Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Vodafone Office Voice 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 Vodafone Office Voice 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 Vodafone Office Voice SIP trunking service and Avaya SIP telephony solution consisting Avaya Aura Communication Manager, and Avaya IP and digital telephones. The communication between Avaya Aura Communication Manager and Vodafone 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 Avaya Aura Communication Manager and the trunking service offered by Vodafone. 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 no answer, calling party hang-up, called party hang-up
- Outbound calls to domestic and international PSTN and GSM 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 Hold / Resume
- Call Forwarding unrestricted / Busy / No Answer
- Supervised Call Transfer / Blind Call Transfer
- Conference Call
- Fax Send / Receive using T.38, using both the G.711 and G.729 codecs.
- Simultaneous Calls
- Long Calls
- Extension to Cellular (EC500)
- Recovery from trunk failure

Direct media connection (shuffling) was not tested, as this was not supported by the test environment. Wherever possible, the tests were performed with combinations of local extensions, PSTN telephones, and GSM handsets registered with various providers.

1.2. Support

Support is available at: http://www.vodafone.nl/zakelijk/vast_internet/Vodafone_Office_Voice/

2. Reference Configuration

The following diagram illustrates the configuration used for testing:

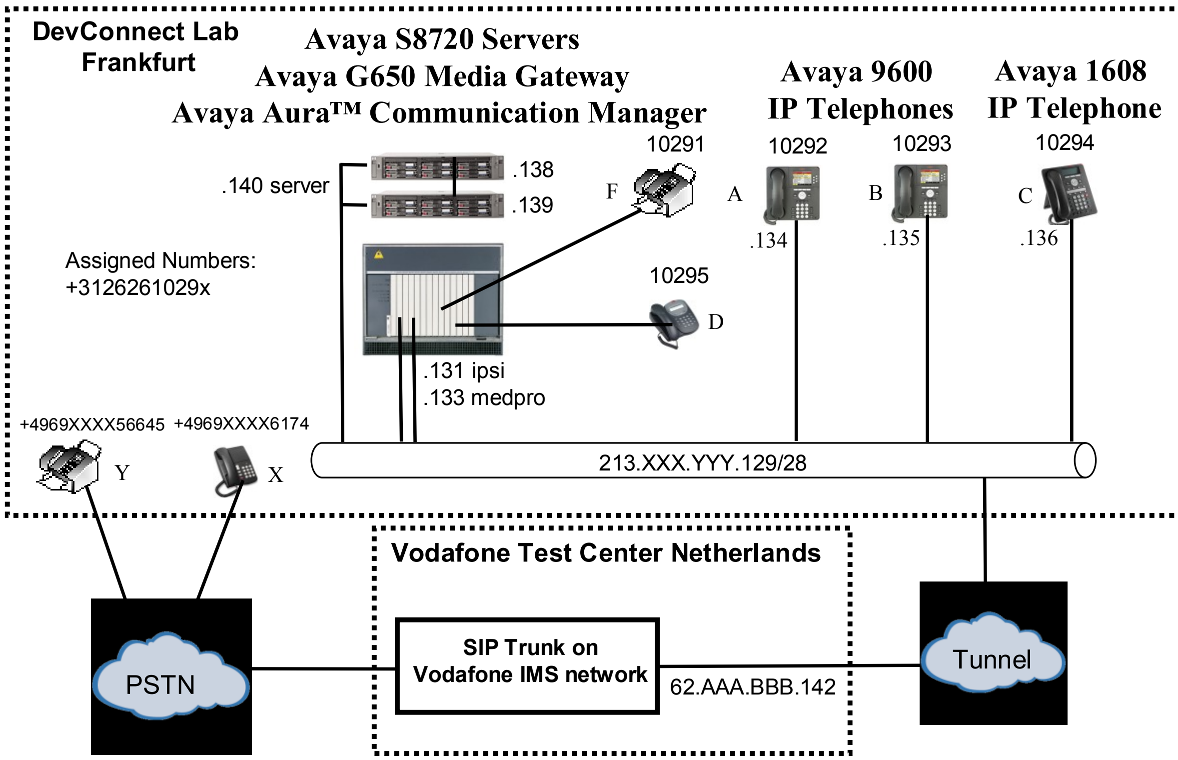


Figure 1: System Configuration

In the above diagram, Avaya IP Telephones are attached to the Avaya S8720 Server running Avaya Aura Communication Manager via Processor Ethernet.

The Avaya Aura Communication Manager / Vodafone Office Voice SIP trunk configuration used for testing does not support direct IP media connections. Avaya Aura Communication Manager and the Vodafone Office Voice SIP trunk are configured to support T.38 fax transmission.

3. Equipment and Software Validated

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

Equipment	Firmware/Software Version
Avaya S8720 Server	R015x.02.0.947.3 Update: 02.0.947.3-17294
Avaya TN2312BP IP Server Interface	HW15/FW046
Avaya TN2302AP IP Media Processor	HW20/FW120
Avaya TN793CP Analog Line Interface	HW07/FW010
Avaya TN2214CP Digital Line Interface	HW08/FW015
Avaya 9640G IP Telephones (H.323)	3.002
Avaya 1608 IP Telephone (H.323)	3.0
ACME PACKET Net-net 4250 session border controller	Version C5.1.0 patch 9 (Build 170)

Table 1: Equipment and Software Validated

4. Configuration

4.1. Avaya Aura Communication Manager

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

4.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Avaya Aura Communication Manager is licensed to meet the minimum requirements to interoperate with the Vodafone 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 Off-PBX Telephones – EC500 (p.1)	This parameter must be large enough to support the number of stations which are paired with cell phones.
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   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V15                                 Software Package: Standard
       Location: 2                                RFA System ID (SID): 1
       Platform: 6                                RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 44000 670
                                      Maximum Stations: 36000 165
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 1000  1
                    Maximum Off-PBX Telephones -   OPS: 1000  18
                    Maximum Off-PBX Telephones - PBFMC: 1000  0
                    Maximum Off-PBX Telephones - PVFMC: 1000  0
                    Maximum Off-PBX Telephones - SCCAN: 1000  0
```

Figure 1: Optional Features Form, Page 1

```
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

4.1.2. 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”.

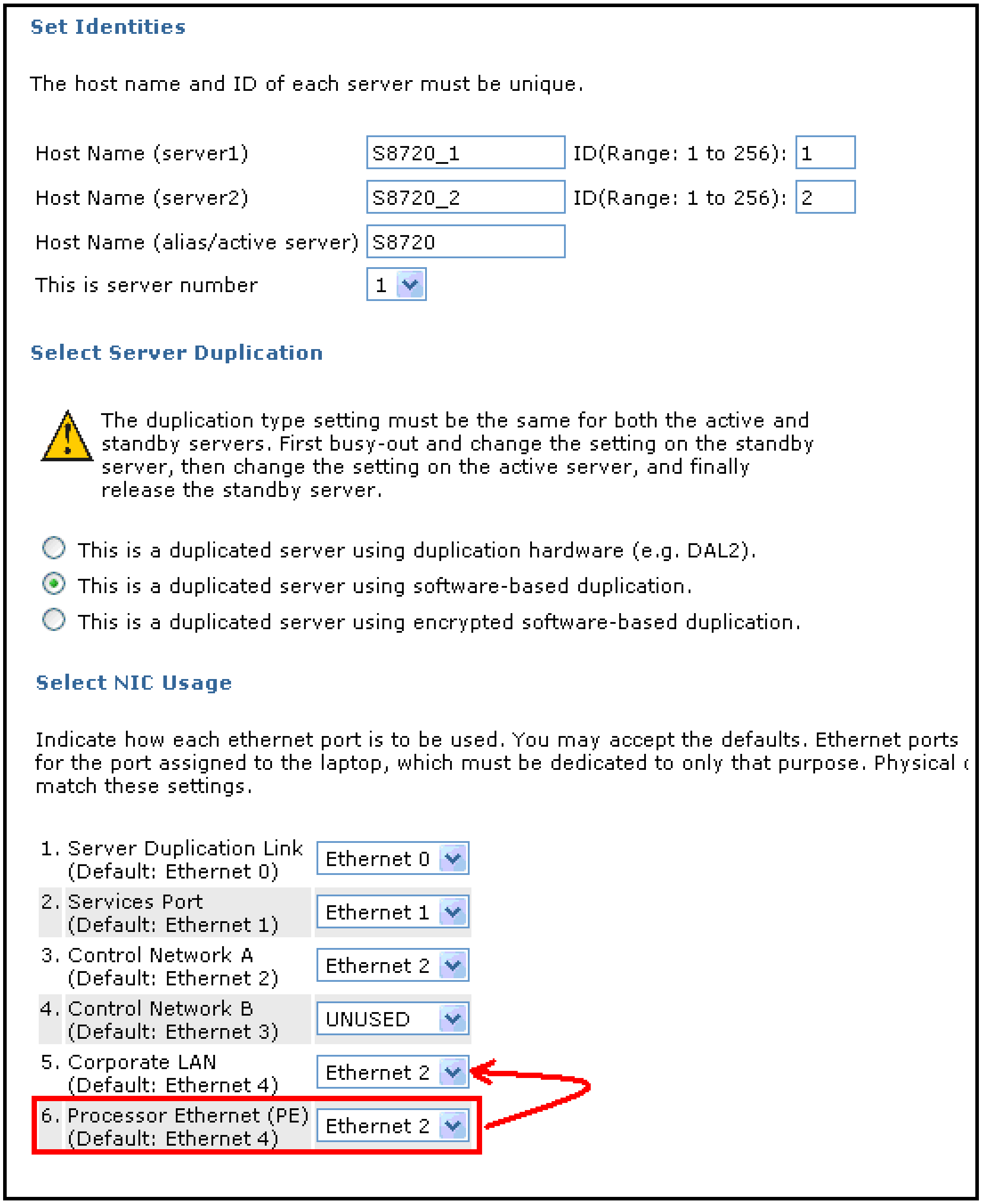
Table 3: Feature-Related System Parameters

```
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

4.1.3. Configure Processor Ethernet

Configure the S8720 server from the Web interface via the “Install” -> “Configure Server” menu item. From the “Set Identities” form, assign the “Processor Ethernet” interface to the same interface as that used by the “Corporate LAN” interface.



Set Identities

The host name and ID of each server must be unique.

Host Name (server1)	S8720_1	ID(Range: 1 to 256):	1
Host Name (server2)	S8720_2	ID(Range: 1 to 256):	2
Host Name (alias/active server)	S8720		
This is server number	1		

Select Server Duplication

The duplication type setting must be the same for both the active and standby servers. First busy-out and change the setting on the standby server, then change the setting on the active server, and finally release the standby server.

- This is a duplicated server using duplication hardware (e.g. DAL2).
- This is a duplicated server using software-based duplication.
- This is a duplicated server using encrypted software-based duplication.

Select NIC Usage

Indicate how each ethernet port is to be used. You may accept the defaults. Ethernet ports for the port assigned to the laptop, which must be dedicated to only that purpose. Physical match these settings.

1. Server Duplication Link (Default: Ethernet 0)	Ethernet 0
2. Services Port (Default: Ethernet 1)	Ethernet 1
3. Control Network A (Default: Ethernet 2)	Ethernet 2
4. Control Network B (Default: Ethernet 3)	UNUSED
5. Corporate LAN (Default: Ethernet 4)	Ethernet 2
6. Processor Ethernet (PE) (Default: Ethernet 4)	Ethernet 2

Figure 6: Server Set Identities Form

On the “Configure Interfaces” form, assign the “Gateway” address to the “IP address for PE Health Check” (note that the addresses are only partly shown, for security reasons).

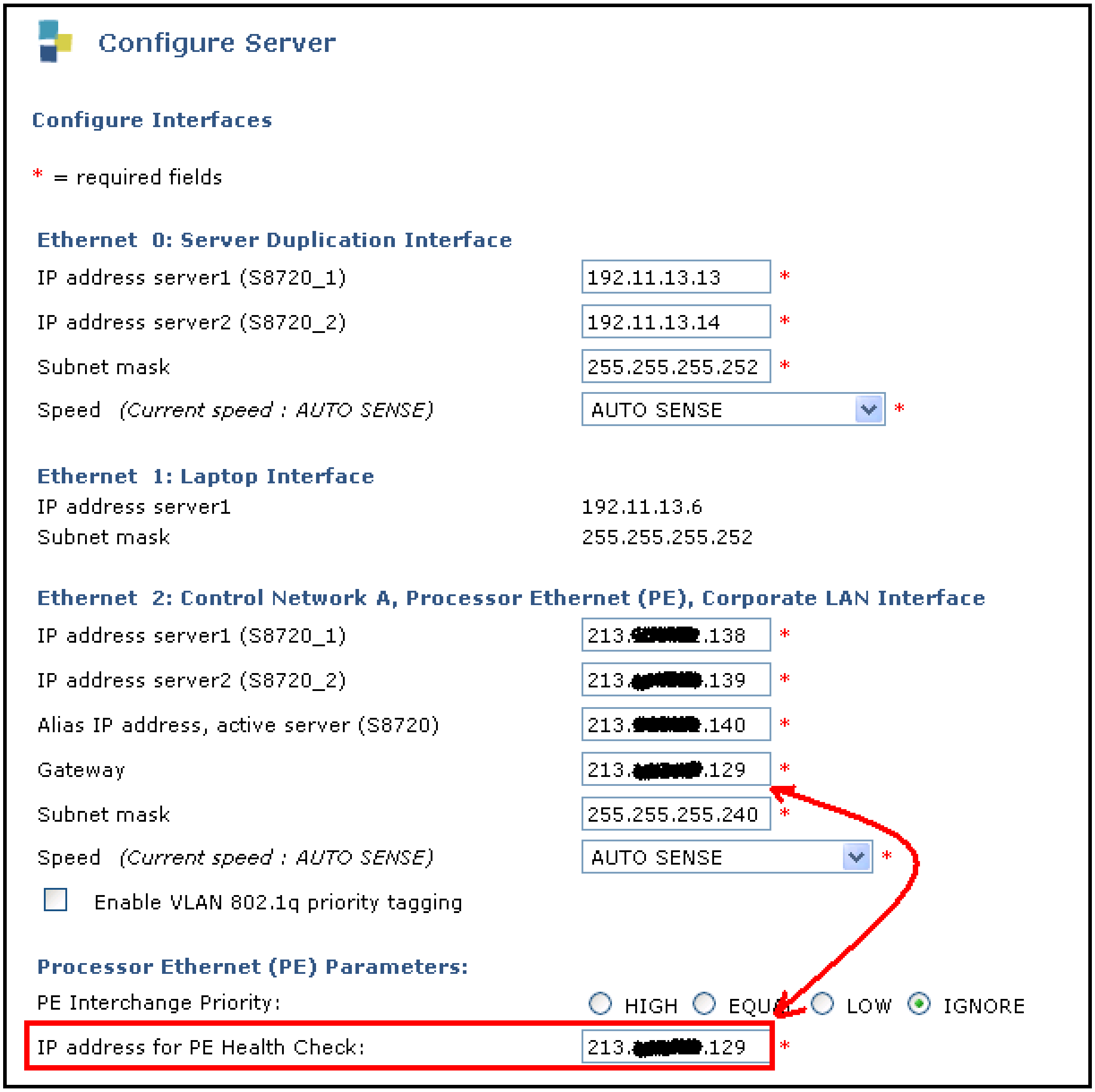


Figure 7: Server Configure Interfaces Form

Enter the "add ip-interface procr" command, and set the "Enable Interface" parameter to "y" to enable Processor Ethernet.

```
add ip-interface procr                                        Page   1 of   1
                                  IP INTERFACES


                  Type: PROCR
                                                       Target socket load: 19200

      Enable Interface? y                           Allow H.323 Endpoints? y
                                                     Allow H.248 Gateways? y
        Network Region: 1                             Gatekeeper Priority: 5



                                 IPV4 PARAMETERS
             Node Name: procr
           Subnet Mask: /28
```

Figure 8: Media Gateway Form

4.1.4. Dial Plan

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

Parameter	Usage
Dialed string: "0"	Use a "0" as Facilities Access Code (FAC) to access external telephone numbers.
Dialed string: "1"	Five digit numbers starting with "1" are for local extensions.
Dialed string: "*86"	The dialed string "*86" is the Trunk Access Code (TAC) used in **Figure 12**.
Dialed string: "*7"	The dialed strings beginning with "*7" are used for Feature Access Codes for EC500.
Dialed string: "#7"	The dialed strings beginning with "#7" are used for Feature Access Codes for EC500.

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
        1          5   ext
        *86        3   dac
        *7         3   fac
        #7         3   fac
```

Figure 9: Dialplan Analysis Table Form

4.1.5. SIP Interface to Vodafone Office Voice

Use the **disable test-number 1387** command to prevent Avaya Aura Communication Manager from using "ping" messages to test if the availability of the SIP trunk connection, as "ping" messages are not recognized by Vodafone Office Voice.

Use the **change node-names ip** command to assign the name "Vodafone" to the IP address of Vodafone Office Voice (note that the address is only partly shown, for security reasons).

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
procr               213.XXX.YYY.140
Vodafone            62.AAA.BBB.142
```

Figure 10: IP Node Names Form

Use the **add signaling-group** command to allocate a signaling group for the interface to the Vodafone Office Voice service using the following parameters:

Parameter	Usage
Group Type	Enter "sip".
Transport Method	Enter "tcp".
Near-end Node Name	Enter "procr" to designate the S8720 processor pair as the near end node name.
Far-end Node Name	Enter "Vodafone".
Near-end Listen Port	Enter "5060".
Far-end Listen Port	Enter "5060".
Far-end Network Region	Enter "1".
DTMF over IP	Enter "rtp-payload". This value is used to have Avaya Aura Communication Manager send DTMF transmissions using RFC 2833.
Direct IP-IP Audio Connections	Enter "n" to disallow direct IP-IP endpoint connections (shuffling). This value was chosen due limitations inherit to the network topology used for the test configuration. This value should be set to "y" for configurations which allow direct media.

Table 5: Signaling-Group Parameters

```
add signaling-group 86                                       Page   1 of   1
                                SIGNALING GROUP

 Group Number: 86             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n




   Near-end Node Name: procr                 Far-end Node Name: Vodafone
 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? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n
                                                  Alternate Route Timer(sec): 6
```

Figure 11: 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 Vodafone 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 9**
Service Type (p.1)	Enter "public-ntwrk".
Signaling Group (p.1)	Enter the number of the signaling group allocated in **Figure 11**.
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.
Redirect On OPTIM Failure (p.2)	Enter a timeout value, in milliseconds, to recover from failed responses for EC500. For the tested configuration, the average response time was 7000ms, so a value of 9000ms was chosen.
Preferred Minimum Session Refresh Interval (p.2)	Enter "900" seconds, as required by the Vodafone Office Voice SIP trunk interface. This should be half of the Session Refresh Interval which is configured for the Vodafone Office Voice SIP trunk.
Prepend '+' to Calling Number (p.4)	Enter "y".
Send Transferring Party Information (p.4)	Enter "y".
Send Diversion Header (p.4)	Enter "y".

Table 6: Trunk Group Parameters

```
add trunk-group 86                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 86                  Group Type: sip           CDR Reports: y
  Group Name: VODAFONE                   COR: 1        TN: 1        TAC: *86
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                 Signaling Group: 86
                                               Number of Members: 30
```

Figure 12: Trunk Group Form, p.1

```
add trunk-group 86                                          Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 9000

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

Figure 13: Trunk Group Form, p.2

```
change trunk-group 86                                          Page   4 of  21
                               PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? y
      Send Transferring Party Information? y
                 Network Call Redirection? n
                     Send Diversion Header? y
                  Support Request History? n
             Telephone Event Payload Type:
```

Figure 14: Trunk Group Form, p.4

4.1.6. 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 17**) 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 15: Feature Access Code Form

Use the **change ars analysis** command to designate that all ars calls to German numbers beginning with "0" with a minimum length of "7" digits and a maximum length of "15" digits be routed via route pattern "86" 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
    049                  7    15    86        pubu         n
```

Figure 16: ARS Digit Analysis Table Form

Use the **change route-pattern** command to designate that calls routed via route pattern "86" be routed via trunk group "86", and that the "0" digit which was removed by the "Auto Route Selection Access Code 1", shown in **Figure 15,** 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 86                                         Page   1 of   3
                    Pattern Number: 1   Pattern Name: Vodafone
                                          Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 86   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 17: 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 public-unknown-numbering 5                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext            Trk       CPN           CPN
Len Code           Grp(s)    Prefix        Len
                                                    Total Administered: 1
5  1029          86        312626         11          Maximum Entries: 9999
```

Figure 18: Public Unknown Numbering Form

4.1.7. Incoming Call Routing

Use the **change inc-call-handling-trmt trunk-group** command to map calls arriving from trunk group "86" from public numbering format to the extensions of the locally attached endpoints shown in **Figure 1**.

```
change change inc-call-handling-trmt trunk-group 86                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    11 312626           6
```

Figure 19: Incoming Call Handling Treatment Form

4.1.8. Configure Codec Sets

Use the **change ip-codec-set** command to designate a codec set to be used for communication with the Vodafone Office Voice SIP trunk. Testing was done with both the G.729B 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.729B" and "G.711A" as the codecs to be used for communication with the Vodafone Office Voice SIP trunk. The Vodafone IMS network supports G.729A, G.729B, and G.711A for incoming calls, and makes outgoing calls with G.729B and G.711A codecs.
FAX Mode (p. 2)	Enter "t.38-standard" to specify that the T.38 standard should be used to transmit FAX documents via the Vodafone 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.729B             n         2        20
 2: G.711A             n         2        20
```

Figure 20: 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 21: IP Codec Set Form, p.2

4.1.9. Configure IP Network Region

Use the **change ip-network-region <x>** command to designate a network region to be used for the Vodafone 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 20**.

Table 8: IP Network Region Parameters

```
change change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: vodafone.nl
    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 22: IP Network Region Form, p.2

4.1.10. 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.
EC500 (p. 4)	Add an EC500 button to activate/deactivate EC500.

Table 9: Station Parameters for IP Telephones

```
add station 10292                                              Page   1 of   5
                                     STATION

Extension: 10292                         Lock Messages? n              BCC: 0
     Type: 9640                          Security Code: 123456          TN: 1
     Port: S00143                      Coverage Path 1:                COR: 1
     Name: extn 10292                  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: 10292
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                  Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? n


                                              Customizable Labels? y
```

Figure 23: Station Form for IP Telephones, page 1

```
add station 10292                                              Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                          Headset? n
       Jack:                                          Speaker? n
      Cable:                                         Mounting: d
      Floor:                                      Cord Length: 0
   Building:                                        Set Color:

ABBREVIATED DIALING
     List1:                   List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                         5:
 2: call-appr                         6:
 3: call-appr                         7:
 4: ec500       Timer? n              8:

    voice-mail Number:
```

Figure 24: Station Form for IP Telephones, page 4

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 Vodafone 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 25: 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 Vodafone 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 26: Class of Service Form

4.1.11. 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 10291                                           Page   1 of   4
                                     STATION

Extension: 10291                         Lock Messages? n              BCC: 0
     Type: 2500                          Security Code:                 TN: 1
     Port: 01A0601                     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 27: Station Form for FAX Device

4.1.12. Configure EC500

Use the **change feature-access-codes** command to allocate feature access codes to control the operation of EC500, using the parameters shown in the following table.

Parameter	Usage
EC500 Self-Administration Access Codes	Enter an unused access code.
Enhanced EC500 Activation	Enter the code which is to be used to activate EC500.
Deactivation	Enter the code which is to be used to deactivate EC500.

Table 13: Station Parameters for FAX Device

```
change feature-access-codes                                     Page   2 of   9
                               FEATURE ACCESS CODE (FAC)
                   Contact Closure  Pulse Code:

                  Data Origination Access Code:
                      Data Privacy Access Code:
              Directed Call Pickup Access Code:
       Directed Group Call Pickup  Access Code:
     Emergency Access to Attendant Access Code:
        EC500 Self-Administration Access Codes: *72
                     Enhanced EC500 Activation: *71    Deactivation: #71
           Enterprise Mobility User Activation:        Deactivation:
  Extended Call Fwd Activate Busy D/A      All:        Deactivation:
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code:
             Group Control Restrict Activation:        Deactivation:
                      Hunt Group Busy Activation:      Deactivation:
                              ISDN Access Code:
                Last Number Dialed Access Code:
     Leave Word Calling Message Retrieval Lock:
   Leave Word Calling Message Retrieval Unlock:
```

Figure 28: Station Form for FAX Device

Enter the **change telecommuting-access** command to specify the extension that is to be dialed from mobile phones to perform EC500 commands.

```
change telecommuting-access                                     Page   1 of   1
                              TELECOMMUTING ACCESS

                     Telecommuting Access Extension: 10299
```

Figure 29: Telecommuting-Access Form

Enter the **change off-pbx-telephone configuration-set** command to specify that "Cellular Voice Mail Detection" is not to be used.

```
change off-pbx-telephone configuration-set 1                    Page   1 of   1



                                     CONFIGURATION SET: 1

                         Configuration Set Description:
                                  Calling Number Style: network
                                   CDR for Origination: phone-number
                    CDR for Calls to EC500 Destination? y
                           Fast Connect on Origination? n
                          Post Connect Dialing Options: dtmf
                         Cellular Voice Mail Detection: none
                                         Barge-in Tone? n
                           Calling Number Verification? y
             Call Appearance Selection for Origination: primary-first
                                     Confirmed Answer? n

 Use Shared Voice Connections for Second Call Answered? n
Use Shared Voice Connections for Second Call Initiated? n
```

Figure 30: Off-Pbx-Telephone Configuration-Set Form

4.2. 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 Vodafone Office Voice SIP trunk. Since Processor Ethernet is used for the tested configuration, the Server Address must be assigned to the Processor Ethernet address of the Avaya S8720 server pair. These values can either be assigned manually to each telephone, or automatically via DHCP.

5. General Test Approach and Test Results

The following issues were encountered during testing:

- For the tested configuration, the Vodafone Office Voice SIP trunk was configured to use the G.729A codec. However, it sends Session Description Protocol (SDP) records which contain media type 18 (G.729) which do not contain an "annex b" specification, thus implying the default codec of G.729B (**see [6]**). For this reason, Avaya Aura Communication Manager must be configured for the G.729B codec.
- Calls to local extensions which are forwarded via the Vodafone Office Voice SIP trunk will show a configurable administrative number (which is common for all local extensions) or will show the number of the caller as the calling party number. The number which is shown as calling party number depends on the provider of the destination.

6. 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 Vodafone Office Voice SIP trunk.

```
status signaling-group 86
                         STATUS SIGNALING GROUP

        Group ID: 86                               Active NCA-TSC Count: 0
      Group Type: sip                               Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

Figure 31: 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.

7. Conclusion

These Application Notes contain instructions for configuring Avaya Aura Communication Manager to connect to the Vodafone Office Voice SIP trunk. All test cases passed with exceptions noted in **Section 5**.

8. 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 8720 Processor Ethernet interface and the Vodafone Office Voice IMS network.

Incoming call:

```
Session Initiation Protocol
    Request-Line: INVITE sip:+31262610294@vodafone.nl;transport=udp;user=phone SIP/2.0
    Message Header
        Via: SIP/2.0/TCP 62.140.143.142:5060;branch=z9hG4bKr3lnni105891le46b7g0.1
        To: <sip:+31262610294@vodafone.nl;transport=UDP>
        From: <sip:+4969XXXX@vodafone.nl>;tag=SDie12001-t672bd
        Call-ID: SDie12001-37cabe839a593d07b64d2e4a56d04b1a-vrvvfv3
        CSeq: 2 INVITE
        Max-Forwards: 66
        Content-Length: 213
        Contact: <sip:jNetX@62.140.143.142:5060;transport=tcp>
        Content-Type: application/sdp
        Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, PRACK
        Accept: application/sdp
        Supported: 100rel
        P-Asserted-Identity: <sip:+49697XXXX@vodafone.nl:5060;user=phone>
        Session-Expires: 1800
        Min-SE: 120
        Route: <sip:+31262610294;tgrp=pbx1;trunk-
context=vtc2@VTCenterprisePBX3i:5060;user=phone;tgrp=pbx1;trunk-context=vtc2;lr;transport=tcp>
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1252047175 1252047175 IN IP4 62.140.143.142
            Session Name (s): Basic Session
            Connection Information (c): IN IP4 62.140.143.142
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 10304 RTP/AVP 0 8 18 99 102
            Media Attribute (a): rtpmap:99 telephone-event/8000
            Media Attribute (a): rtpmap:102 G726-32/8000
            Media Attribute (a): ptime:20
```

Outgoing call:

```
Session Initiation Protocol
Request-Line: INVITE sip:004915209160351@62.140.143.142 SIP/2.0
Message Header
    From: "extn 10293" <sip:+31262610293@vodafone.nl>;tag=8046a4b567aede1b4114a99516f00
    To: "004915209160351" <sip:0049152091 6XXXX@62.140.143.142>
    Call-ID: 8046a4b567aede1b5114a99516f00
    CSeq: 1 INVITE
    Max-Forwards: 61
    Route: <sip:62.140.143.142;lr;phase=terminating;transport=tcp>
    Record-Route: <sip:213.XXX.YYY.140;lr;transport=tcp>
    Via: SIP/2.0/TCP 213.XXX.YYY.140;branch=z9hG4bK8046a4b567aede1b6114a99516f00
    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 10293" <sip:+31262610293@213.XXX.YYY.140;transport=tcp>
    Session-Expires: 1800;refresher=uac
    Min-SE: 1800
    P-Asserted-Identity: "extn 10293" <sip:+31262610293@vodafone.nl>
    Accept-Language: en
    P-Charging-Vector: icid-value="AAS:257-b5a446801deae67994a11b36f51"
    Content-Type: application/sdp
    Alert-Info: <cid:internal@invalid.unknown.domain>;avaya-cm-alert-type=internal
    Content-Length: 169
Message body
    Session Description Protocol
        Session Description Protocol Version (v): 0
        Owner/Creator, Session Id (o): - 1 1 IN IP4 213.160.12.140
        Session Name (s): -
        Connection Information (c): IN IP4 213.160.12.133
        Bandwidth Information (b): AS:64
        Time Description, active time (t): 0 0
        Media Description, name and address (m): audio 2216 RTP/AVP 18 127
        Media Attribute (a): rtpmap:18 G729/8000
        Media Attribute (a): rtpmap:127 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.