                                            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

```
add trunk-group 1                                           Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? y

                    Send Diversion Header? y
                  Support Request History? n
             Telephone Event Payload Type: 101
```

Figure 10: Trunk Group Form, p.4

3.1.5. Outgoing Call Routing

For the test configuration, outgoing dialed numbers have the format 0<national number>, or 00<country code><number>. 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 13**) 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 11: Feature Access Code Form

Use the **change ars analysis** command to designate that ars calls to all numbers 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 12: 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 11,** should be restored as the leading digit of the called party number so that it has the format "00"<country code><number> or "0"<national number>.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: COLT
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 1    0       1                0                                    n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

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

Figure 13: Route Pattern Form

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

```
change display public-unknown-numbering 4                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext            Trk       CPN          CPN
Len Code           Grp(s)    Prefix       Len
                                                   Total Administered: 1
 4  236            1         0044207190    14          Maximum Entries: 240
```

Figure 14: Public Unknown Numbering Form

3.1.6. 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 endpoints shown in **Figure 1**.

```
change display inc-call-handling-trmt trunk-group 1                    Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    14 0044207190       10
```

Figure 15: Incoming Call Handling Treatment Form

3.1.7. Configure Codec Sets

Use the **change ip-codec-set** command to designate a codec set to be used for communication with the COLT Office Voice SIP trunk. Testing was done with both the G.729A and G.711A 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.729A" and "G.711A" as the codecs to be used for communication with the COLT Office Voice SIP trunk.
FAX Mode (p. 2)	Enter "t.38-standard" to specify that the T.38 standard should be used to transmit FAX documents via the COLT Office Voice SIP trunk.
TDD/TTY Mode (p. 2)	Enter "off".

Table 7: 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.729A            n         2        20
 2: G.711A            n         2        20
```

Figure 16: IP Codec Set Form, p.1

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

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

Figure 17: IP Codec Set Form, p.2

3.1.8. Configure IP Network Region

Use the **change ip-network-region <x>** command to designate a network region to be used for the COLT Office Voice 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 16**.

Table 8: IP Network Region Parameters

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: ffm.com
    Name: FFM
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 18: IP Network Region Form, p.2

3.1.9. Configure Telephone Stations

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

Parameter	Usage
Type (p. 1)	Enter the type identifier of local telephone.
Security Code (p. 1)	Enter the security code to be assigned to the station for security purposes.
Name (p. 1)	Enter a name to identify the station or its user.

Table 9: Station Parameters for IP Telephones

```
add station 2366                                             Page   1 of   5
                                     STATION

Extension: 2366                         Lock Messages? n              BCC: 0
     Type: 9620                         Security Code: 123456          TN: 1
     Port: S00000                     Coverage Path 1:                COR: 1
     Name: extn 2366                  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: 2366
            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? 5
```

Figure 19: Station Form for IP Telephones, page 1

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 Office Voice SIP trunk, as shown in **Figure 1**.

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

Table 10: 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 20: 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 Office Voice SIP trunk, as shown in **Figure 1**.

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

Table 11: 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 21: Class of Service Form

3.1.10. Configure FAX Devices

Use the **add station** command to add the fax device show in **Figure 1** 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 to be attached.
Name	Enter a name to identify the FAX or its user.

Table 12: Station Parameters for FAX Device

```
add station 2368                                             Page   1 of   4
                                     STATION

Extension: 2368                          Lock Messages? n               BCC: 0
     Type: 2500                          Security Code:                  TN: 1
     Port: 001V702                     Coverage Path 1:                 COR: 1
     Name: SIP 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 22: Station Form for FAX Device

4. General Test Approach and Test Results

The following issues were encountered during testing:

- Calls to local extensions which are forwarded via the COLT Office Voice SIP trunk will show a configurable administrative number (which is common for all local extensions).
- Outgoing FAX messages can be sent with the G.729 codec only if the G.711 codec is included as a secondary codec for Communication Manager, as shown in **Figure 16**.

5. Verification Steps

- Use the “status signaling-group <x>” command from the SAT terminal to verify that the “Group State” has a value of “in-service”, where <x> is the number of the SIP trunk attached to the COLT Office Voice SIP trunk.

```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1                               Active NCA-TSC Count: 0
      Group Type: sip                              Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

Figure 23: Signaling-Group Status

- Verify that local extensions can call to and receive calls from endpoints attached to the PSTN and mobile networks.
- Verify the calling party number is presented correctly at the called endpoint for both incoming and outgoing calls.
- Verify that unanswered incoming calls can be dialed via the call log of the called endpoint.
- Verify that locally attached FAX devices can send and receive facsimile messages without dropouts.

6. Conclusion

These Application Notes contain instructions for configuring Communication Manager to connect to the COLT Office Voice SIP trunk. All test cases passed with exceptions noted in **Section 4**.

7. Additional References

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

[1] *Administering Avaya Aura™ Communication Manager*, January 2009, Issue 5.0, Document Number 03-300509.

[2] *Avaya Aura™ Communication Manager Feature Description and Implementation,* May 2009, Issue 7, Document Number 555-245-205.

[3] *Avaya Extension to Cellular User Guide Avaya Aura™ Communication Manager,* April 2009, Issue 12, Document Number 210-100-700

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

[6] RFC 3555 - *MIME Type Registration of RTP Payload Formats*, July 2003, IETF Standard

Appendix A: Sample SIP INVITE Messages

These traces were captured using a port which mirrored the connection between the Avaya S8300 Processor Ethernet interface and the COLT Office Voice IMS network.

Incoming call:

```
Session Initiation Protocol
    Request-Line: INVITE sip:00442071902366@192.168.31.20:5060 SIP/2.0
    Message Header
        Via: SIP/2.0/TCP 10.49.2.1:5060;branch=z9hG4bK0fBc09d1ca55db724f6
        From: <sip:0049163685XXXX@10.49.2.1>;tag=gK0f2569b4
        To: <sip:00442071902366@192.168.31.20>
        Call-ID: 252653521_27523328@10.49.2.1
        CSeq: 14782 INVITE
        Max-Forwards: 70
        Allow:
INVITE,ACK,CANCEL,BYE,REGISTER,REFER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE,OPTIONS,MESSAGE,PUBLISH
        Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
        Contact: <sip:0049163685XXXX@10.49.2.1:5060;transport=tcp>
        P-Asserted-Identity: <sip:0049163685XXXX@10.49.2.1:5060>
        Supported: timer,100rel
        Session-Expires: 1800
        Min-SE: 90
        Content-Length:  302
        Content-Disposition: session; handling=optional
        Content-Type: application/sdp
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): Sonus_UAC 31141 13965 IN IP4 10.49.2.1
            Session Name (s): SIP Media Capabilities
            Connection Information (c): IN IP4 10.49.2.10
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 21110 RTP/AVP 18 8 2 101
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): fmtp:18 annexb=no
            Media Attribute (a): rtpmap:8 PCMA/8000
            Media Attribute (a): rtpmap:2 G726-32/8000
            Media Attribute (a): rtpmap:101 telephone-event/8000
            Media Attribute (a): fmtp:101 0-15
            Media Attribute (a): sendrecv
            Media Attribute (a): ptime:20
```

Outgoing call:

```
Session Initiation Protocol
Request-Line: INVITE sip:00491636853087@10.49.2.1 SIP/2.0
Message Header
    From: "extn 2366" <sip:00442071902366@ffm.com>;tag=802053a36bc5de118c4ac4c4d500
    To: "0049163685XXXX" <sip:0049163685XXXX@10.49.2.1>
    Call-ID: 802053a36bc5de119c4ac4c4d500
    CSeq: 1 INVITE
    Max-Forwards: 71
    Route: <sip:10.49.2.1;lr;phase=terminating;transport=tcp>
    Record-Route: <sip:192.168.31.20;lr;transport=tcp>
    Via: SIP/2.0/TCP 192.168.31.20;branch=z9hG4bK802053a36bc5de11ac4ac4c4d500
    User-Agent: Avaya CM/R015x.02.0.947.3
    Supported: timer, replaces, join, 100rel
    Allow: INVITE, CANCEL, BYE, ACK, PRACK, SUBSCRIBE, NOTIFY, REFER, OPTIONS, INFO, PUBLISH
    Contact: "extn 2366" <sip:00442071902366@192.168.31.20;transport=tcp>
    Session-Expires: 1800;refresher=uac
    Min-SE: 1800
    P-Asserted-Identity: "extn 2366" <sip:00442071902366@ffm.com>
    P-Charging-Vector: icid-value="AAS:69-a35320801dec56bc44a0c17d5c4"
    Content-Type: application/sdp
    Alert-Info: <cid:internal@invalid.unknown.domain>;avaya-cm-alert-type=internal
    Content-Length: 165
Message Body
    Session Description Protocol
        Session Description Protocol Version (v): 0
        Owner/Creator, Session Id (o): - 1 1 IN IP4 192.168.31.20
        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 2452 RTP/AVP 8 101
        Media Attribute (a): rtpmap:8 PCMA/8000
        Media Attribute (a): rtpmap:101 telephone-event/8000
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.