Avaya Solution & Interoperability Test Lab

Application Notes for NIKSUN NetVoice to Analyze H.323 Voice over IP Calls on Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the NIKSUN NetVoice to monitor and analyze H.323 Voice over IP (VoIP) traffic over an IP link between two Avaya Communication Manager systems. During compliance testing, telephone calls established between Avaya IP telephones, Avaya Digital telephones, and analog telephones traversing an IP link were successfully captured, analyzed and reported. In addition, the H.323 VoIP calls using G.711 IP codecs were successfully reconstructed from the captured RTP packets and played back. Information in these 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.

1. Introduction

These Application Notes describe a compliance-tested configuration comprised of NIKSUN NetVoice application running on NIKSUN NetVCR appliance and the Avaya Communication Manager resources, including the Avaya Media Servers, the Avaya Media Gateways, and the Avaya IP Telephones. NetVCR is an appliance that passively captures and records all inbound/outbound IP traffic including VoIP traffic. NetVoice is a software application that analyzes H.323 protocols for the captured H.323VoIP traffic, reconstructs and plays back H.323 calls, and generates Call Detail Record (CDR) reports. With these capabilities, NetVoice can help troubleshoot Avaya VoIP network by analyzing Voice over IP (VoIP) traffic between two Avaya Communication Manager systems linked by an H.323 IP trunk.

Figure 1 shows the sample network configuration, simulating a main site and a branch site connected via an H.323 IP trunk. The main site consists of an Avaya S8700 Media Server and an Avaya G600 Media Gateway. The branch site consists of an Avaya S8300B Media Server with an Avaya G700 Media Gateway. The IP infrastructure includes Avaya P333T-PWR switches and an Avaya P333R Stackable Switch. Each site supports Avaya IP, Digital, and analog telephones.

An IP link between the P333T-PWR at the main site and the P333R Stackable Switch at the branch site connects the two sites. An H.323 IP trunk is administered between the two sites to support voice calls. The inbound/outbound traffic on the inter-site IP link is mirrored to one of the P333T-PWR ports on the main site. NIKSUN NetVCR is connected to the mirrored port via 100 Mbps Ethernet connection, to capture and record all IP traffic. The NetVoice software application on NetVCR filters the VoIP traffic out of the all the recorded IP packets to analyze the VoIP traffic. A PC with an Internet Browser interface is connected to NetVCR to access the NetVoice application.

Note that the configuration is also applicable to other Avaya Media Servers and Media Gateways. The other infrastructure components, such as Avaya P333R Stackable Switch, support the verification and illustration of the Avaya/NIKSUN solution. The configuration of the infrastructure components is not the focus of these Application Notes and is not described.

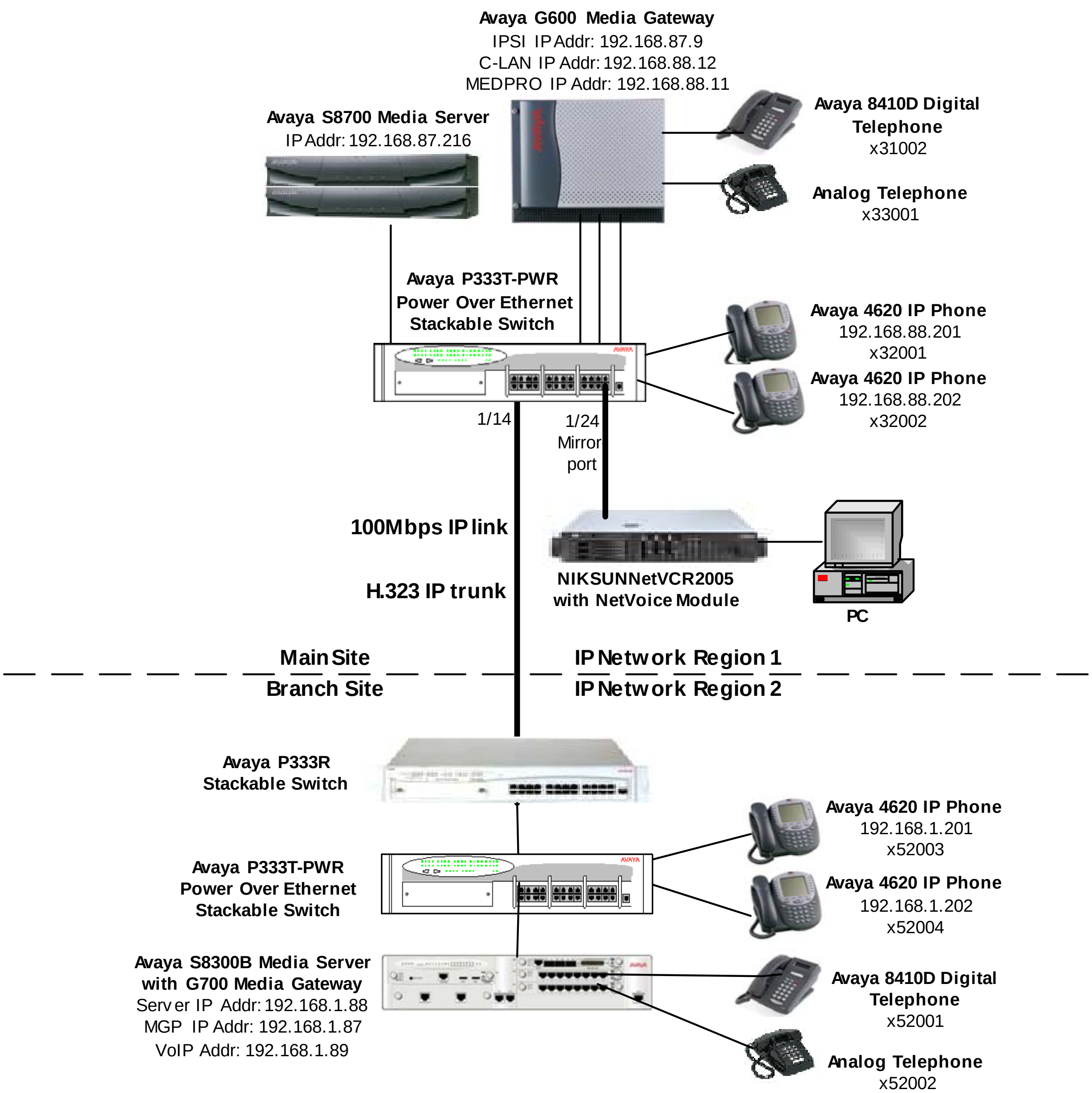


Figure 1: Sample Network Configuration

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 2.1.1 (R012x.01.1.414.1)
Avaya G600 Media Gateway • TN799DP C-LAN • TN2312AP IPSI • TN2302AP MedPro • TN2224BDigital Line • TN793 Analog Line	 11 9 93 10 6
Avaya S8300B Media Server	Avaya Communication Manager 2.1.1 (R012x.01.1.414.1)
Avaya G700 Media Gateway • Media Gateway Processor • MB_VOIP • MM712AP DCP Media Module • MM711AP Analog Media Module	 22.10.0(B) (MGP) 211(MB_VoIP) 5 17
Avaya 4620 IP Telephones	2.100
Avaya 8410D Digital Telephones	-
Analog Telephones	
Avaya P333T-PWR Power Over Ethernet Stackable Switch	4.0.17
Avaya P333R Stackable Switch	4.0.8
NIKSUN NetVCR 2005	3.1
NIKSUN NetVoice 2005	2.0.ic12
PC	Windows 2000 Professional SP4

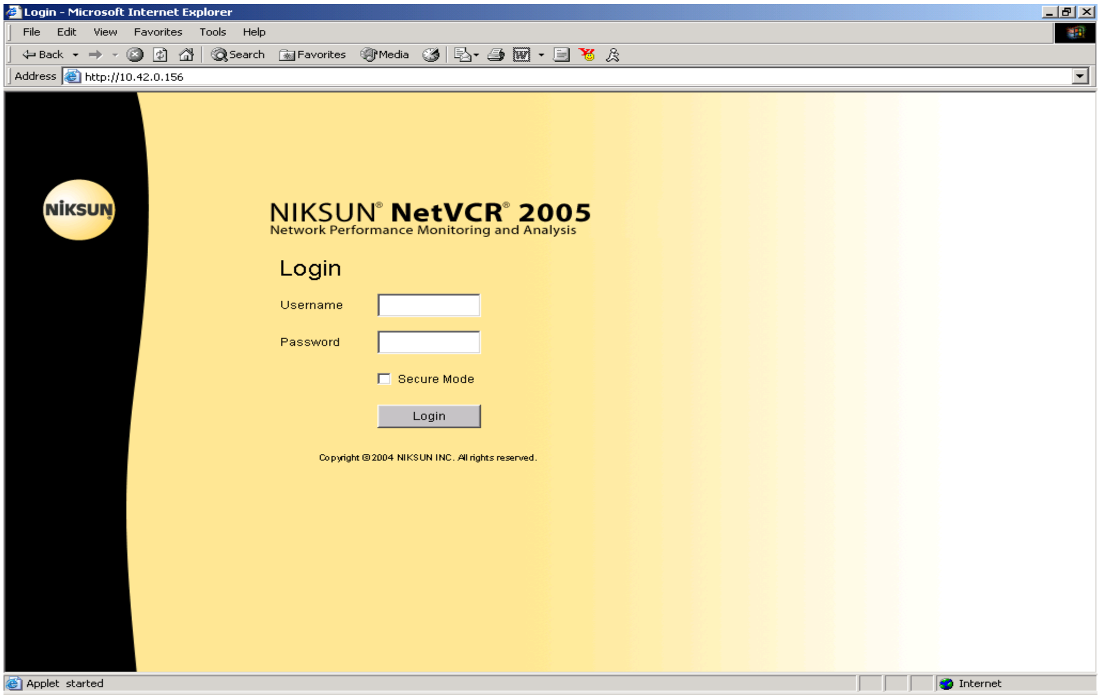
3. Configure NIKSUN NetVoice Application

This section describes the procedure to configure NetVoice application to ensure that the inter-site H.323 Voice over IP (VoIP) packets generated by Avaya Communication Manager resources, such as Avaya Media Servers, Avaya Media Gateways, and Avaya IP Telephones, can be captured and analyzed. NetVoice is an application that runs on NIKSUN NetVCR. This section also covers the procedure to configure NetVCR to enable NetVoice application to operate on this platform.

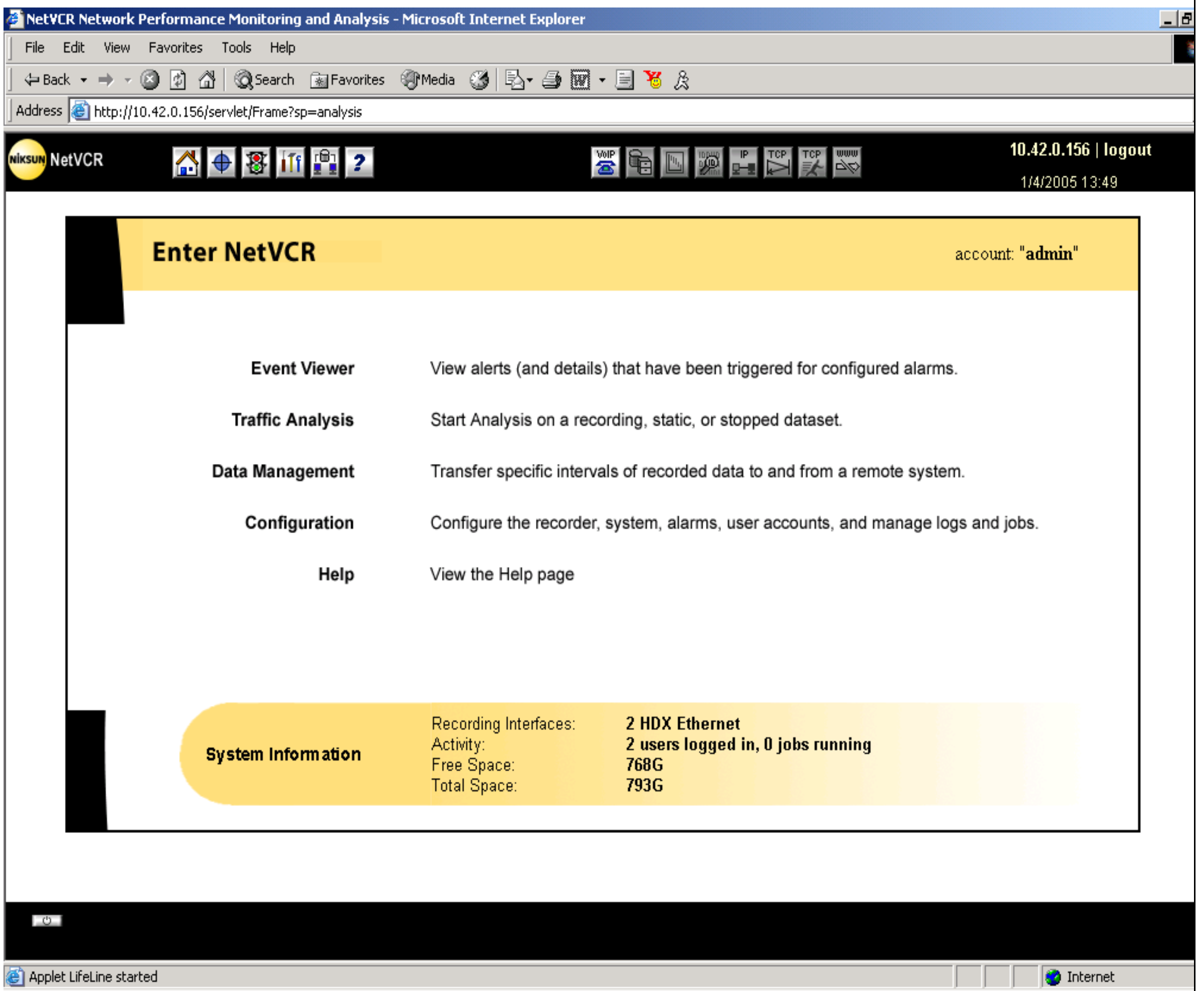
In addition, the Analysis section describes how to:

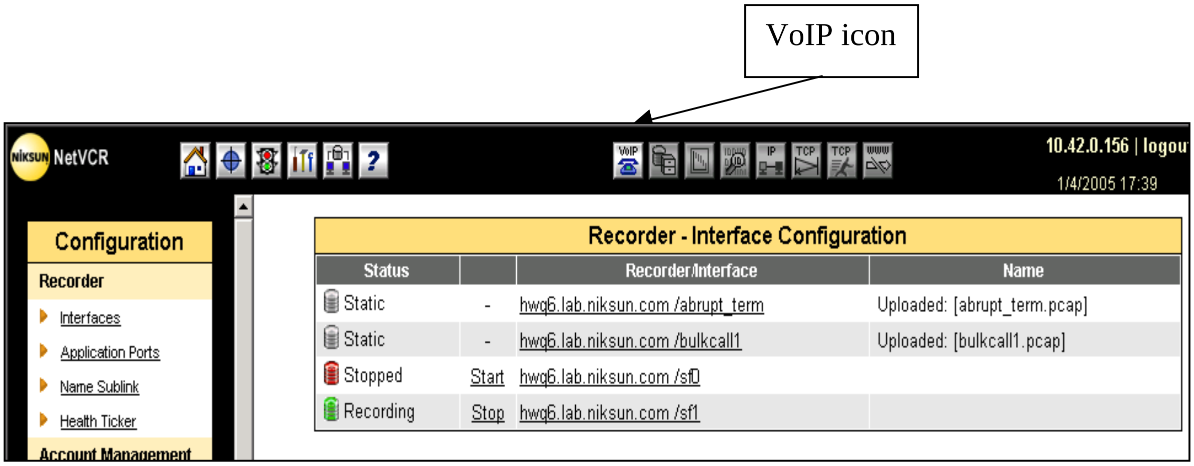
- Perform protocol analysis,
- Reconstruct VoIP call sessions for playback, and
- View Call Detail Records.

3.1. Configure NetVCR appliance

Step	Description
1.	Open a browser on the PC, enter the NetVCR IP address as the URL and log in with the appropriate password. 

Step	Description
2.	Click **Login**. The **Enter NetVCR** screen is displayed.



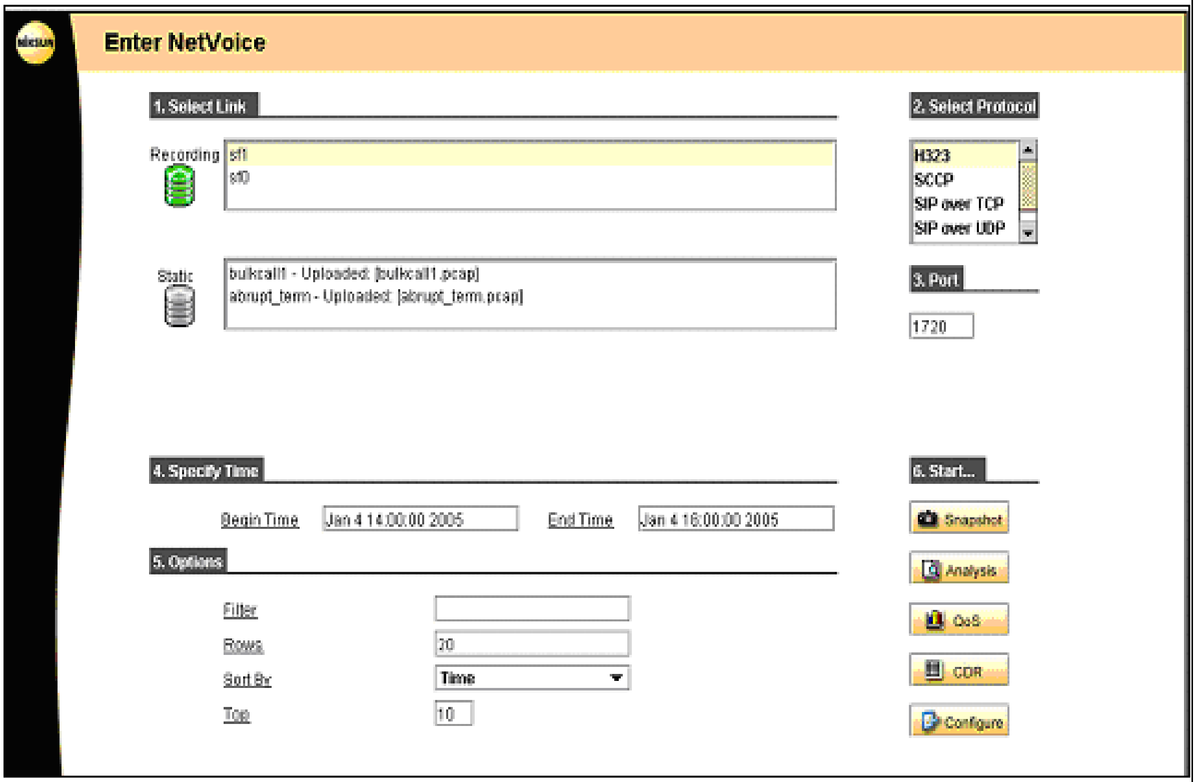
Step	Description
3.	From the **Enter NetVCR** screen click on **Configuration**. The **Recorder – Interface Configuration** screen appears. From the **Recorder – Interface Configuration**, select a **Recorder Interface** and click on **Start** to start recording the IP packets on the IP link (or **Stop** to stop recording). The **Status** will change to **Recording**. In the following example, the IP packets on the link connected to interface **/sf1** are recorded on NetVCR. 

3.2. Configure NetVoice Application

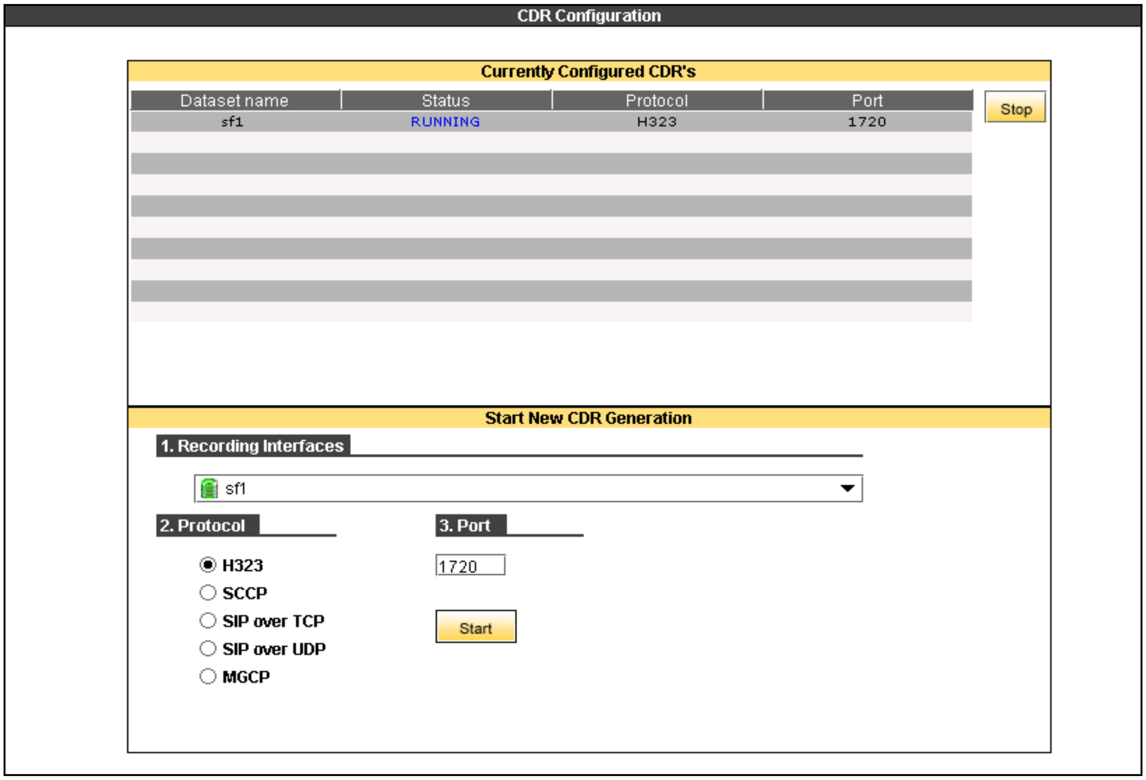
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>

From the NetVCR toolbar (see the picture in Section 3.1, Step 3), click on the **VoIP** icon to launch the NetVoice application. The **Enter NetVoice** screen is displayed. Configure the following parameters:

- **Select Link** for which the VoIP traffic will be analyzed. In the following example, the link **sf1** is selected.
- **Select Protocol** as **H.323**.
- Set the **Port** to **1720**.
- For **Specify Time** fields, select the time interval for which the traffic will be analyzed by setting the **Begin Time** and the **End Time**.

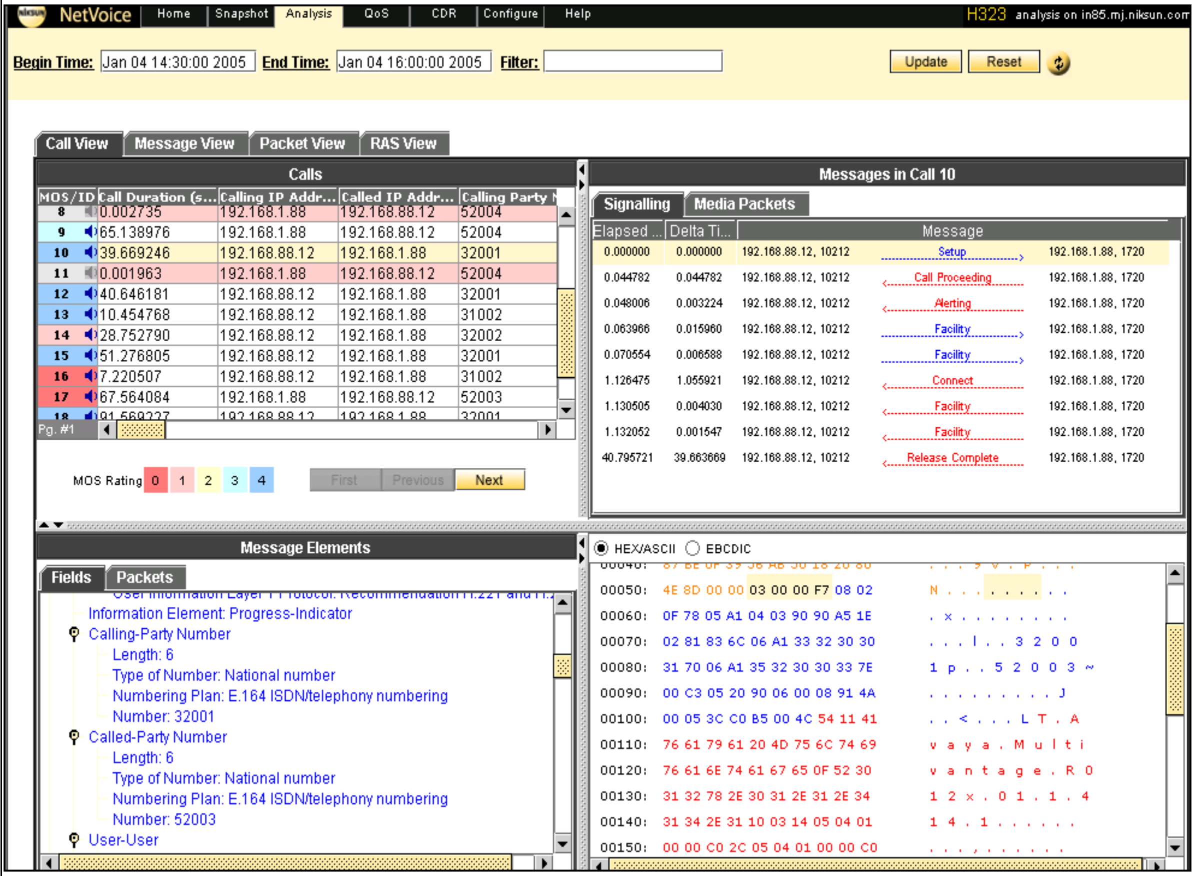


</td></tr>
</table>

Step	Description
2.	From the **Enter NetVoice** screen, click on the **Configure** button located under **6. Start** options. The **CDR Configuration** screen appears. Configure the parameters on this screen to start the generation of the Call Detail Records (CDR), as follows: • **Recording interfaces**: select a recording interface from the drop-down list. In the following example, the interface **sf1** is selected. • **Protocol**: click on the **H.323** protocol. • Set the **Port** to **1720**. • Click the **Start** button to start the CDR generation process. 

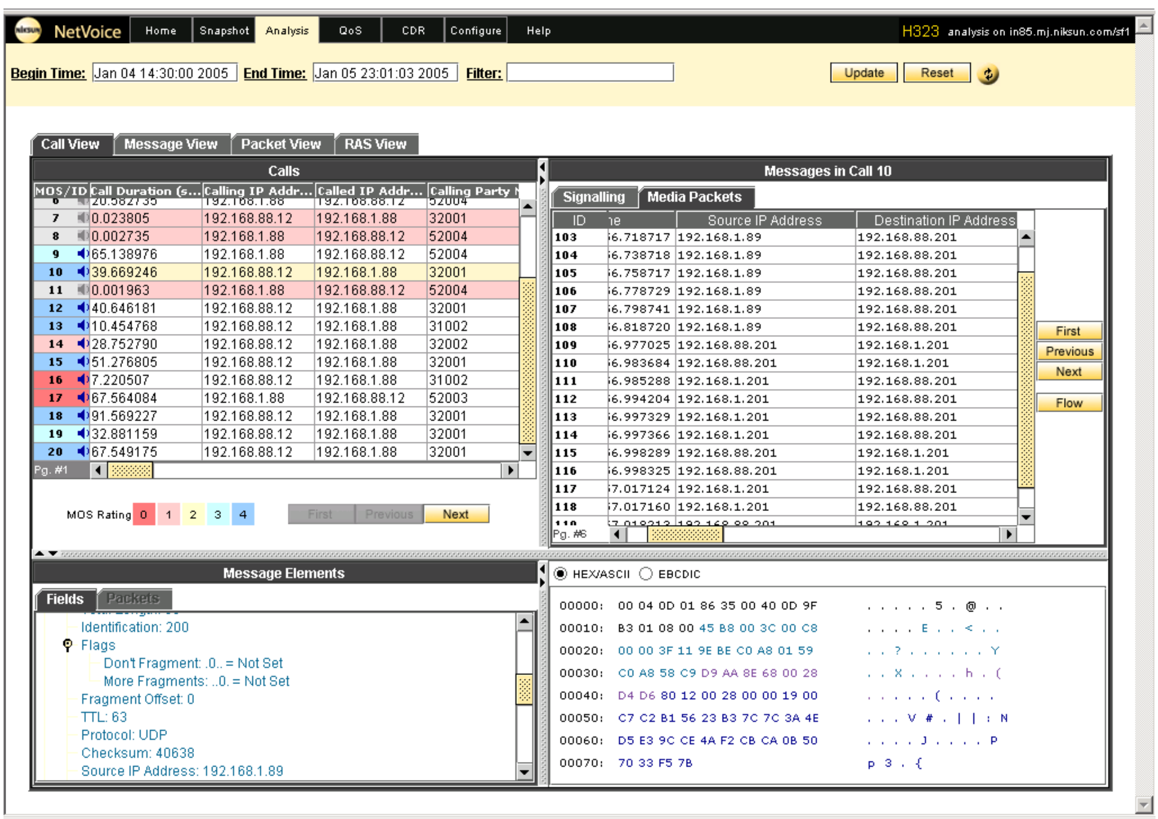
3.3. Traffic Analysis using NetVoice

Step	Description
1.	From the **Enter NetVoice** screen, click on the **Analysis** button. The **Call View** screen appears with the **Calls** pane. Double click a data row in the Calls pane to divide the **Call View** screen into four panes as shown below:



Step	Description
2.	**View the Call Details** From the **Calls** pane, view the list of all the calls, including the call details such as calling IP address, called party IP address, calling party number etc. in a specified time interval. Use the horizontal scroll bar to view all the call details. The example screen in Step 1 shows some of the calls in the time interval between the **Begin Time** set to **Jan 04 14:30:00 2005** and the **End Time** set to **Jan 04 16:00:00 2005**. The row for the call **ID 10**, in this example shows that the Calling Party is **32001**, an IP phone at the main site, the Calling IP Address is **192.168.88.12** (the IP address of the CLAN card in the Avaya G600 Media Gateway at the main site) and the Called IP Address is **192.168.1.88** (the IP address of the Avaya S8300B Media Server at the branch site). Note that since the call is between the main site and the branch site connected with an H.323 IP trunk, the call signaling over the inter-site IP link is exchanged between the CLAN card in the main site and the processor in the branch site, the two endpoints of the H.323 IP trunk, instead of the two IP telephones in the call.
3.	**View the Signaling Messages** Select a data row in the **Calls** pane and view the signaling messages for that call by selecting **Signaling** tab in the **Messages in Call x** pane, where x is the ID number associated with the call. The example screen in Step 1 shows a complete set of signaling messages for call **ID 10 –** from **Setup** to **Release Complete –** for a shuffled call between the two IP telephones. See Step 5 for the description of a shuffled call.
4.	**View the Signaling Message Fields** To view the fields in a particular signaling message, select a message in the **Messages in Call x** pane, and view the fields in this message by selecting the **Fields** tab in the **Message Element** pane. The example screen in Step 1 shows some of the fields for the **Setup** message for call ID 10. For example, the **Calling-Party Number** is **32001** and the **Called-Party Number** is **52003** in the **Setup** message.

Step	Description
5.	**View the Voice Media Packets** Select **Media Packets** tab in the **Messages in Call** pane to view the voice media packets (RTP messages) associated with a call, including the details such as the **Source IP Address** and the **Destination IP Address.** The source and destination IP address can help determine whether the call had been shuffled or not. **Note about the shuffled call**: Set the **Direct IP-IP Audio Connections** to **yes** in the Avaya Communication Manager configuration for H.323 IP signaling group (see Section 4.2.1). Now when an inter-site call is answered, the RTP audio paths are established directly between the IP telephones – shuffling from a path between a VoIP media processor and an IP telephone to a direct path between the two IP telephones. (Note that shuffling can also be enabled for other Avaya Communication Manager resources such as stations and IP Network Regions, when applicable.) The following example screen shows some of the media packets for call **ID 10** between two IP telephones. Note that the first few media packets were between the **Source IP address 192.168.1.89**, the IP address of the VoIP media processor at the branch site, and the **Destination IP address 192.168.88.201**, the IP address of the IP telephone at the main site. When the call was answered, the RTP media packets started flowing directly between the two IP telephones - the IP address 192.168.1.201 is for the IP telephone at the branch site and the IP address 102.168.88.201 is for the IP telephone at the main site).

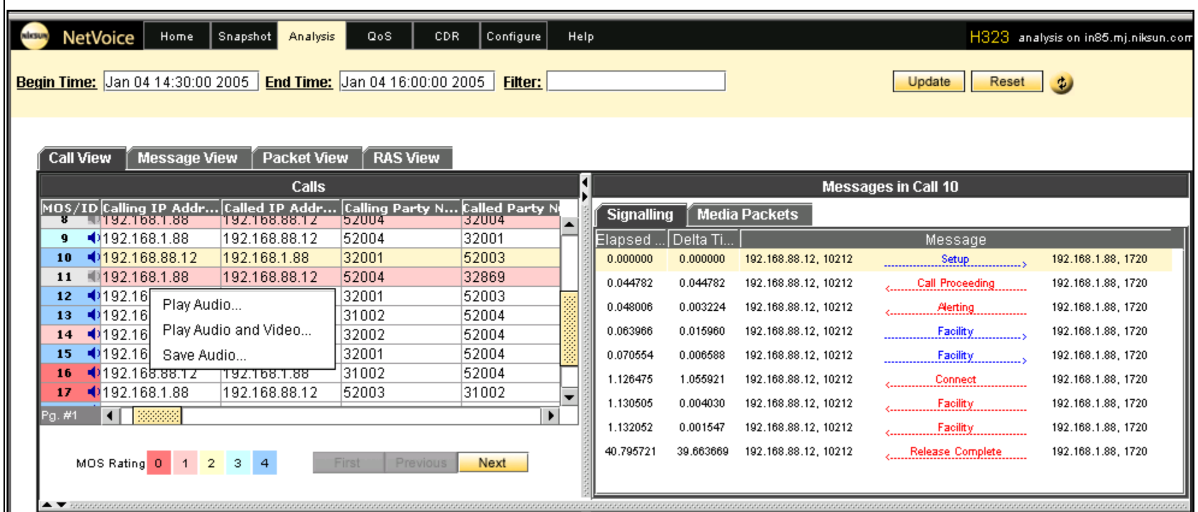


<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>

Playback the H.323 VoIP Call

Select a data row in the **Calls** pane to select a call. To reconstruct and playback the selected call, right click on the “Audio” icon in the **MOS/ID** column. A small window pops up with the playback options. Click **Play Audio** to reconstruct the H.323 voice session from the associated RTP packets and playback.

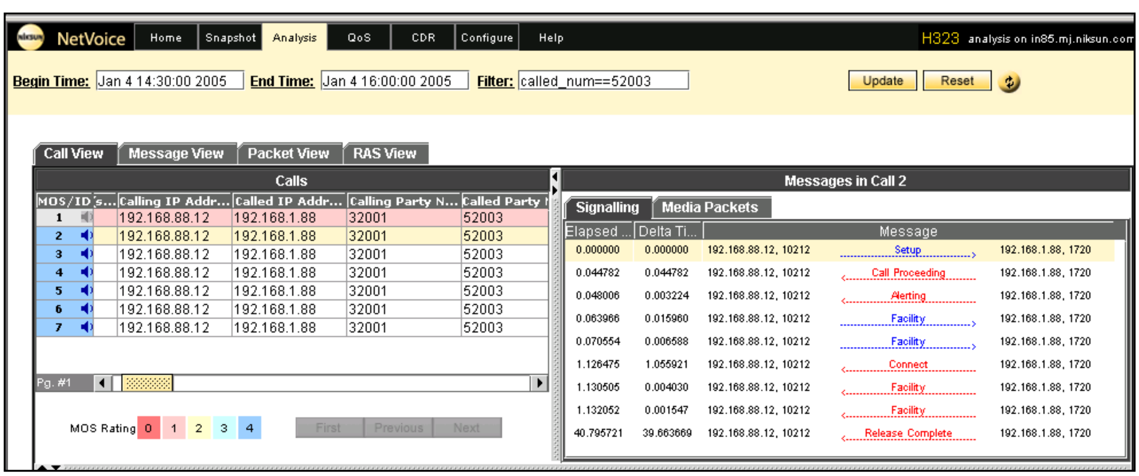
The following screen shows that the Call **ID 10** is being selected for playback. Note that the Mean Opinion Score (**MOS**) rating in the MOS/ID column reports the appropriate voice quality, as computed by NIKSUN’s proprietary algorithm. The MOS rating is color coded as shown in the screen, 0 is poor quality and 4 and above is the best voice quality.



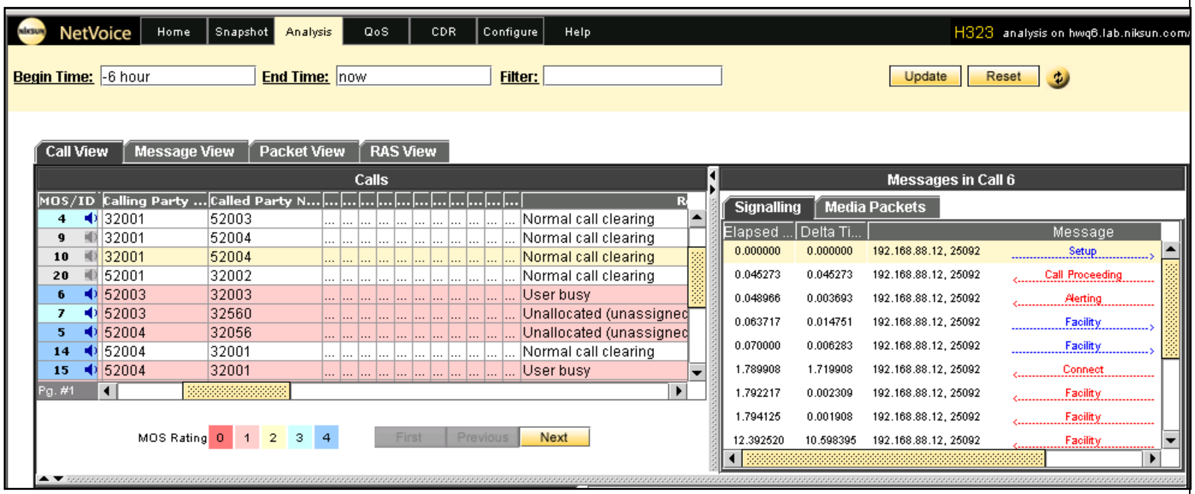
</td></tr>
<tr><td>7.</td><td>

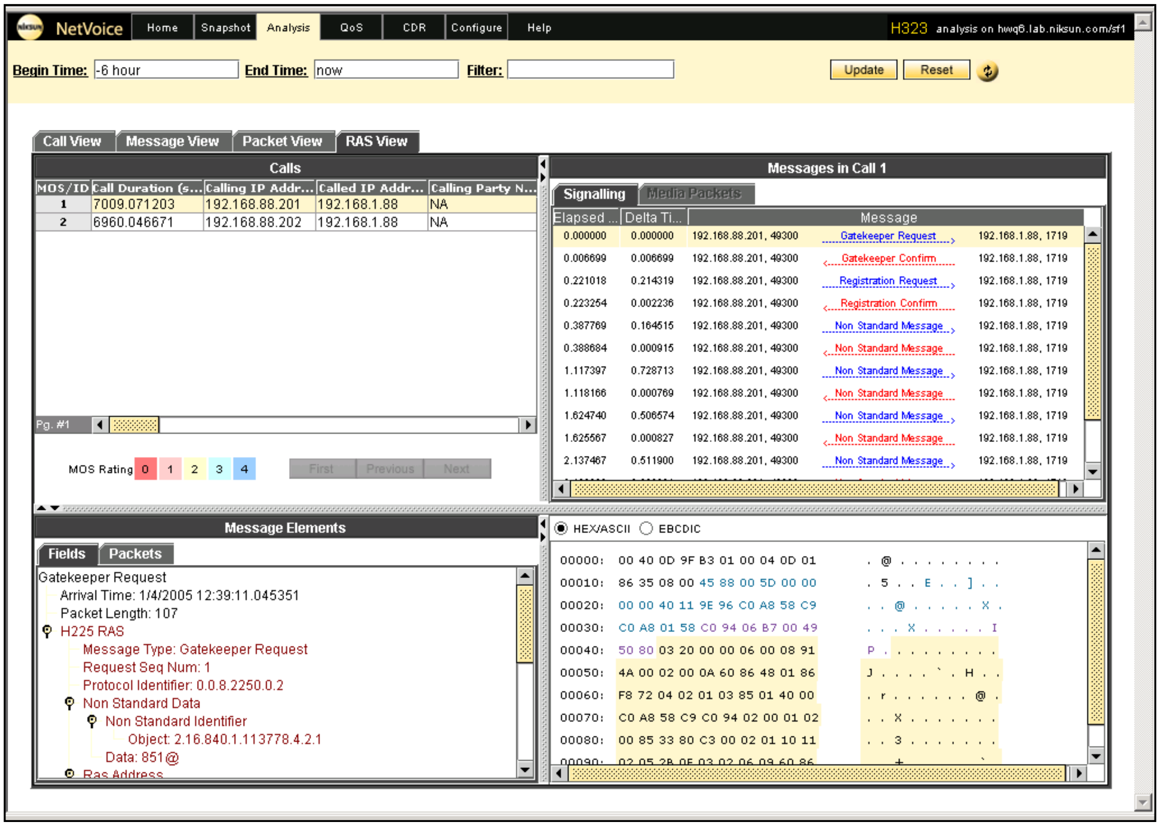
View using Filter

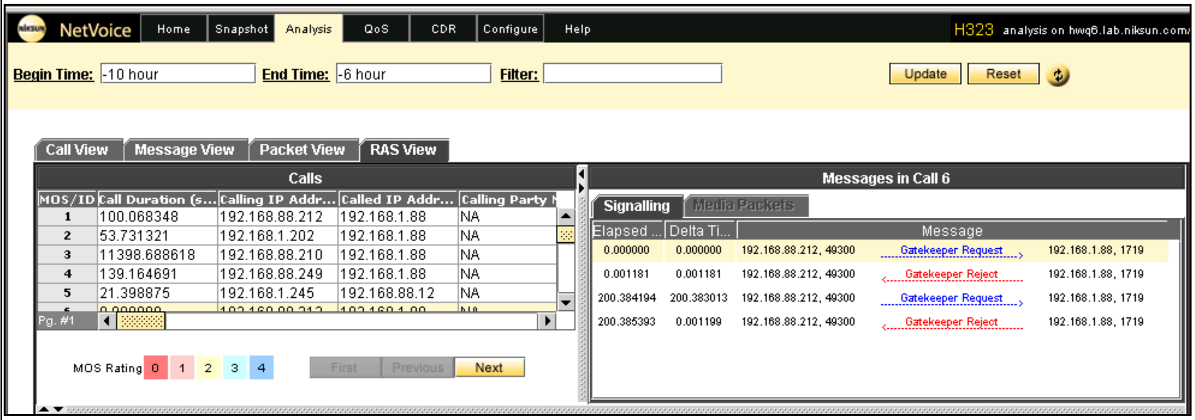
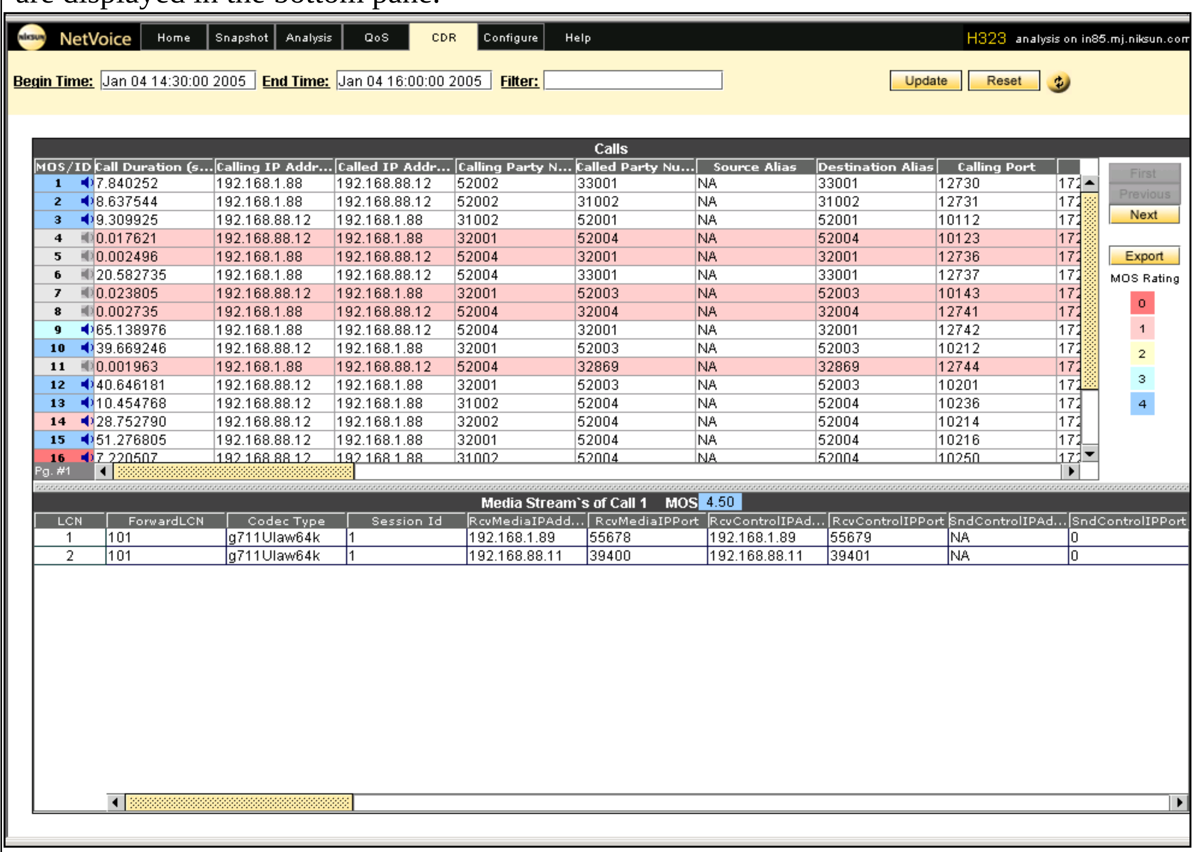
To filter the VoIP packets for a specific criteria, specify the filter expression in the **Filter** field. The following example shows all the calls for **Filter** set to **called number== 52003**.



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

Step	Description
8.	**Calls with various Release Codes** The following example shows another view of the **Calls**, showing call details such as the release code of complete and incomplete calls. For example, **Normal call clearing** release code for a successfully completed call, and **User Busy** for a call to a busy telephone. 

Step	Description
9.	**View the Registration, Admission and Status (RAS) Messages** From the **Call View** screen, click on the **RAS View** tab. The following screen appears, similar to the **Call View** screen. Click on the **Signaling** tab in the **Messages in Call** pane to view the RAS messages for a selected call. The following example shows the RAS messages when an IP telephone with an IP address of **192.168.88.201** at the main site registers with an Avaya Media Server at the branch site with an IP address of **192.168.1.88**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>10.</td><td>View the Reject RAS Messages
The following example shows another view of the RAS Signaling, showing the rejected registration messages, when an incorrect extension and password is entered during registration.

</td></tr>
<tr><td>11.</td><td>View the Call Detailed Record (CDR)
From the Enter NetVoice screen, click on the CDR button. The Calls screen appears with all the call details, similar to Call View. In addition, the Media Streams of a selected call are displayed in the bottom pane.

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

4. Configure the Avaya Communication Manager

4.1. Configure VoIP Attributes

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:
```

4.1.2. IP Network Region

In the sample configurations described in these Application Notes, the main site and the branch site is assigned an IP network region. The main site is assigned an IP network region 1 and the branch site is assigned an IP Network region 2. To configure 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 change ip-network-region form, configure the following:

1. **Codec Set –** Enter the number of the codec set that will be used in this region.
2. **Inter-region IP-IP Direct Audio –** if set to **yes**, RTP audio paths may be established directly between an IP telephone in this region and an IP telephone in another region that also has this parameter set to **yes**. These are also called **shuffled paths.**

```
change ip-network-region 2                                      Page   1 of 19
                               IP NETWORK REGION
  Region: 2
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
```

On Page 3 of the change ip-network-region form, set the **codec-set** to **1** for the calls between source region 1 and destination region 2**,** implying that the G.711 codec will be used for the calls between the two regions. See Section 4.1.1 for codec configuration.

```
change ip-network-region 1                                    Page   3 of  19

                   Inter Network Region Connection Management

 src dst
 rgn rgn      codec-set  direct-WAN  WAN-BW-limits  Intervening-regions
 1   1           1
 1   2           1           y             :NoLimit
 1   3
```

4.1.3. Assign IP Addresses to IP Network Regions

To assign an IP Network Region to a range of IP addresses, enter **change ip-network-map** using the SAT. On Page 1 of the change ip-network-map form, enter one or more IP address ranges and the IP network regions to which they belong. In the example below, IP endpoints (i.e., IP telephones) in the 192.168.1.0 subnet are assigned to IP Network Region 2.

```
change ip-network-map                                            Page   1 of  32
                         IP ADDRESS MAPPING

                                                                Emergency
                                             Subnet             Location
 From IP Address  (To IP Address   or Mask)  Region    VLAN    Extension
 192.168.0  .0    192.168.1  .255             2        n
 192.168.2  .0    192.168.255.255             1        n
    .   .   .        .   .   .                         n
    .   .   .        .   .   .                         n
    .   .   .        .   .   .                         n
```

4.2. Configure H.323 IP trunk between the two sites

4.2.1. Signaling Group

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

- **Group Type and trunk group:** Enter **h.323** for **Group Type** and associate this signaling group with an H.323 trunk group by entering the value of trunk group number in the **Trunk Group for Channel Selection** field.
- **Node names and the listen ports:** Enter the **Near-end Node Name**, the **Far-end Node Name**, the **Near-end Listen Port**, the **Far-end Listen Port**, and the **Far-end Network Region**.
- **Direct IP-IP Audio Connections –** if set to **yes**, RTP audio paths are established directly between IP telephones that use the IP trunk. These are also called the **shuffled paths.**

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

```
add signaling-group 8                                        Page   1 of   5
                                SIGNALING GROUP

 Group Number: 8             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: 8
          Supplementary Service Protocol: a


        Near-end Node Name: clan-01a03       Far-end Node Name: S8300-procr
      Near-end Listen Port: 1720           Far-end Listen Port: 1720
                                        Far-end Network Region: 2
              LRQ Required? n            Calls Share IP Signaling Connection? n
              RRQ Required? n
                                             Bypass If IP Threshold Exceeded? n

              DTMF over IP: in-band           Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
                                              Interworking Message: PROGress
```

4.2.2. Trunk Group

Configure an H.323 IP trunk by adding or changing a trunk group. 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**.
- **Carrier Medium:** Enter **IP**.
- **Service Type:** Enter **tie** to set this as an IP tie trunk between the two servers.

```
add trunk-group 8                                             Page   1 of  22
                                TRUNK GROUP

Group Number: 8                   Group Type: isdn         CDR Reports: y
  Group Name: H.323 Calls to S8300       COR: 1        TN: 1        TAC: 908
   Direction: two-way       Outgoing Display? n        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
```

4.2.3. 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 8                                            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                                            8
  2: T00185                                            8
  3: T00186                                            8
  4: T00187                                            8
  5: T00188                                            8
```

5. Interoperability Compliance Testing

The interoperability compliance testing focused on assessing the capabilities of NetVoice to capture and analyze H.323 VoIP data packets in a VoIP network comprised of Avaya

Communication Manager systems. The test network was divided into two simulated sites connected by an IP link. The H.323 VoIP traffic was generated by the resources in Avaya Media Servers, Avaya Media Gateways, Avaya IP telephones, Avaya Digital telephones, and the analog telephones. An H.323 IP trunk was established over the IP link to carry voice connections between the two sites. NetVoice device was employed at the main site to capture and analyze the VoIP traffic on the inter-site IP link. The traffic analysis focused on

- H.323 VoIP protocol analysis
- Reconstructing VoIP sessions for playback, and
- Call Detail Records.

5.1. General Test Approach

The general approach was to capture and analyze the H.323 VoIP traffic between the main site and the branch site, by making inter-site telephone calls. The main objectives were to verify that:

- A list of all the inter-site H.323 VoIP calls in a specified time interval is reported correctly in NetVoice Call View and NetVoice Call Detailed Record (CDR) screens.
- For each H.323 VoIP call, the call details such as calling number, called number, calling IP address, called IP address etc. are reported correctly in the Call View.
- For each successful or incomplete H.323 VoIP call, the H.323 signaling messages such as Setup, Alerting etc. are reported correctly in the Call View. Calling a busy telephone is an example of an incomplete call.
- Each H.323 message can be decoded to show the message elements.
- For G.711 codec, the H.323 VoIP call can be reconstructed and played back. Note that G.729 calls cannot be played pack, but the associated RTP messages are monitored and captured.
- The inter-site voice media packets, such as RTP packets, between the two IP telephones or the two Media Processors are reported correctly, including the source and destination IP address.
- For each call, the Mean Opinion Score (MOS) reports the appropriate voice quality, as computed by the NIKSUN's proprietary algorithm.
- Inter-site calls among Avaya IP, digital, and analog telephones can be analyzed and reported correctly.
- Shuffled and non-shuffled calls can be analyzed and reported correctly. Shuffled calls are H.323 VoIP calls where the two IP telephones directly exchange RTP packets.
- In Hold, Conference, and Transfer scenarios for the inter-site calls, each leg of the call can be analyzed and reported correctly. See Section 5.2 Test Results, for related observations on playing back some legs of the calls.
- Registration, Admission and Status (RAS) are correctly reported for the successful and rejected registrations.
- NetVoice can capture and analyze the calls under bulk call volume.

5.2. Test Results

All test cases completed successfully. NetVoice was able to capture and analyze H.323 VoIP traffic across an IP link between the two Avaya Communication Manager systems. Call Signaling and RAS signaling for successful and rejected calls was reported correctly. In addition, the H.323 VoIP calls using G.711 IP codecs were successfully reconstructed from the captured RTP packets and played back, including the calls that were enabled for shuffling in Avaya Communication Manager.

The following observations were made in some scenarios during the testing regarding the reconstructing VoIP calls for playback:

1. When a user puts an inter-site call on Hold and makes a second call, any conversation between the time that user presses the Hold button and the second call is answered, is played back as part of the first call.
2. When a user transfers an inter-site call to a second user on the other site (e.g. the user on the main site transfers the calls to another user on the branch site), playing back the first call plays back only one-way instead of two-way conversation for the first leg of the call (before the transfer button was pressed). This observation was made when the Direct IP-IP Audio was enabled in the Avaya Communication Manager.
3. When a user conferences an inter-site call with a second user on the other site (e.g. the user on the main site conference in another user on the branch site), playing back the first call plays back only one-way instead of two-way conversation for the first leg of the call (before the conference button was pressed). This observation was made when the Direct IP-IP Audio was enabled in the Avaya Communication Manager.

6. Verification Steps

The following steps may be used to verify the configuration:

- Enable the shuffling feature for IP-IP telephone calls via SAT by setting the **Direct IP-IP Audio Connections** to **yes** in the Avaya Communication Manager configuration for H.323 IP trunk signaling group (see Section 4.2.1).
- Establish, maintain, and tear down an IP-IP telephone call phone from the main site to the branch site.
- Using the **NetVoice Call View** screen, verify that the call signaling is reported properly and verify that the voice call can be played back. See Section 3.3 for procedure to view NetVoice screens and perform traffic analysis.

7. Support

For technical support on the NetVoice products, consult www.niksun.com/support or contact NIKSUN Technical Support at 1-888-821-2003.

8. Conclusion

These Application Notes illustrate the procedure for configuring NikSun NetVoice to monitor and analyze an IP link carrying H.323 Voice over IP (VoIP) traffic between the two Avaya

Communication Manager systems. With the appropriate configuration, NetVoice can successfully filter analyze the H.323 VoIP calls established between the Avaya IP telephones, Avaya Digital telephones, and analog telephones over the IP link.

9. Additional References

The following documents are relevant to these Application Notes:

1) *Administrator's Guide for Avaya Communication Manager*, Issue 8, June 2004, Document Number 555-233-506.
2) *NIKSUN NetVoice2005 User's Guide Version 2.0*
3) *NIKSUN NetVCR User's Guide Version 2005*

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