Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Newfound Communications IP Call Recorder with Avaya SIP Enablement Services and Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the Newfound Communications IP Call Recorder to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager.

The Newfound Communications IP Call Recorder is a SIP-based IP call recording solution that provides recording capabilities to VoiceXML applications running on a VoiceXML platform. The IP Call Recorder runs on the Newfound VoIP Media Gateway (VMG). A VoiceXML platform is provided as part of the VMG or a customer provided platform could be used. The function of the VoiceXML application is customer-specific but often provides an IVR menu for inbound callers to the enterprise. For the purposes of the compliance test, a demo VoiceXML IVR application provided by Newfound was used to exercise specific SIP call flows and recording capabilities. The IP Call Recorder can record any portion of an active call even after it is transferred to another party.

Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the procedure for configuring the Newfound Communications IP Call Recorder to interoperate with Avaya SIP Enablement Services (SES) and Avaya Communication Manager.

The Newfound Communications IP Call Recorder is a SIP-based IP call recording solution that provides recording capabilities to VoiceXML applications running on a VoiceXML platform. The IP Call Recorder runs on the Newfound VoIP Media Gateway (VMG). A VoiceXML platform is provided as part of the VMG or a customer provided platform could be used. The function of the VoiceXML application is customer-specific but often provides an IVR menu for inbound callers to the enterprise. For the purposes of the compliance test, a demo VoiceXML IVR application provided by Newfound was used to exercise specific SIP call flows and recording capabilities. The IP Call Recorder can record any portion of an active call even after it is transferred to another party.

The IP Call Recorder has two modes for recording calls. The first mode known as full call recording will record all calls all the time. It is enabled through configuration. The second mode is known as ad-hoc recording which allows the VoiceXML application to control the starting and stopping of the recording. Any portion of a call can be recorded and depending on the format used, one direction of the call can be recorded on the right stereo channel and the other direction can be recorded on the left stereo channel. This allows for easier editing of the individual sides of the call. The supported recording formats are listed in **Appendix A**.

1.1. Configuration

Figure 1 illustrates the configuration used in these Application Notes. The sample configuration shows an enterprise with an Avaya SES and an Avaya S8300 Media Server running Avaya Communication Manager in an Avaya G700 Media Gateway. There are two Newfound VoIP Media Gateways (VMGs) for redundancy. Each VMG contains the same set of software components including the IP Call Recorder, VoiceXML platform and VoiceXML application. The Avaya SES will send calls to the redundant VMG if the primary VMG is unavailable. Endpoints include four Avaya 4600 Series IP Telephones (with SIP firmware), an Avaya one-X Desktop Edition, an Avaya 6408D Digital Telephone, and an Avaya 6210 Analog Telephone. The Avaya SIP Telephones download the settings file from the TFTP server located on site. An ISDN-PRI trunk connects the Avaya G700 Media Gateway to the PSTN.

The VMG does not register with the Avaya SES as an endpoint but instead is configured as a trusted host. Address Maps are configured on the Avaya SES to route calls between the Avaya SES and the VMG. In the case of redundant VMGs, two contacts are added to the Host Address Map on the Avaya SES; one for the primary VMG and one for the secondary VMG.

For interoperability, the IP Call Recorder requires the use of the G.711mu codec, and transmission of DTMF tones using RFC2833. In addition, the Direct IP - IP Audio feature (also know as media shuffling) must be disabled. This is due to an incompatibility in the way this feature is implemented between the Newfound and Avaya products.

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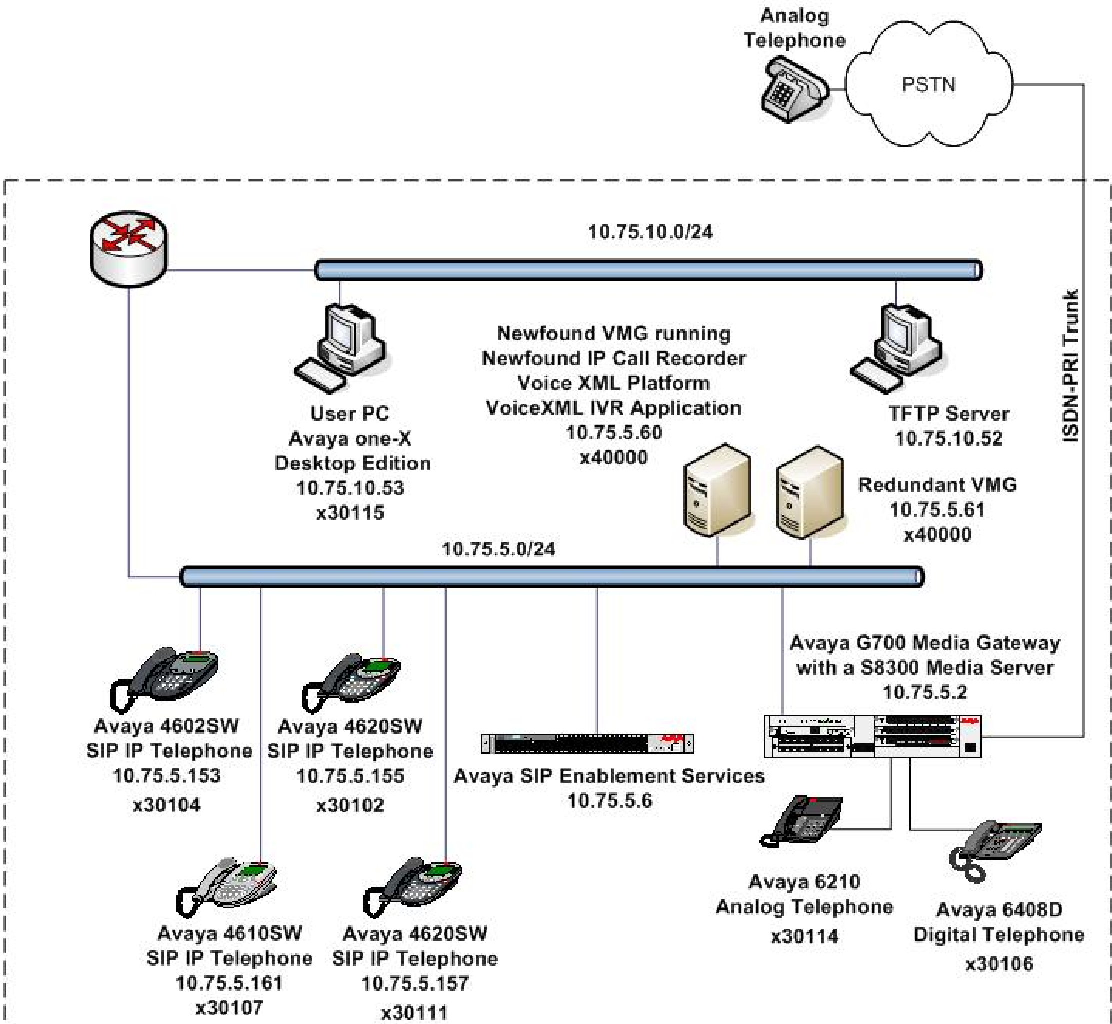

Figure 1: IP Call Recorder Test Configuration

1.2. Operation

To better understand the relationship between the components in the IP Call Recorder solution, **Figure 2** shows the call flow for a typical inbound call from the PSTN to the VoiceXML application. A caller on the PSTN dials a PSTN number that is associated with the enterprise IVR (the demo VoiceXML application). The call arrives at Avaya Communication Manager across the ISDN-PRI trunk from the PSTN (1). Avaya Communication Manager maps the incoming PSTN number to a 5-digit string (40000) that will be used by Automatic Alternate Routing (AAR) to route the call to the SIP trunk between Avaya Communication Manager and the Avaya SES (2). The Avaya SES forwards the call to the VMG (3). The VMG acts as a SIP back-to-back user agent (B2BUA) and initiates a new SIP call to the VoiceXML platform in response to the call that was received from Avaya Communication Manager (4). The VoiceXML platform then uses the dialed digits (40000) received on the call to load the proper VoiceXML application (5). The RTP media flows between the Avaya Media Gateway and the VoiceXML application through the VMG (6). In this manner, the IP Call Recorder has access to the media stream for recording. At this point, the PSTN caller is connected to the demo VoiceXML application serving as an IVR which can play prompts or accept input from the caller (7). The VoiceXML application can also invoke the

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recording or playback of any portion of the call by sending Extensible Markup Language Remote Procedure Call (XML RPC) events to the IP Call Recorder residing on the VMG (8).

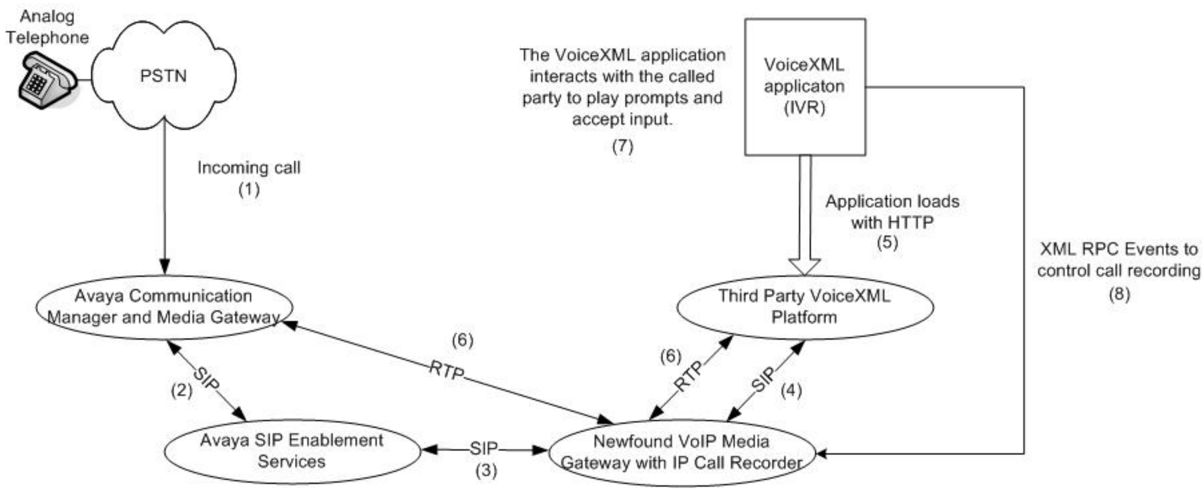

Figure 2: Incoming Call Flow

The demo Voice-XML application also allows the call shown in **Figure 2** to be transferred to another endpoint while still allowing the IP Call Recorder to record the call. **Figure 3** shows the high-level call flow for the blind transfer scenario from the point the demo Voice-XML application transfers the call. In this case, the demo Voice-XML application initiates a new SIP signaling connection directly to the Avaya SES, bypassing the VMG, using a different IP source port (1). The Avaya SES routes the call to Avaya Communication Manager (2). Avaya Communication Manager acts as a SIP B2BUA and initiates a new SIP call to the final endpoint via the Avaya SES (2). The Avaya SES routes the call to the SIP endpoint (3). The RTP media stream continues to flow through the VMG for recording. Since media shuffling is disabled, media also continues to flow through the Avaya Media Gateway, even for SIP endpoints (4). In summary, media flows from the PSTN to the Avaya Media Gateway to the VMG back to the Avaya Media Gateway and finally to the SIP telephone.

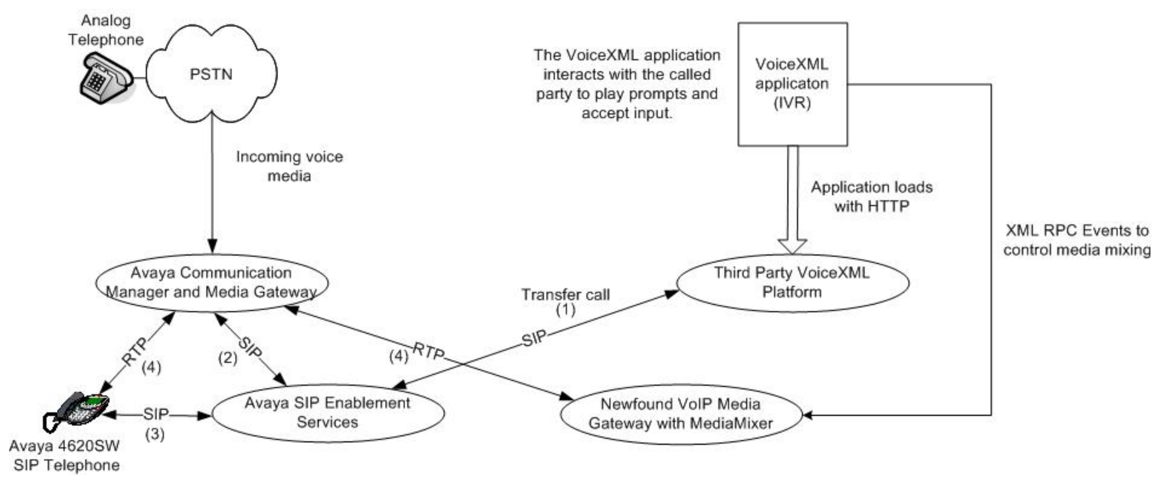

Figure 3: Blind Transfer Call Flow

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2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300 Media Server with Avaya G700 Media Gateway	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1) and Service Pack 01.2.632.1-12866
Avaya SIP Enablement Services (SES)	3.1.1
Avaya 4602SW IP Telephones Avaya 4610SW IP Telephones Avaya 4620SW IP Telephones	SIP version 2.2.2
Avaya one-X Desktop Edition	2.1
Avaya 6408D Digital Telephone	-
Analog Telephones	-
Windows PCs	Windows XP Professional
Newfound VoIP Media Gateway (including VoiceXML Platform)	1.4
Newfound IP Call Recorder	1.0
VoiceXML Demo IVR Application	-

3. Configure Avaya Communication Manager

The following configuration of Avaya Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration in this section, perform a **save translation** command to make the changes permanent.

The communication between Avaya Communication Manager and Avaya SES is via a SIP trunk group. All SIP signaling for calls between Avaya Communication Manager and the VMG passes through Avaya SES via this trunk group. This section describes the steps for configuring this trunk group and associated signaling group. For more information on configuring Avaya Communication Manager to support SIP, please refer to [3].

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Step	Description
1.	Use the **display system-parameters customer-options** command to verify that sufficient SIP trunk capacity exists. On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk. The license file installed on the system controls the maximum permitted. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.
2.	On Page 4, verify that the features shown in bold in the example below are enabled.

Screen for step 1:

```
display system-parameters customer-options                        Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100   10
           Maximum Concurrently Registered IP Stations: 20    0
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   24

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 0     0
                    Maximum G250/G350/G700 VAL Sources: 5     1
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 10    0

        (NOTE: You must logoff & login to effect the permission changes.)
```

Screen for step 2:

```
display system-parameters customer-options                        Page   4 of  10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                              IP Stations? y
           Enable 'dadmin' Login? y               Internet Protocol (IP) PNC? n
           Enhanced Conferencing? y                       ISDN Feature Plus? n
                  Enhanced EC500? y            ISDN 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? n                     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 Appl. Server Interface (MASI)? n
             Hospitality (Basic)? y          Multimedia Call Handling (Basic)? n
 Hospitality (G3V3 Enhancements)? n       Multimedia Call Handling (Enhanced)? n
                      IP Trunks? y
```


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<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>
On Page 5, verify that the features shown in bold in the example below are enabled.

```
display system-parameters customer-options                      Page   5 of  10
                                OPTIONAL FEATURES

                Multinational Locations? n            Station and Trunk MSP? n
 Multiple Level Precedence & Preemption? n     Station as Virtual Extension? n
                     Multiple Locations? n
                                             System Management Data Transfer? n
          Personal Station Access (PSA)? n              Tenant Partitioning? n
                        Posted Messages? n       Terminal Trans. Init. (TTI)? n
                        PNC Duplication? n              Time of Day Routing? n
                   Port Network Support? n             Uniform Dialing Plan? n
                                               Usage Allocation Enhancements? y
              Processor and System MSP? n       TN2501 VAL Maximum Capacity? y
                     Private Networking? y
                     Processor Ethernet? y              Wideband Switching? n
                                                                  Wireless? n
                          Remote Office? n
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```
</td></tr>
<tr><td>4.</td><td>
Use the change node-names ip command to assign the node name and IP address for the Avaya SES. In this case, SES and 10.75.5.6 are being used, respectively. The node name SES will be used throughout the other configuration forms of Avaya Communication Manager. In this example, procr and 10.75.5.2 are the name and IP address assigned to the Avaya S8300 Media Server.

```
change node-names ip                                            Page   1 of   1
                                  IP NODE NAMES
    Name              IP Address           Name               IP Address
SES                 10 .75 .5  .6                              .   .   .
default             0  .0  .0  .0                              .   .   .
myaudix             10 .75 .5  .7                              .   .   .
procr               10 .75 .5  .2                              .   .   .
```
</td></tr>
</table>

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Step	Description
5.	Use the **change ip-network-region *n*** command, where ***n*** is the number of the region to be changed, to define the connectivity settings for all VoIP resources and IP endpoints within the region. Select an IP network region that will contain the Avaya SES server. The association between this IP network region and the Avaya SES server will be done on the **Signaling Group** form as shown in Step 7. In the case of the compliance test, the same IP network region that contains the Avaya S8300 Media Server and Avaya IP Telephones was selected to contain the Avaya SES server. By default, the Media Server and IP telephones are in IP Network Region 1. On the **IP Network Region** form: ▪ The **Authoritative Domain** field is configured to match the domain name configured on Avaya SES. In this configuration, the domain name is ***business.com***. This name will appear in the "From" header of SIP messages originating from this IP region. ▪ By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G700 Media Gateway. This is true for both intra-region and inter-region IP-IP Direct Audio. Shuffling can be further restricted at the trunk level on the **Signaling Group** form. For interoperability with the IP Call Recorder, shuffling will be disabled on the **Signaling Group** form (Step 7) for the SIP trunk. This will still allow shuffling to occur between IP endpoints other than SIP endpoints. ▪ The **Codec Set** is set to the number of the IP codec set to be used for calls within this IP network region. If different IP network regions are used for the Avaya S8300 Media Server and the Avaya SES server, then Page 3 of each **IP Network Region** form must be used to specify the codec set for inter-region communications. ▪ The default values can be used for all other fields.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
Region: 1
Location:         Authoritative Domain: business.com
    Name:
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: 3327
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
```


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Step	Description

6.	Use the **change ip-codec-set *n*** command, where ***n*** is the codec set value specified in Step 5, to enter the supported audio codecs for calls routed to Avaya SES. Multiple codecs can be listed in priority order to allow the codec to be negotiated during call establishment. The list should include the codecs the enterprise wishes to support within the normal trade-off of bandwidth versus voice quality. The example below shows the values used in the compliance test. For interoperability with the IP Call Recorder, **G.711MU** must be included in the list.
	``
	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: G.729AB n 2 20
	 3:
	``

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Step	Description
7.	Use the **add signaling-group *n*** command, where ***n*** is the number of an unused signaling group, to create the SIP signaling group as follows: ▪ Set the **Group Type** field to ***sip***. ▪ The **Transport Method** field will default to ***tls*** (Transport Layer Security). TLS is the only link protocol that is supported for communication between Avaya SES and Avaya Communication Manager. ▪ Specify the Avaya S8300 Media Server (node name ***procr***) and the Avaya SES Server (node name ***SES***) as the two ends of the signaling group in the **Near-end Node Name** and the **Far-end Node Name** fields, respectively. These field values are taken from the **IP Node Names** form shown in Step 4. For alternative configurations that use a C-LAN board, the near (local) end of the SIP signaling group will be the C-LAN board instead of the Media Server. ▪ Ensure that the recommended TLS port value of ***5061*** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. ▪ In the **Far-end Network Region** field, enter the IP network region value assigned in the **IP Network Region** form in Step 5. This defines which IP network region contains the Avaya SES server. If the **Far-end Network Region** field is different from the near-end network region, the preferred codec will be selected from the IP codec set assigned for the inter-region connectivity for the pair of network regions. ▪ Enter the domain name of Avaya SES in the **Far-end Domain** field. In this configuration, the domain name is ***business.com***. This domain is specified in the Uniform Resource Identifier (URI) of the SIP "To" header in the INVITE message. ▪ The **Direct IP-IP Audio Connections** field is set to ***n***. This disables shuffling on the SIP trunk to the Avaya SES. This is required for interoperability with the IP Call Recorder. ▪ The **DTMF over IP** field must be set to the default value of ***rtp-payload*** for a SIP trunk. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833. This is required by the IP Call Recorder. ▪ The default values for the other fields may be used.

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

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




   Near-end Node Name: procr                  Far-end Node Name: SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
       Far-end Domain: business.com

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? n
                                                        IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```


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<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>8.</td><td>
Add a SIP trunk group by using the add trunk-group n command, where n is the number of an unused trunk group. For the compliance test, trunk group number 1 was chosen.

On Page 1, set the fields to the following values:

Set the Group Type field to sip.
Choose a descriptive Group Name.
Specify an available trunk access code (TAC) that is consistent with the existing dial plan.
Set the Service Type field to tie.
Specify the signaling group associated with this trunk group in the Signaling Group field as previously specified in Step 7.
Specify the Number of Members supported by this SIP trunk group. As mentioned earlier, each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk.
The default values may be retained for the other fields.

<pre>
add trunk-group 1   Page   1 of 21
                                TRUNK GROUP

Group Number: 1   Group Type: sip   CDR Reports: y
  Group Name: SES Trk Grp   COR: 1   TN: 1   TAC: 101
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n

                                                      Signaling Group: 1
                                                    Number of Members: 24
</pre>
</td></tr>
<tr><td>9.</td><td>
On Page 3:

Verify the Numbering Format field is set to public. This field specifies the format of the calling party number sent to the far-end.
The default values may be retained for the other fields.

<pre>
add trunk-group 1   Page   3 of 21
TRUNK FEATURES
          ACA Assignment? n   Measured: none
                                                         Maintenance Tests? y

                     Numbering Format: public
                                                 Prepend '+' to Calling Number? n

                                                   Replace Unavailable Numbers? n
</pre>
</td></tr>
</table>

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Step	Description
10.	Use the **change public-unknown-numbering 0** command to define the full calling party number to be sent to the far-end. Add an entry for the trunk group defined in Step 8. In the example shown below, all calls originating from a 5-digit extension beginning with 3 and routed across trunk group 1 will be sent as a 5 digit calling number. This calling party number will be sent to the far-end in the SIP "From" header.
11.	As mentioned in Section 1.2, AAR was used by Avaya Communication Manager to route calls to the Avaya SES that were bound for the demo Voice-XML application. AAR requires a route pattern for this purpose that points to the SIP trunk created in Step 8. Create a route pattern that will use the SIP trunk that connects to Avaya SES. The compliance test defined route pattern 1 as the route for all outbound calls. For more information on AAR see [1] and [2]. To create a route pattern, use the **change route-pattern *n*** command, where ***n*** is the number of an unused route pattern. Enter a descriptive name for the **Pattern Name** field. Set the **Grp No** field to the trunk group number created for the SIP trunk. Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level. The default values may be retained for all other fields.

Step 10 screen:

```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                   Total                                  Total
Ext Ext    Trk       CPN            CPN Ext Ext    Trk       CPN           CPN
Len Code   Grp(s)    Prefix         Len Len Code   Grp(s)    Prefix        Len

 5 3       1                         5
 5 3       99                        5
```

Step 11 screen:

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: SIP
                             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                                                              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 3 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
```


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Step	Description

12.	Use the **change locations** command to assign the default SIP route pattern to the location. In the compliance test, all SIP endpoints at the enterprise are part of a single location defined in Avaya Communication Manager. This location uses the default name of ***Main*** and is shown in the example below. Enter the route pattern number from the previous step in the **Proxy Sel. Rte. Pat.** field. The default values may be retained for all other fields.
	
	`change locations                                             Page   1 of   4`
	`                                   LOCATIONS`
	
	`                ARS Prefix 1 Required For 10-Digit NANP Calls? y`
	
	` Loc.  Name           Timezone Rule  NPA  ARS   Attd         Pre-   Proxy Sel.`
	` No.                    Offset               FAC   FAC          fix    Rte. Pat.`
	` 1:   Main            + 00:00   0                                    1`
	` 2:`
	` 3:`

13.	To map a PSTN number to the number for the demo VoiceXML application at the enterprise, use the **change inc-call-handling-trmt trunk-group *n*** command, where ***n*** is the ISDN-PRI trunk group number connected to the PSTN. For the compliance test, trunk group 2 was used for the ISDN-PRI trunk to the PSTN. The example below shows an incoming 11-digit number being deleted and replaced with the number associated with the VoiceXML application.
	
	`change inc-call-handling-trmt trunk-group 2                    Page   1 of   3`
	`                       INCOMING CALL HANDLING TREATMENT`
	` Service/      Called    Called        Del  Insert`
	` Feature        Len      Number`
	` tie             10  17325551234       11   40000`

14.	Add an entry in the dial plan that defines that any 5-digit string beginning with a 4 will be routed by AAR. To do this, use the **change dialplan analysis** command and add the entry highlighted below.
	
	`change dialplan analysis                                     Page   1 of  12`
	`                             DIAL PLAN ANALYSIS TABLE`
	`                                                         Percent Full:    3`
	
	`      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call`
	`      String  Length Type      String  Length Type      String  Length Type`
	`        1       3    dac`
	`        3       5    ext`
	`        4       5    aar`
	`        8       1    fac`
	`        9       1    fac`
	`        *       3    fac`
	`        #       3    fac`

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Step	Description
15.	Create an entry in the **AAR Digit Analysis Table** to route dialed digits of 40000 to route pattern 1 that points to the SIP trunk connected to the Avaya SES (see Step 11). To do this, use the **change aar analysis *n*** command, where ***n*** is a set of dialed digits in the table. A portion of the table will be displayed starting at ***n***. Tab to the bottom of the displayed entries and create a new entry in the table like the one highlighted below.

```
change aar analysis 1                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                         Percent Full:    3

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    2                     7    7     254      aar          n
    3                     4    4     4        aar          n
    39001                 5    5     99       aar          n
    40000                 5    5     1        aar          n
    5                     7    7     254      aar          n
    6                     7    7     254      aar          n
    7                     7    7     254      aar          n
    8                     7    7     254      aar          n
    9                     7    7     254      aar          n
```


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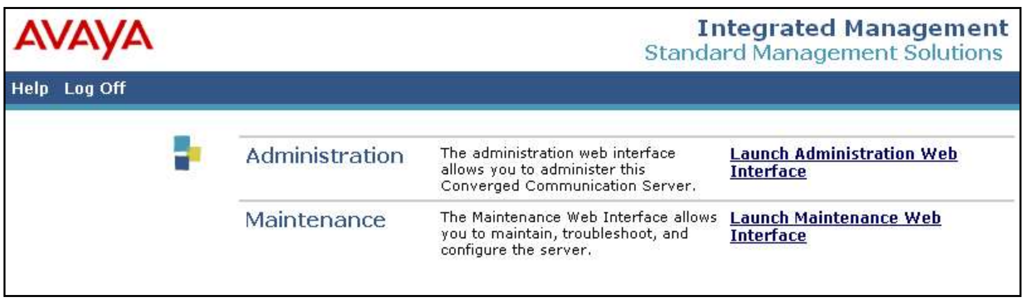
4. Configure Avaya SES

This section covers the configuration of Avaya SES. Avaya SES is configured via an Internet browser using the administration web interface. It is assumed that Avaya SES software and the license file have already been installed on the server. During the software installation, the installation script is run from the Linux shell of the server to specify the IP network properties of the server along with other parameters. For additional information on these installation tasks, refer to [4].

Step	Description
1.	Access the Avaya SES administration web interface by entering http://<*ip-addr*>/admin as the URL in a Web browser, where <*ip-addr*> is the IP address of the Avaya SES server. Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown below. 

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Step	Description
2.	The Avaya SES Administration Home Page will be displayed as shown below. 

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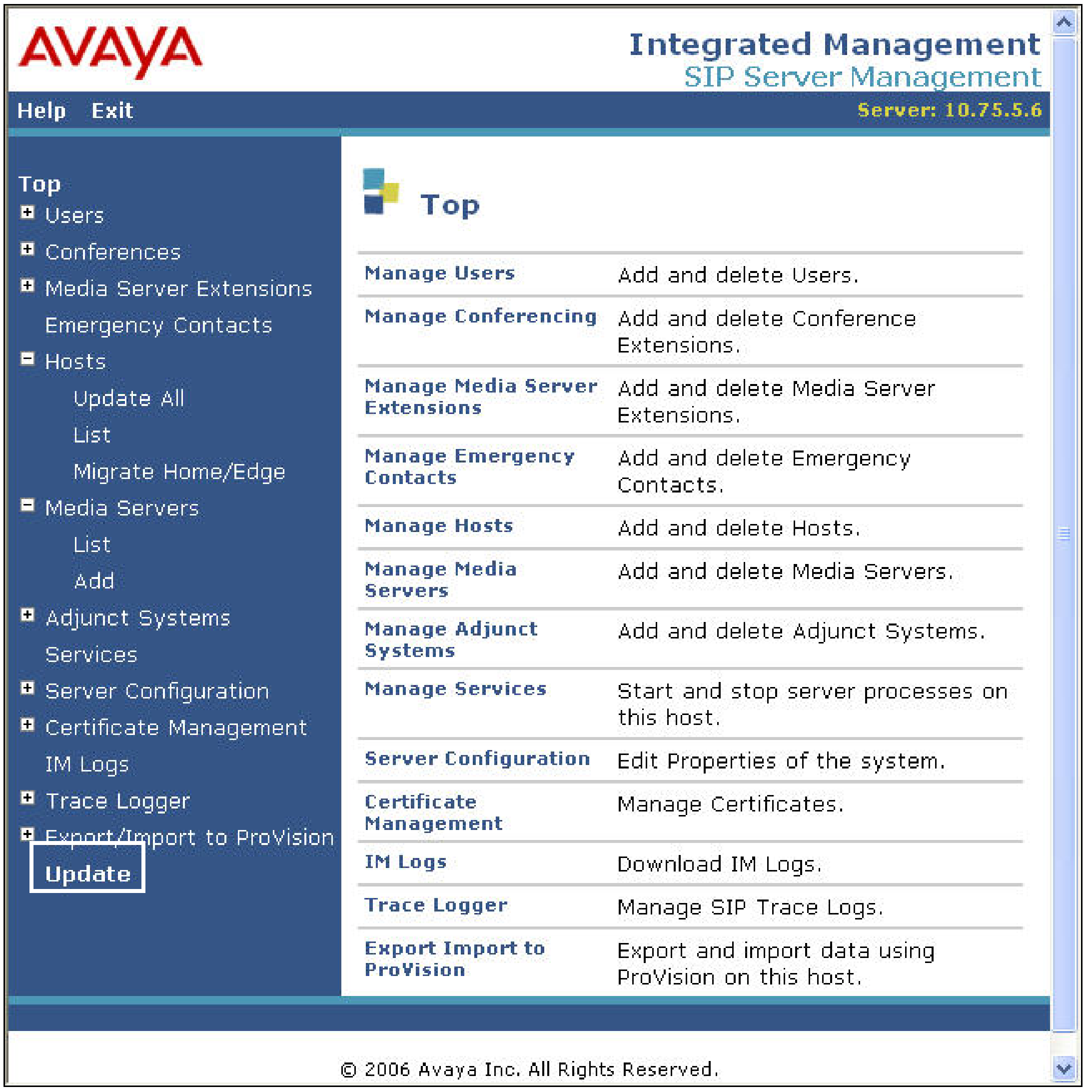
Step	Description
3.	After making changes within Avaya SES, it is necessary to commit the database changes using the **Update** link that appears when changes are pending. Perform this step by clicking on the **Update** link found in the bottom of the blue navigation bar on the left side of any of the Avaya SES administration pages as shown below. It is recommended that this be done after making each set of changes described in the following steps. 

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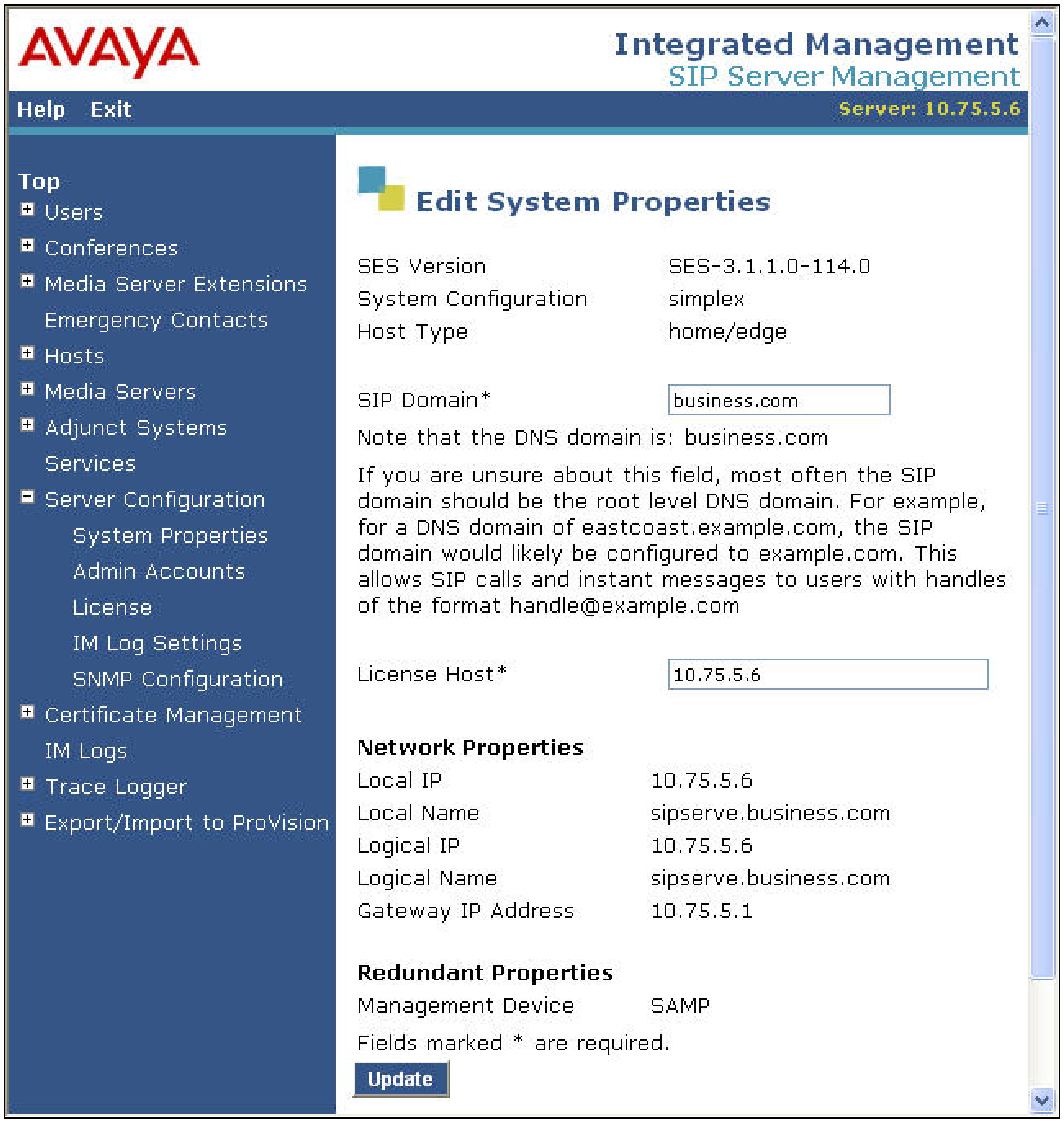
Step	Description
4.	From the left pane of the administration web interface, expand the **Server Configuration** option and select **System Properties**. The **Edit System Properties** page displays the software version in the **SES Version** field and the network properties entered during the installation process. On the **Edit System Properties** page: ▪ Verify the **SIP Domain** name assigned to Avaya SES. This must match the **Authoritative Domain** field configured on Avaya Communication Manager shown in Section 3, Step 5. ▪ Verify the **License Host** field. This is the host name, the fully qualified domain name, or the IP address of the SIP proxy server that is running the Avaya WebLM application and has the associated license file installed. ▪ After reviewing the **Edit System Properties** page, if any changes have been made, click the **Update** button. 

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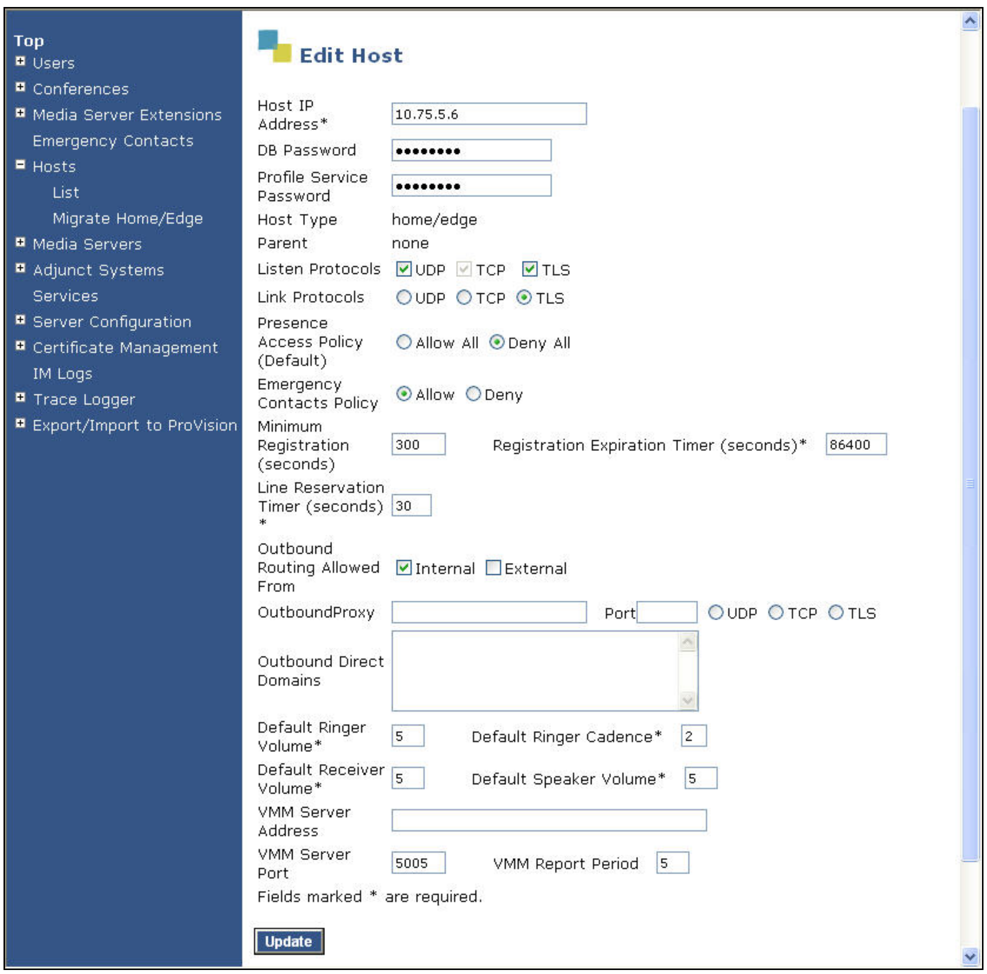
Step	Description
5.	After verifying the domain on the **Edit System Properties** page, verify the host computer entry for Avaya SES. The following example shows the **Edit Host** page since the host had already been added to the system. The **Edit Host** page shown below is accessible by clicking on the **Hosts → List** link in the left pane and then clicking on the **Edit** link under the **Commands** section of the subsequent page that is displayed (but not shown). ▪ In the **Host IP Address** field, verify the IP address of the Avaya SES server. ▪ Although the fields are hidden, the **DB Password** and **Profile Service Password** will reflect the values that were specified during the system installation. ▪ Since only one Avaya SES is used in the configuration, the **Host Type** will be set to *home/edge*. ▪ The default values for the other fields were used. If any changes were made, scroll down to the bottom of the page and click the **Update** button. 

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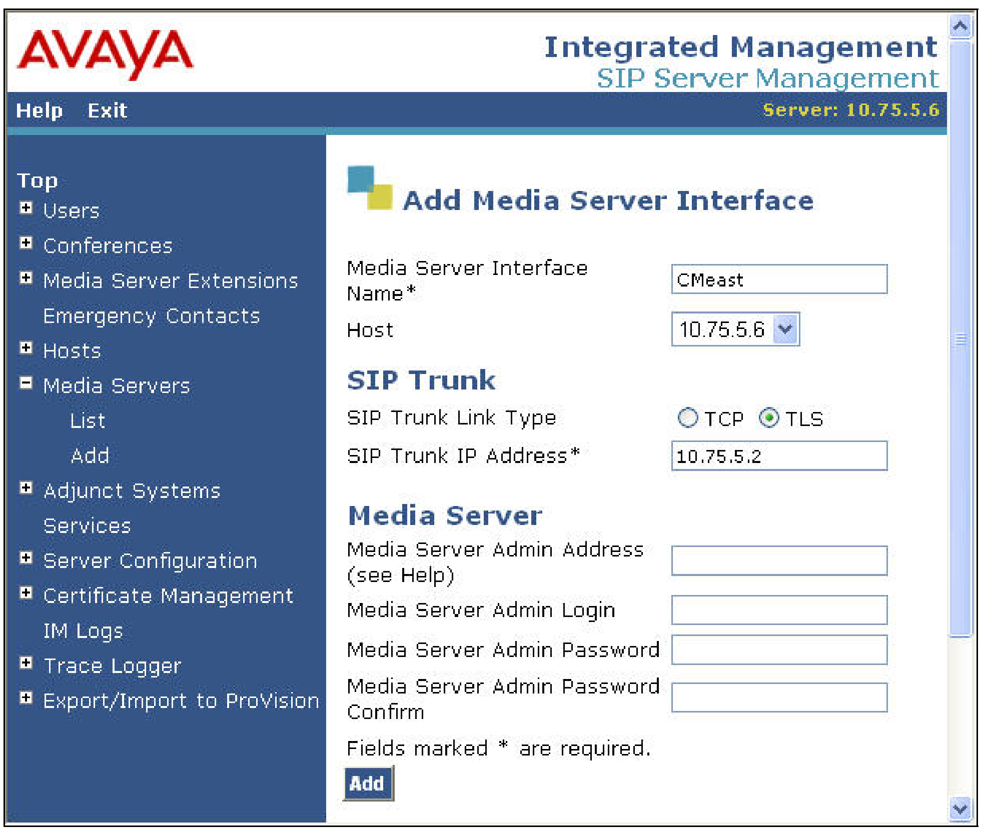
Step	Description
6.	From the left pane of the administration web interface, expand the **Media Servers** option and select **Add** to add the Avaya Media Server to the list of media servers known to Avaya SES. Adding the media server will create the Avaya SES side of the SIP trunk previously created in Avaya Communication Manager. On the **Add Media Server Interface** page, enter the following information: ▪ Enter a descriptive name in the **Media Server Interface Name** field (e.g. CMeast). ▪ In the **Host** field, select the Avaya SES server from the pull-down menu that will serve as the SIP proxy for this media server. Since there is only one Avaya SES server in this configuration, the **Host** field is set to the host shown in Step 5. ▪ Select ***TLS*** (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer. TLS is the only link protocol that is supported for communication between Avaya SES and Avaya Communication Manager. ▪ Enter the IP address of the Avaya S8300 Media Server in the **SIP Trunk IP Address** field. In alternative configurations that use a C-LAN board, the **SIP Trunk IP Address** would be the IP address of the C-LAN board. ▪ The default values may be retained for all other fields. ▪ After completing the **Add Media Server Interface** page, click the **Add** button. 

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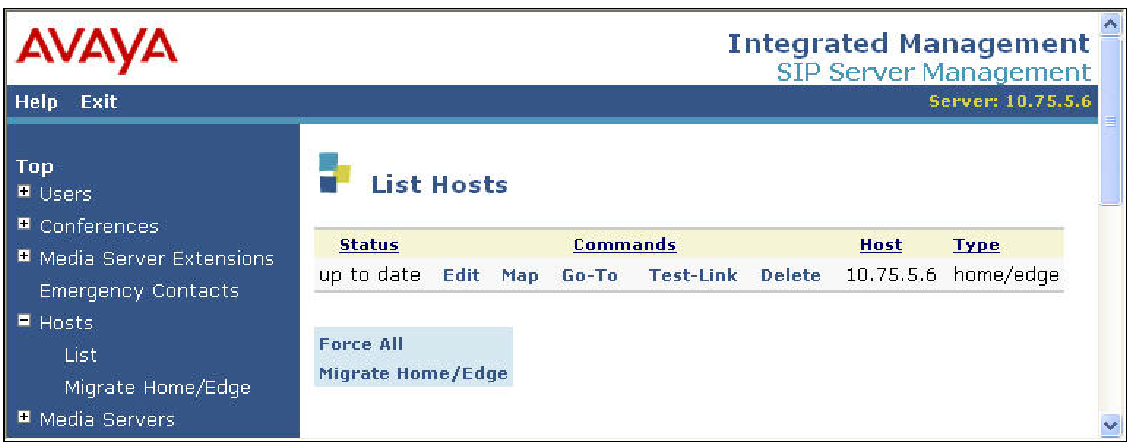
Step	Description
7.	A Host Address Map is required on Avaya SES to direct calls outbound from Avaya Communication Manager to the VMG. The VMG does not register as an endpoint with Avaya SES so calls are not automatically routed to the VMG based on a registered extension. Instead, an Address Map is used to route calls based on the contents of the SIP INVITE URI matching a specified pattern to determine the proper destination of the call. The URI takes the form of *sip:user@domain*, where *domain* can be a domain name or an IP address. The user portion can be an alpha-numeric name, telephone number or extension. In the case of the compliance test, the user portion contained the called party number. Calls with a called party number of 40000 were routed to the VMG. Thus, the Host Address Map was configured to match all calls dialing 40000. To configure a **Host Address Map**: ▪ Expand the **Hosts** option in the left pane of the administration web interface and select **List**. This will display the **List Hosts** page below. ▪ Click on the **Map** link to display the **List Host Address Map** page (not shown). On the **List Host Address Map** page, click on the **Add Map In New Group** link. 

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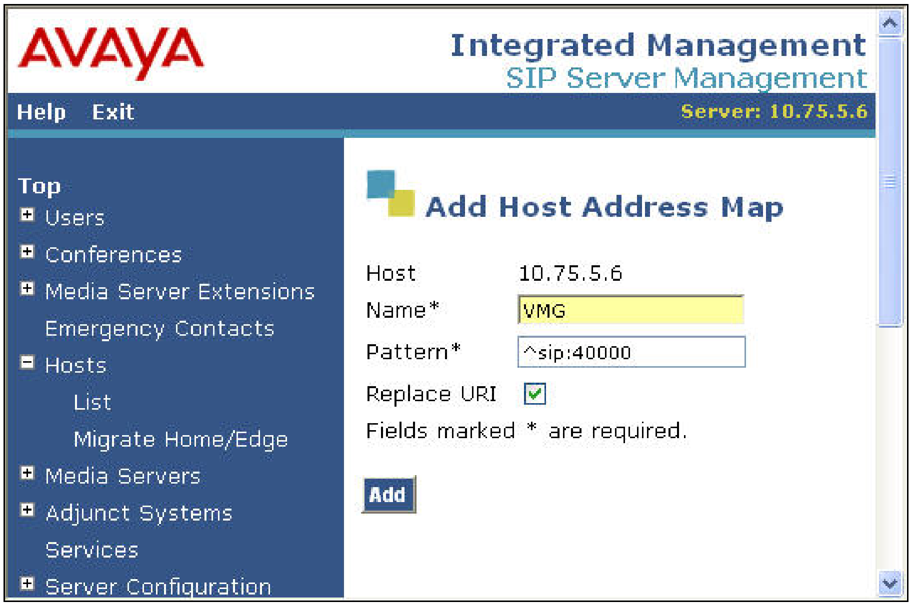
Step	Description
8.	On the **Add Host Address Map** page that appears: ▪ Enter a descriptive name in the **Name** field. ▪ In the **Pattern** field, enter an expression to define the matching criteria for calls to be routed to the VMG. The example below shows the expression used in the compliance test. This expression will match any URI that begins with ***sip:40000***. **Appendix B** contains additional information on the syntax used for address map patterns. Click the **Add** button. 

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<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>
Next, a Host Contact must be entered for the Address Map that was previously defined. The contact defines the destination IP address, port number and transport protocol to use when routing calls that match the Address Map.

To add a Host Contact:

Open the List Host Address Map page as described (but not shown) in Step 7.
Click on the Add Another Contact link associated with the address map added previously to open the Add Host Contact page shown below.
In the Contact field, enter the destination IP address (ip_addr), port number (port) and transport protocol (protocol) in the following format.

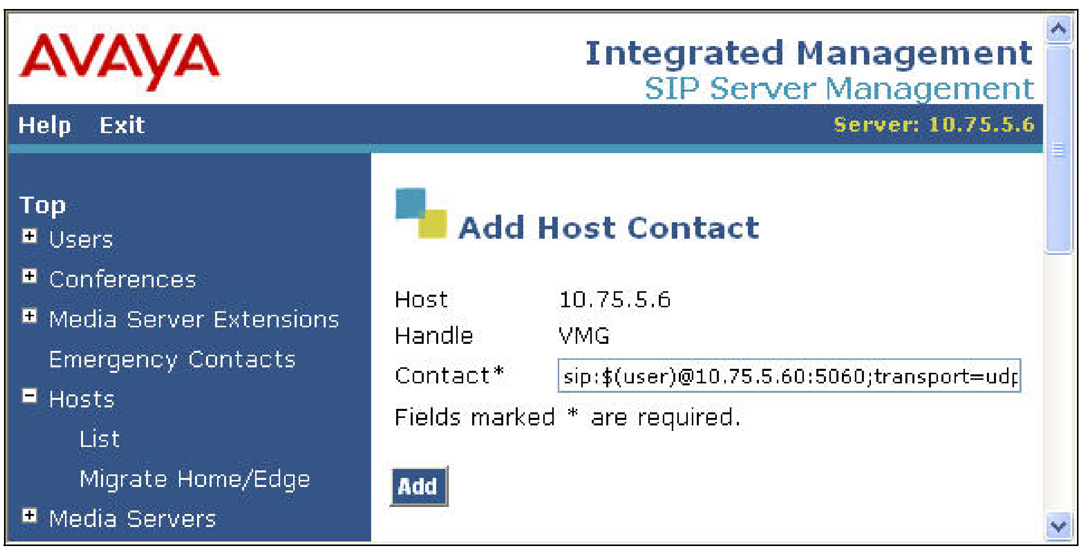
<code>sip:$(user)@ip_addr:port;transport=protocol</code>

The user part in the original request URI is inserted in place of the “$(user)” string before the message is sent to the destination.

For the compliance test, the VMG had IP address of 10.75.5.60. Thus, the following contact value was used:

<code>sip:$(user)@10.75.5.60:5060;transport=udp</code>

Click the Add button.



</td></tr>
</table>

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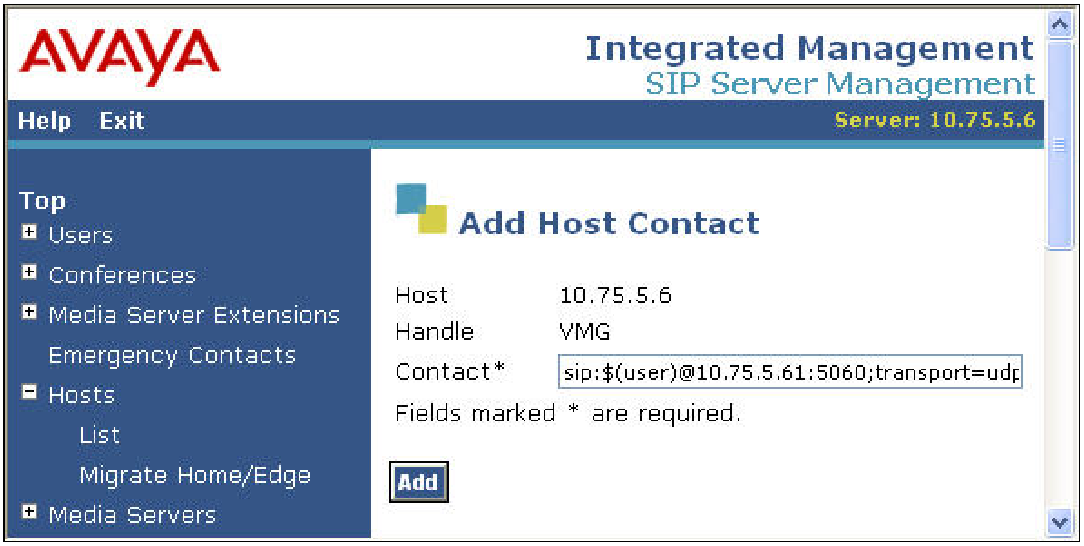
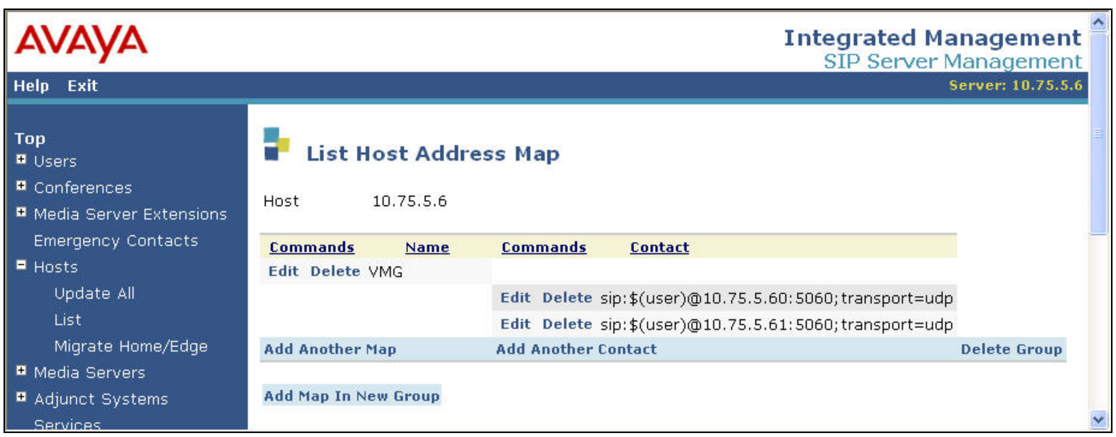
Step	Description
10.	Repeat Step 9 for the second (redundant) VMG. The contact information is the same except the IP address of the second VMG (10.75.5.61) is used as shown below. 
11.	After configuring the Host Address Map and Contact, the **List Host Address Map** page will appear as shown below. 

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<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>12.</td><td>
A Media Server Address Map is required on Avaya SES to direct calls inbound to Avaya Communication Manager in the same way that outbound calls from Avaya Communication Manager required a Host Address Map.

In the case of the compliance test, calls from the demo VoiceXML application are sent to the Avaya SES. These calls are the result of a user performing a transfer from the IVR to another user as shown in Figure 3. Since these calls originate from the IVR and not from a registered extension, the calls may not be automatically routed from the Avaya SES to Avaya Communication Manager. In particular, calls from the demo VoiceXML application to non-SIP endpoints will not be routed. Thus, a Media Server Address Map is required to route these calls. For simplicity, the Media Server Address Map was configured to match all calls dialed with a 5 digit number beginning with 3.

To configure a Media Server Address Map:

Expand the Media Servers option in the left pane of the administration web interface and select List. This will display the List Media Servers page below.
Click on the Map link associated with the appropriate media server to display the List Media Server Address Map page (not shown). On the List Media Server Address Map page, click on the Add Map In New Group link.

</td></tr>
</table>

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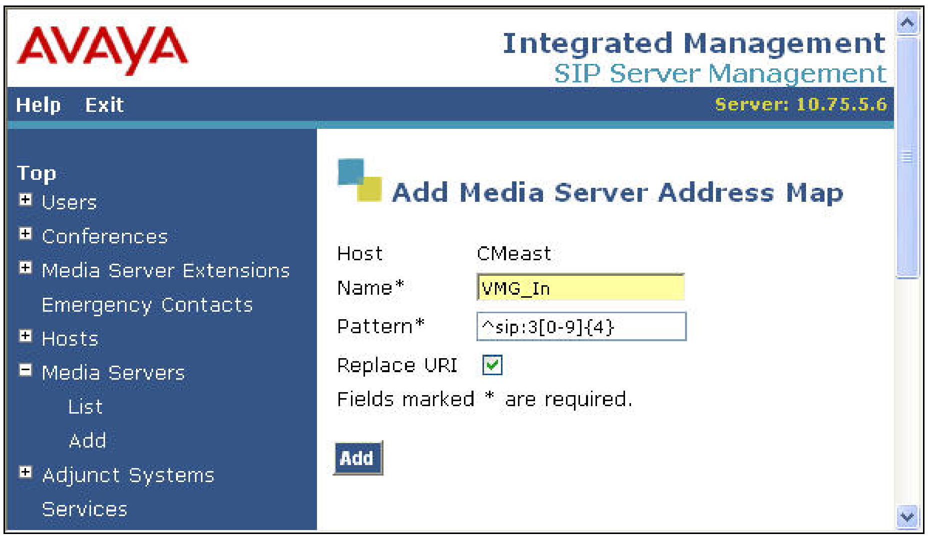
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>13.</td><td>

On the **Add Media Server Address Map** page that appears:

- Enter a descriptive name in the **Name** field.
- In the **Pattern** field, enter an expression to define the matching criteria for calls to be routed from the demo VoiceXML application to Avaya Communication Manager. The example below shows the expression used in the compliance test. This expression will match any URI that begins with ***sip:3*** followed by any digit between ***0-9*** for the next ***4*** digits. **Appendix B** contains additional information on the syntax used for address map patterns.

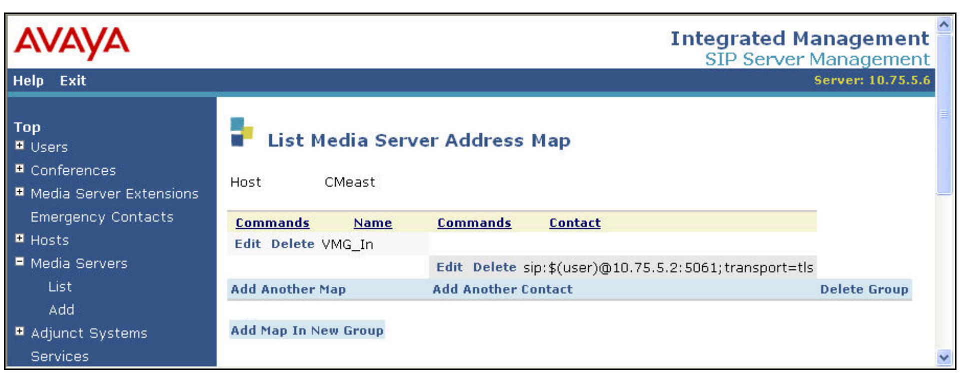
Click the **Add** button.



</td></tr>
<tr><td>14.</td><td>

After configuring the Media Server Address Map, the **List Media Server Address Map** page appears as shown below. The first Media Server Contact is created automatically and directs the calls to the IP address of the Avaya Media Server (***10.75.5.2***) using port 5061 and TLS as the transport protocol. The user portion in the original request URI is substituted for "$(user)". For the compliance test, the **Contact** field for the Media Server Address Map is displayed as:

```
sip:$(user)@10.75.5.2:5061;transport=tls
```



</td></tr>
</table>

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Step	Description

15.	Lastly, the IP address of the VMG must be configured as a trusted host on Avaya SES. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the designated IP address.
	
	To configure a trusted host:
	▪ Connect to Avaya SES and log in using proper credentials.
	▪ Enter the following **trustedhost** command at the Linux shell prompt.
	
	`trustedhost –a 10.75.5.60 –n 10.75.5.6 –c VMG`
	
	▪ Repeat the **trustedhost** command for the second VMG.
	
	`trustedhost –a 10.75.5.61 –n 10.75.5.6 –c VMG2`
	
	▪ Use the following **trustedhost** command to verify the entry is correct.
	
	`trustedhost –L`
	
	▪ **Important Note:** Complete the trusted host configuration by returning to the main Avaya SES administration web interface and clicking the **Update** link as shown in Step 3. If the Update link is not visible, refresh the page by selecting the **Top** link from the left menu. This step is required even though the trusted host was configured via the Linux shell.
	
	The screen below illustrates the results of the **trustedhost** commands.

```
admin@sipserve> trustedhost -a 10.75.5.60 -n 10.75.5.6 -c VMG
10.75.5.60 is added to trusted host list.

admin@sipserve> trustedhost -a 10.75.5.61 -n 10.75.5.6 -c VMG2
10.75.5.61 is added to trusted host list.

admin@sipserve> trustedhost -L
Third party trusted hosts.
 Trusted Host IP address  |   SES Host IP address     |         Comment
--------------------------+----------------------------+-------------------------
10.75.5.61                | 10.75.5.6                 | VMG2
10.75.5.60                | 10.75.5.6                 | VMG
```


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5. Configure the Newfound IP Call Recorder

The Newfound IP Call Recorder does not require any additional configuration for interworking with the Avaya SES and Avaya Communication Manager beyond the standard installation. This includes all components of the IP Call Recorder solution including the VMG and VoiceXML platform. As part of the standard installation, the IP addresses are defined for the VMG, VoiceXML platform and Avaya SES, if necessary. All configuration is performed by Newfound prior to shipping the IP Call Recorder to the end customer. For details on the installation and configuration procedures for the IP Call Recorder, refer to [7].

5.1. Configuring the VoIP Media Gateway (VMG)

The installation procedures referenced above create a XML configuration file called *vmgConfig.xml* in the home installation directory which is normally set to *C:/Newfound/VoIPMediaGateway*. The configuration file used for the compliance test is shown in **Appendix C**. Excerpts from the configuration file relating to the compliance test are highlighted below.

The following line in the configuration file defines the IP address for the VMG.

```
<VMGIpAddress>10.75.5.60</VMGIpAddress>
```

The following lines define the IP address and port number used by the VoiceXML platform. In the case of the compliance test, the VoiceXML platform was co-resident with the IP Call Recorder on the VMG. As a result, port 5065 was used for SIP signaling to the VoiceXML platform instead of the standard port 5060. Port 5060 was used by the IP Call Recorder.

```
<IvrGateways>
  <Host>10.75.5.60:5065</Host>
</IvrGateways>
```

The following lines show no entries under **AuthorizedIncomingGateways**. This indicates that the IP Call Recorder will accept calls from any IP address including the Avaya SES IP address. This is the default mode. If the calls are to be accepted only from the Avaya SES then an entry can be inserted under **AuthorizedIncomingGateways** in the form of `<Host>10.75.5.6</Host>`.

```
<Sip>
  … text omitted …
  <AuthorizedIncomingGateways>
    <!-- leave empty to allow invites from any GW, or restrict by IP -->
    <!-- <Host></Host> -->
  </AuthorizedIncomingGateways>
  … text omitted …
</Sip>
```


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The following lines define the hosts that can send XML RPC calls to the IP Call Recorder to control the recording. In the case of the compliance test, the only host (IP address) making XML RPC calls to the IP Call Recorder was the host where the VoiceXML application resides. Since the VoiceXML application resided on the VMG, both entries shown below under **AllowedHosts** refer to the VMG IP address.

```
<XmlRpc>
...
  <AllowedHosts>
  <!-- Hosts that are allowed to make XMLRPC calls to the IP Call Recorder on this system
(admin console, etc) -->
  <Host>10.75.5.60</Host>
  <Host>localhost</Host>
  </AllowedHosts>
</XmlRpc>
```

Two modes of recording are provided by the IP Call Recorder: full call recording and ad-hoc recording. Full call recording will record all calls all the time. Both modes were tested during the compliance test. The example configuration file in **Appendix C** shows the case with full call recording enabled. The relevant lines from the configuration file are shown below. To disable full call recording, change **enabled** to **disabled** in the excerpt below. To use ad-hoc recording, which provides control of the recording from the VoiceXML application, set full call recording to disabled. Either mode may be used to record calls even if the call has been transferred to another party.

```
<Recording>
  ... text omitted ...
  <FullCallRecording>enabled</FullCallRecording>
  ... text omitted ...
</Recording>
```

5.2. Configuring the VoiceXML Platform

The IP Call Recorder can use a number of different third-party VoiceXML platforms to provide the recording solution to the end customer. The configuration procedures will vary with each platform and will be performed by Newfound. Thus, the description of these procedures is beyond the scope of these Application Notes. However, it is important to note that either the VoiceXML platform or the VoiceXML application will need to be aware of the following information.

- Avaya SES IP address and SIP domain – This information is needed to create and send SIP INVITE messages to the Avaya SES when the VoiceXML application performs a transfer of an incoming call to another user.
- Dialed digits – This information is needed if the dialed digits defines which VoiceXML application will be executed. In the compliance test, any call that was sent to the IP Call Recorder (independent of the dialed digits) executed the demo VoiceXML application.

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5.3. Controlling Recording in the VoiceXML Application

Ad-hoc recording refers to recording which is controlled by the VoiceXML application. **Appendix D** shows an example of a VoiceXML application that enables and disables the IP Call Recorder. The application records the conversation where the application prompts the caller for his or her favorite color. It then starts a new recording in a different format and prompts the caller for his or her second favorite color, ends recording, and closes the application.
The following VoiceXML code from **Appendix D** starts the recording just before the application prompts the caller for his or her favorite color. The namelist is a list of variables defined earlier in the VoiceXML file which include such parameters as the command name, file name, and file type among others.

```
<subdialog name="record_start" src="../sendVMGCommand.jsp" namelist="command channelId fileId fileType mixerIp" />
```

After the caller responds to the prompts, the following VoiceXML code turns off the recording.

```
<subdialog name="record_end" src="../sendVMGCommand.jsp" namelist="command channelId mixerIp" />
```

6. Interoperability Compliance Testing

This section describes the compliance testing used to verify the interoperability between the Newfound IP Call Recorder, Avaya SIP Enablement Services (SES) and Avaya Communication Manager. This section covers the general test approach and the test results.

6.1. General Test Approach

The general test approach was to perform call recording on calls to the demo VoiceXML application or transferred through the application to another user. All calls to the demo VoiceXML application pass through the IP Call Recorder residing on the VMG for this purpose. Both full call and ad-hoc recording modes were tested where applicable.

6.2. Test Results

The IP Call Recorder successfully passed compliance testing. The following functionality was verified for calls passing through the IP Call Recorder.

- Recording of calls from the PSTN to the demo VoiceXML application.
- Recording of calls from SIP and non-SIP endpoints at the enterprise to the demo VoiceXML application.
- Recording of blind transferred calls to both SIP and non-SIP endpoints. In a blind transferred call, the demo VoiceXML application drops out of the call after initiating the transfer. The IP Call Recorder stays in the path.
- Recording of consultative transferred calls to both SIP and non-SIP endpoints. In a consultative transferred call, the demo VoiceXML application stays on the call until the transferred call is answered. If the call is not answered, then the caller can continue to

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interact with the demo VoiceXML application to select another menu option. The IP Call Recorder stays in the path.

- Recording of bridged transferred calls to both SIP and non-SIP endpoints. In a bridged transferred call, the demo VoiceXML application stays on the call for the duration of the call. At any time, the caller can interact with the demo VoiceXML application to select another menu option. The IP Call Recorder stays in the path.
- Support for recording multiple calls simultaneously.
- Support for recording long duration calls (> 1 hour).
- VoiceXML control of the recording via DTMF input or speech recognition.
- Proper system recovery after an IP Call Recorder restart.

It should be noted that since media shuffling must be disabled for interoperability that the media of each call flows through the Avaya Media Gateway for the duration of the call. Thus, media processing resources of the Avaya Media Gateway are consumed for each call.

7. Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling group is in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group is in-service.
- Verify that calls can be placed from the PSTN to the VMG and that call recording can be enabled and disabled.

8. Support

For technical support on the IP Call Recorder, contact Newfound Communications via the support link at www.newfoundcomm.net.

9. Conclusion

These Application Notes describe the procedures required to configure the Newfound IP Call Recorder to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager. The Newfound IP Call Recorder successfully passed compliance testing. For interoperability, media shuffling must be disabled. The impact of this configuration is noted in **Section 6.2**.

10. Additional References

[1] *Feature Description and Implementation For Avaya Communication Manager*, Doc # 555-245-205, Issue 4.0, February 2006.

[2] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 2.1, May 2006.

[3] *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700 and S8710 Media Server*, Doc # 555-245-206, Issue 6, February 2006.

[4] *Installing and Administering SIP Enablement Services R3.1*, Doc# 03-600768, Issue 1.5, February 2006.

[5] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, version 6.0, Doc # 210-100-500, Issue 9, June 2005.

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[6] *Avaya IA 770 INTUITY AUDIX Messaging Application*, Doc # 11-300532, May 2005.
[7] *Newfound MediaMixer, Newfound IP Call Recorder: Administration and Reference Guide for the VoIP Media Gateway [VoiceXML Edition]*, Version 1.4, 2007.
[8] *RTP Payload for DTMF digits, Telephony Tones and Telephony Signals*, RFC2833, H. Schulzrinne and S. Petrack, May 2000.

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

Product documentation for the Newfound IP Call Recorder is available from Newfound Communications.

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APPENDIX A: Supported Recording Formats

The following recording formats are supported by the IP Call Recorder.

- WAVE_PCM_8B8K_STEREO
- WAVE_PCM_16B8K_STEREO
- WAVE_PCM_8B8K_MONO
- WAVE_PCM_16B8K_MONO
- WAVE_ULAW_8B8K_STEREO
- WAVE_ALAW_8B8K_STEREO
- WAVE_ULAW_8B8K_MONO
- WAVE_ALAW_8B8K_MONO
- RAW_ULAW_8B8K_MONO
- RAW_ALAW_8B8K_MONO
- RAW_PCM_8B8K_MONO
- WAVE_ADPCM_8B8K_MONO
- WAVE_ADPCM_8B8K_STEREO
- WAVE_ADPCM_16B8K_MONO
- WAVE_ADPCM_16B8K_STEREO

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APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in Avaya SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period **`.`** matches any character once (and only once).
  - A asterisk **`*`** matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to the matched. Ranges are designated by using a hyphen. Thus, the expression **`[12345]`** or **`[1-5]`** both describe a pattern that will match any single digit between 1 and 5.
  - Curley brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' time. Thus, **`5{3}`** matches '555' and **`[0-9]{10}`** indicates any 10 digit number.
  - The circumflex character **`^`** as the first character in the pattern indicates that the string must begin with the character following the circumflex.

Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10 digit number in the North American dial plan would be:

`^sip:1[0-9]{10}`

This reads as: "Strings that begin with exactly **sip:1** and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

```
INVITE sip:17325551638@20.1.1.54:5060;transport=udp SIP/2.0
```


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APPENDIX C: IP Call Recorder Configuration File


```
  <?xml version="1.0" encoding="UTF-8" ?>
- <configuration>
    <NumberOfChannels>6</NumberOfChannels>
  - <Recording>
      <FullCallRecording>enabled</FullCallRecording>
      <!-- enabled / disabled  -->
      <Location>C:\Newfound\VoIPMediaGateway\Cache\</Location>
      <!-- Base location for recordings  -->
      <FileNameFormat />
    - <Format>
        <RIFFHeader>enabled</RIFFHeader>
        <!-- with or without headers  -->
        <Encoding>linear</Encoding>
        <BitRate>8</BitRate>
        <NumberOfChannels>2</NumberOfChannels>
        <!-- 1-mono 2-stereo  -->
      </Format>
    </Recording>
  - <Sip>
      <SipPort>5060</SipPort>
      <FromFormat>%N@%I:%P;channel=%C</FromFormat>
      <!-- ANI@IP:PORT;channel=channel#  -->
      <ToFormat>%N@%I:%P</ToFormat>
      <RegistrarAddress />
      <RegistrarUserName />
      <RegistrarPassword />
      <RegistrarExpires>0</RegistrarExpires>
      <InviteTimeout>30</InviteTimeout>
      <TransferInviteTimeout>119</TransferInviteTimeout>
      <SipProxyAddress />
      <ForwardHoldRequest>disabled</ForwardHoldRequest>
    - <Supported>
        <Headers />
        <ComfortNoise />
      </Supported>
    - <AuthorizedIncomingGateways>
        <!-- leave empty to allow invites from any GW, or restrict by IP  -->
        <!--  <Host></Host>  -->
      </AuthorizedIncomingGateways>
    </Sip>
  - <XmlRpc>
      <ServerPort>8888</ServerPort>
    - <AllowedHosts>
        <!-- Hosts that are allowed to make XMLRPC calls to the IP Call Recorder on this
        system (admin console, etc)  -->
        <Host>10.75.5.60</Host>
        <Host>localhost</Host>
      </AllowedHosts>
    </XmlRpc>
  - <Rtp>
      <ListenPortStart>8000</ListenPortStart>
      <SendPortStart>19000</SendPortStart>
      <MinimumPacketSize>1</MinimumPacketSize>
      <MaximumPacketSize>160</MaximumPacketSize>
    </Rtp>
  - <IvrGateways>
      <Host>10.75.5.60:5065</Host>
    </IvrGateways>
    <VMGIpAddress>10.75.5.60</VMGIpAddress>
```


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APPENDIX C (page 2): Configuration File Continued


```
- <Cache>
    <NumberOfCacheEntries>128</NumberOfCacheEntries>
    <LocalRepository>C:\Newfound\VoIPMediaGateway\Cache/</LocalRepository>
    <RemoteRepositoryURL>http://loudeye.com/muzak/get_some.php</RemoteRepositoryURL>
  </Cache>
- <FailOver>
    <Mode>Message</Mode>
    <GatewayToInvite />
    <MessageFile>busy.ulaw</MessageFile>
  </FailOver>
- <PortCapacity>
    <NumberOfPortCapacityLines>1</NumberOfPortCapacityLines>
    <MessageFile>portCapacityReached.ulaw</MessageFile>
  </PortCapacity>
</configuration>
```


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APPENDIX D: Example of VoiceXML Control of the IP Call Recorder

```
  <?xml version="1.0" ?>
- <vxml version="2.0" xmlns="http://www.w3.org/2001/vxml">
  - <!--

          ****  Vars used by VMG via XMLRPC

    -->
    <var name="command" />
    <var name="fileId" />
    <var name="waitTime" expr="'0'" />
    <var name="fileType" />
    <var name="channelIdTmp" expr="session.connection.remote.uri.match(/channel=channel\d+/).join()" />
    <var name="channelId" expr="channelIdTmp.match(/\d+/).join()" />
    <var name="from" expr="session.connection.remote.uri.match(/@[\d.]+/).join()" />
    <var name="mixerIp" expr="from.match(/[\d.]+/).join()" />
  - <form id="start">
    - <block>
        <assign name="fileId" expr="'TestRecording_16bit_stereo_pcm.wav'" />
        <assign name="command" expr="'RecordStart'" />
        <assign name="fileType" expr="'3'" />
        <!-- see the VMG QuickStart Guide for more on file types   -->
        <assign name="waitTime" expr="0" />
      </block>
      <!-- This will start recording immediately (waitTime = 0)   -->
      <!-- and send the output to the specified file.             -->
      <subdialog name="record_start" src="../sendVMGCommand.jsp" namelist="command channelId fileId
        fileType mixerIp" />
    - <field name="color">
        <prompt>Please say the name of your favorite color.</prompt>
      - <grammar mode="voice" root="color_choice" version="1.0">
        - <rule id="color_choice">
          - <one-of>
              <item>red</item>
              <item>blue</item>
              <item>green</item>
              <item>yellow</item>
              <item>orange</item>
              <item>purple</item>
              <item>black</item>
              <item>white</item>
            </one-of>
          </rule>
        </grammar>
      - <filled>
        - <prompt>
            You have chosen
            <value expr="color" />
            .
          </prompt>
        </filled>
      </field>
      <!-- This will stop recording immediately (waitTime = 0)   -->
    - <block>
        <assign name="command" expr="'RecordEnd'" />
        <assign name="waitTime" expr="0" />
      </block>
      <subdialog name="record_end" src="../sendVMGCommand.jsp" namelist="command channelId mixerIp" />
```


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APPENDIX D (page 2): Example Continued

```
- <block>
    <assign name="fileId" expr="'TestRecording_8bit_mono_ulaw.wav'" />
    <assign name="command" expr="'RecordStart'" />
    <assign name="fileType" expr="'8'" />
    <!--  see the VMG QuickStart Guide for more on file types   -->
  </block>
  <subdialog name="record_start2" src="../sendVMGCommand.jsp" namelist="command channelId fileId
    fileType mixerIp" />
- <field name="color2">
    <prompt>Please say the name of your second favorite color.</prompt>
  - <grammar mode="voice" root="color_choice2" version="1.0">
    - <rule id="color_choice2">
      - <one-of>
          <item>red</item>
          <item>blue</item>
          <item>green</item>
          <item>yellow</item>
          <item>orange</item>
          <item>purple</item>
          <item>black</item>
          <item>white</item>
        </one-of>
      </rule>
    </grammar>
  - <filled>
    - <prompt>
        I have you down for
        <value expr="color2" />
        .
      </prompt>
    </filled>
  </field>
  <!--  This will stop recording immediately (waitTime = 0)   -->
- <block>
    <assign name="command" expr="'RecordEnd'" />
    <assign name="waitTime" expr="0" />
  </block>
  <subdialog name="record_end2" src="../sendVMGCommand.jsp" namelist="command channelId
    mixerIp" />
 </form>
</vxml>
```


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