Avaya Solution & Interoperability Test Lab

Application Notes for H.323 Voice over IP Trunking between Avaya Communication Manager and VoIP Americas Nativevoip VoIP Service - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring an H.323 Voice over IP trunk between Avaya Communication Manager and the VoIP Americas Nativevoip VoIP service to access the Public Switched Telephone Network. The compliance testing covered a subset of IP-to-PSTN gateways in the VoIP Americas infrastructure for H.323 IP trunking. During compliance testing, telephone calls were successfully established over the H.323 IP trunk without shuffling, between the Avaya IP telephones, Avaya digital telephones, and analog telephones and the telephones in the Public Switched Telephone Network. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab with an H.323 IP trunk terminating at the VoIP Americas Network Access Point in Florida.

1. Introduction

These Application Notes describe the procedure for configuring H.323 Voice over IP trunk between Avaya Communication Manager and the VoIP Americas Nativevoip VoIP service to access the Public Switched Telephone Network (PSTN). The testing covered a subset of IP-to-PSTN gateways in the VoIP Americas infrastructure. Nativevoip VoIP service allows new and existing IP PBX enterprise customers to peer with the PSTN in their native VoIP protocols, such as H.323, utilizing an IP trunk in place of traditional analog and digital trunks to connect to the PSTN.

Figure 1 shows the compliance tested network configuration, simulating an enterprise customer site connected via an H.323 IP trunk to the VoIP Americas Nativevoip VoIP service to access the PSTN. The enterprise site consisted of an Avaya S8700 Media Server and an Avaya G650 Media Gateway. The enterprise site supported Avaya IP telephones, Avaya digital telephones, analog telephones, a fax machine and a modem. The PSTN supported analog and digital telephones, a fax machine and a modem. All the IP addresses in the simulated enterprise site were public IP addresses. Avaya S8700 Media Server, the IPSI, CLAN and MEDPRO resources in the Avaya G650 Media Gateway and the Avaya IP telephones were directly connected to an Internet Service Provider network to access Nativevoip VoIP service. VoIP Americas IP-to-PSTN gateway was located in their Network Access Point in Florida. The IP address of VoIP Americas IP to PSTN gateway was also publicly routable. An H.323 IP trunk was established between the Avaya Communication Manager and VoIP Americas IP-to-PSTN gateway.

Shuffling, also called Direct IP-IP Audio Connections, was disabled for the H.323 trunk in Avaya Communication Manager. With this feature enabled, the RTP audio paths of a call over an H.323 trunk are directly established between an Avaya IP telephone and the terminating IP endpoint, such as VoIP Americas IP-to-PSTN gateway, without using the IP media processor (MEDPRO) in the audio path. Since shuffling was disabled, calls between Avaya IP telephones and PSTN telephones required the resources of the Avaya G650 Media Gateway.

Note that the configuration is also applicable to other Avaya Media Servers and Media Gateways. The administration of the infrastructure components in VoIP Americas network is not the focus of these Application Notes and is not described.

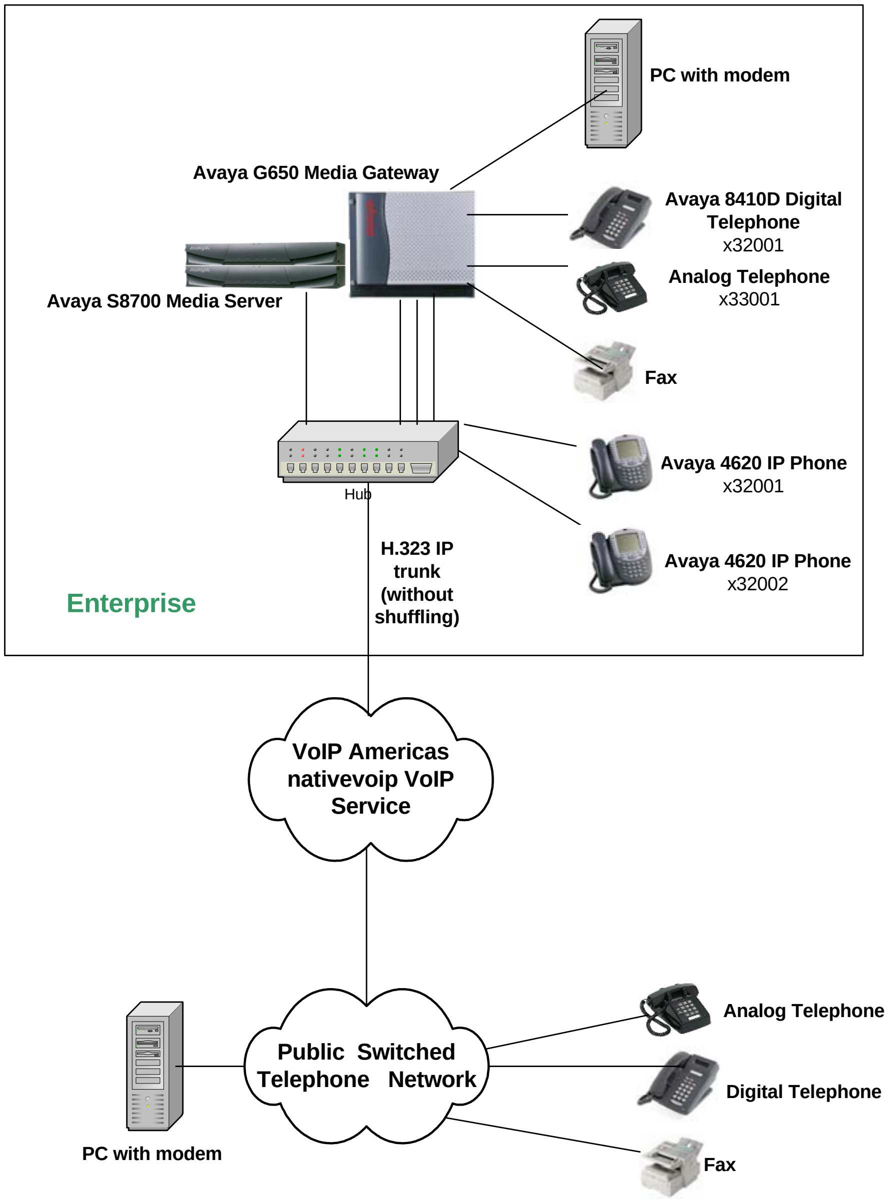

Figure 1: Sample Network Configuration

2. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configurations provided:

Equipment	Software/Firmware
Avaya S8700 Media Server	Avaya Communication Manager 2.2 (R012x.02.0.111.4)
Avaya G650 Media Gateway • TN799DP C-LAN • TN2312AP IPSI • TN2302AP MedPro • TN2224BDigital Line • TN793 Analog Line	 HW11 FW12 HW01 FW12 HW20 FW95 10 6
Avaya 4620 IP Telephones	2.130
Avaya 8410D Digital Telephones	-
Analog Telephones, Fax Machine and Modems	
Avaya P333T-PWR Power Over Ethernet Stackable Switch	4.0.17
VoIP Americas IP to PSTN Gateway for Nativevoip VoIP service	-

3. Configure VoIP Americas IP-to-PSTN gateway for Nativevoip VoIP Service

As a service provider, VoIP Americas is responsible for configuring the relevant IP-to-PSTN gateway at their Network Access Point appropriately to work with Avaya Communication Manager at an enterprise customer site. The following list highlights the information that VoIP Americas administered on their IP-to-PSTN gateway in the compliance tested network configuration. The detailed configuration information is private and is not described here.

- Public IP address of VoIP Americas IP-to-PSTN gateway.
- Public IP address of CLAN card in the Avaya G650 Media Gateway.
- IP codec set – G.711 or G.729 determined by the customer's preference.
- G.711 T.38 fax with failover to pass through as G.711 fax.
- G.729 fax.
- Out-of-band or in-band DTMF determined by the customer's preference.
- DID numbers and dial plan for routing the calls from PSTN to the customer site.

In addition, the PSTN connected to the IP-to-PSTN gateway was validated to carry the non-voice traffic, such as fax.

4. Configure Avaya Communication Manager

This section presents configuration steps for the Avaya S8700 Media Server. Before proceeding, use the command d**isplay system-parameters special-applications** and page forward to Page 4 to verify that **Special Application SA8507** is enabled. SA8507 must be enabled to achieve the interoperability documented in these Application Notes. If SA8507 is not enabled, contact your authorized Avaya sales representative.

```
display system-parameters special-applications                   Page   4 of   5
                             SPECIAL APPLICATIONS

     (SA8481) - Replace Calling Party Number with ASAI ANI? n
               (SA8500) - Expanded UUI Display Information? n
                 (SA8506) - Altura Interoperability (FIPN)? n
                (SA8507) - H245 Support With Other Vendors? y
                (SA8508) - Multiple Emergency Access Codes? n
 (SA8510) - NTT Mapping of ISDN Called-Party Subaddress IE? n
                      (SA8517) - Authorization Code By COR? n
   (SA8518) - Automatic Callback with Called Party Queuing? n
          (SA8520) - Hoteling Application for IP Terminals? n
  (SA8558) - Increase Automatic MWI & VuStats (S8700 only)? n
                 (SA8567) - PHS X-Station Mobility over IP? n
       (SA8569) - No Service Observing Tone Heard by Agent? n
                 (SA8573) - Call xfer via ASAI on CAS Main? n
          (SA8582) - PSA Location and Display Enhancements? n
               (SA8587) - Networked PSA via QSIG Diversion? n
                         (SA8589) - Background BSR Polling? n
                     (SA8601) - Two-Digit AUX Reason Codes? n
     (SA8608) - Increase Crisis Alert Buttons (S8700 only)? n
                      (SA8621) - SCH Feature Enhancements? n
```

4.1. Configuring VoIP Attributes

This section illustrates the parameters used in the administration of the H.323 Signaling Group and IP Trunk Group. See Section 4.2 for their relevant usage.

4.1.1. IP Audio Codec Set

Administer the desired audio codec – G.711 or G.729 – using the IP Codec Set form. To specify the codecs, enter **change ip-codec-set p** using the System Access Terminal (SAT), where **p** is the number of a codec set, and modify the IP Codec Set form accordingly. The default settings are shown below:

```
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.711MU         n         2        20
 2:
```

To enable T.38 fax routing, page forward to Page 2 and set the **FAX** to **t.38-standard,** as illustrated in the following example. Note that T.38 fax requires the TN2302AP Media Processor with version HW10 or greater and FW95 or greater. (For regular fax routing, page forward to Page 2 and set the **FAX** to **off**. This will treat the fax as an ordinary voice call).

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

                          IP Codec Set



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

For modem calls, keep the default setting of **Modem** as **off.** The modem calls will be treated as voice calls.

4.1.2. IP Network Region

The most relevant attribute on IP Network Region form related to this application is the **Codec Set**. The IP Network Region is used to obtain the codec set used for negotiation of trunk bearer capability.

To configure the IP network region, enter **change ip-network-region m** using the SAT, where **m** is the number of the region. On Page 1 of the IP Network Region form, modify the **Codec Set** to the number of the codec set that will be used in this region. The following example illustrates that codec set 1 will be used for IP network region 1. Note that the Shuffling, also called Direct IP to IP Audio Connections, is disabled on the Signaling form – see Section 4.2.2.

```
change ip-network-region 1                                      Page   1 of 19
                               IP NETWORK REGION
  Region: 1
Location:                  Home Domain:
    Name:
                                Intra-region IP-IP Direct Audio: yes
AUDIO PARAMETERS                Inter-region IP-IP Direct Audio: yes
   Codec Set: 1                            IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3049                       RTCP Reporting Enabled? n

DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 34
        Audio PHB Value: 46
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      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
```

4.2. Configuring H.323 IP Trunk between Avaya Communication Manager and VoIP Americas

This section illustrates the parameters used in the administration of the H.323 Signaling Group and IP Trunk Group.

4.2.1. IP Node Names

The following illustrates a subset of the IP Node Names screen that maps logical names to IP addresses. These node names are presented because they will appear in other screens, such as the screen defining the H.323 signaling group. Note that the IP addresses for CLAN, MedPro and VoIPAmerias in the compliance tested configuration were administered as public IP addresses (not shown here).

```
change node-names ip                                            Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address            Name              IP Address
CLAN                                                            .   .   .
MedPro                                                          .   .   .
VoIPAmericas                                                    .   .   .
default             0   .0   .0   .0                            .   .   .
procr         . . .              . . .
```

4.2.2. Signaling Group

Administer a signaling group by entering **change signaling-group n,** where n is the number of the signaling group number.

- **Group Type**: Enter **h.323** for **Group Type**.
- **Node names and the listen ports:** The **CLAN** is the near-end of the signaling group. The far-end is set to **VoIPAmericas** (the node name of the VoIP Americas IP to PSTN gateway). Retain the default near-end listen port (**1720**) and enter **1720** as the far-end listen port. In general, the **Far-end Network Region** field can be left blank, or it can be populated with a network region number. In the compliance-tested configuration, the **Far-end Network Region** field is set to **1**.
- **Direct IP-IP Audio Connections –** This field must be set to **n** to disable shuffling, or interoperability problems will be experienced. Since shuffling must remain disabled for the signaling group, calls between Avaya IP telephones and PSTN telephones will require the media processor resources of the Avaya G650 Media Gateway.
- **DTMF over IP**: Set this field to either **out-of-band** or **in-band**. Note that in-band DTMF is relevant to G.711 calls only. This setting must match the DTMF setting in VoIP Americas IP to PSTN gateway (see Section 3).

The following example shows how to add a signaling group at the main site.

```
add signaling-group 1                                           Page   1 of  5
                                SIGNALING GROUP

 Group Number: 1              Group Type: h.323
                           Remote Office? n         Max number of NCA TSC: 0
                                     SBS? n          Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection:
          Supplementary Service Protocol: a         Network Call Transfer? n
                         T303 Timer(sec): 10

        Near-end Node Name: CLAN              Far-end Node Name: VoIPAmericas
      Near-end Listen Port: 1720            Far-end Listen Port: 1720
                                         Far-end Network Region: 1
            LRQ Required? n               Calls Share IP Signaling Connection? n
            RRQ Required? n                     H245 Control Addr On FACility? n
        Media Encryption? n                 Bypass If IP Threshold Exceeded? n

            DTMF over IP: out-of-band          Direct IP-IP Audio Connections? n
                                                         IP Audio Hairpinning? n
                                               Interworking Message: PROGress
```

4.2.3. Trunk Group

Configure an H.323 IP trunk. Enter **add trunk-group n**, where n is the trunk group number. Administer the trunk group parameters, with the following settings

- **Group Type:** Enter **isdn**.
- **TAC:** Enter a trunk access code, such as **910**, according to the dial plan.
- **Carrier Medium:** Enter **IP**.
- **Service Type:** Enter **tie** to set this as an IP tie trunk between the two servers.

```
add trunk-group 10                                                         Page   1 of  22
                                TRUNK GROUP

Group Number: 10                  Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL                 COR: 1       TN: 1        TAC: 910
   Direction: two-way        Outgoing Display? y         Carrier Medium: IP
 Dial Access? y                Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n             TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: ascend
                                                   Digital Loss Group: 18
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200        Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

4.2.4. Trunk Group Members

Configure the trunk group members on Page 6 of the trunk-group form, by setting the port to **IP** and the previously configured signaling group. After submitting the form, the port field values are changed as shown below.

```
display trunk-group 10                                                     Page   6 of 22
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/5
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   5

       Port    Code Sfx Name        Night           Sig Grp
  1: T00207                                            1
  2: T00185                                            1
  3: T00186                                            1
  4: T00187                                            1
  5: T00188                                            1
```

4.2.5. Associate Signaling Group to an IP trunk Group

The signaling group is associated with the IP Trunk Group. Using the command **change signaling-group 1**, enter the number **10** in the **Trunk Group for Channel Selection** field.

```
change signaling-group 1                                        Page   1 of  5
                                SIGNALING GROUP

 Group Number: 1             Group Type: h.323
                          Remote Office? n         Max number of NCA TSC: 0
                                    SBS? n          Max number of CA TSC: 0
                                                 Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 10
          Supplementary Service Protocol: a         Network Call Transfer? n
                         T303 Timer(sec): 10

        Near-end Node Name: CLAN             Far-end Node Name: VoIPAmericas
      Near-end Listen Port: 1720           Far-end Listen Port: 1720
                                        Far-end Network Region: 1
              LRQ Required? n            Calls Share IP Signaling Connection? n
              RRQ Required? n                  H245 Control Addr On FACility? n
          Media Encryption? n              Bypass If IP Threshold Exceeded? n

              DTMF over IP: out-of-band       Direct IP-IP Audio Connections? n
                                                       IP Audio Hairpinning? n
                       Interworking Message: PROGress
```

5. Interoperability Compliance Testing

The interoperability compliance testing focused on assessing H.323 Voice over IP trunking (without shuffling) between Avaya Communication Manager and VoIP Americas IP to PSTN gateway to provide VoIP Americas Nativevoip VoIP service. An H.323 IP trunk was established between the Avaya Communication Manager at a simulated enterprise site and VoIP Americas IP to PSTN gateway in Florida. Shuffling of audio path over the H.323 trunk was disabled. Avaya Communication Manager was configured to route inbound and outbound calls over the H.323 IP trunk. VoIP Americas IP to PSTN gateway was configured to route the inbound calls from PSTN to Avaya Communication Manager, and route the outbound calls from Avaya Communication Manager to the PSTN.

5.1. General Test Approach

The general approach was to establish calls between Avaya IP telephones, Avaya digital telephones, and analog telephones and the telephones in the Public Switched Telephone Network. IP addresses for all the Avaya Communication Manager resources, such as IPSI, CLAN and MEDPRO cards, Avaya IP telephones and the VoIP Americas IP to PSTN gateway were public IP addresses. The main objectives were to verify that:

- An H.323 IP trunk can be established between Avaya Communication Manager at a simulated enterprise site and VoIP Americas IP to PSTN gateway in Florida, without shuffling, to provide VoIP Americas Nativevoip VoIP service to the enterprise customers (Nativevoip VoIP trunk).

- Outbound voice calls from Avaya IP telephones, Avaya digital telephones, and analog telephones via Nativevoip VoIP trunk can be routed to the telephones in the Public Switched Telephone Network.
- Inbound voice calls from telephones in Public Switched Telephone Network can be routed to Avaya IP telephones, Avaya digital telephones, and analog telephones via Nativevoip VoIP trunk.
- Calling number is displayed on the telephones is displayed, where applicable.
- Inbound and outbound voice calls can be established using G.711 and G.729 codec sets.
- DTMF can be transmitted from an Avaya Communication Manager telephone towards the PSTN and vice versa.
- For G.711 and G.729 codecs, inbound and outbound fax can be transmitted using the T.38 standard protocol.
- For G.711 codec, outbound modem calls can be established.

5.2. Test Results

All the tests, outlined as the main objectives in Section 5.1, completed successfully. Nativevoip VoIP trunk, without shuffling, was successfully established. Voice, fax and modem calls and DTMF transmission for the calls using VoIP Americas Nativevoip VoIP service established with good quality.

6. Verification Steps

The following steps may be used to verify the configuration:

- Establish, maintain, and tear down inbound and outbound voice, fax and modem calls over the Nativevoip VoIP trunk, as outlined in Section 5.1. Verify the voice quality is good and the DTMF are transmitted. Verify that the T.38 fax and modem calls are successful.
- Make simultaneous inbound and outbound calls and verify that voice quality is acceptable.

7. Support

For technical support on the VoIP Americas Nativevoip VoIP service, contact support@voipamericas.com or contact VoIP Americas Support at 1.305.667.3473.

8. Conclusion

As illustrated in these Application Notes, using H.323 trunks Avaya Communication Manager can interoperate with VoIP Americas Nativevoip VoIP service to access the PSTN. For successful interoperability with the VoIP Americas NativevoipVoIP service, the main feature settings in Avaya Communication Manager are as follows. First, Special Application SA8507 must be enabled. Second, Direct IP-IP audio connections, often referred to as “shuffling”, must be disabled on the Avaya H.323 Signaling Group.

9. Additional References

The following documents are relevant to these Application Notes:

1) *Administrator's Guide for Avaya Communication Manager,* January 2005, Document Number 555-233-506.
2) *Administration for Network Connectivity for Avaya Communication Manager,* January 2005, Document Number 555-233-504.

Additional product documentation for Avaya products may be found at http://support.avaya.com.

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 Developer*Connection* Program at devconnect@avaya.com.