Agilent Technologies
N2X Testing SIP and VoIP Quality Using N2X

Application Note

Test Objective

Voice over IP (VoIP) overcomes some of the drawbacks of traditional circuit switched telephony by using network resources more efficiently. It is not necessary to establish a dedicated end-to-end circuit with reserved resources in order for two VoIP endpoints to communicate. Instead, packets are forwarded through the IP network in a connectionless manner, travelling along the path determined in real time by the routers and switches along the way. This delivery model involves a tradeoff between efficient network resource usage and quality of service (QoS) – best effort delivery of IP packets in a connectionless environment does not offer inherent mechanisms to guarantee reliable, timely delivery of VoIP information. Consequently, network design must take into consideration the properties of VoIP traffic, in particular its sensitivity to delay and loss, and ensure adequate QoS. Naturally, the only way to make certain that network designs will perform as expected once deployed is to test them using a tool such as N2X.

This application note shows how N2X can be used to test a device or network’s ability to provide adequate VoIP QoS. SIP sessions will be established over an underlying access protocol. We will measure the SIP scale and performance the device under test (DUT) can support. Once SIP sessions are in place, we will generate simulated VoIP service traffic. N2X will then be used to make delay and loss measurements on the traffic. MOS measurements will also be used in order to approximate the perceived VoIP quality and ultimately render a verdict on whether the DUT is delivering acceptable QoS.

Key Features to Highlight

► Highly scalable SIP emulation (10k active SIP sessions per test port)
► True Multiplay test solution: VoIP, Video and Data on the same port
► MOS estimates

Test Set-up – Network Diagram: Testing against a real DUT
**Intended Audience**

This test is relevant to engineers testing intelligent devices integrating some level of SIP or VoIP awareness. Typically, these will be edge routers with SBC functionality.

The N2X SIP/VoIP test solution will appeal to existing N2X customers seeking to augment their test capabilities to include VoIP. It will also be suitable for new customers testing VoIP as part of a Multiplay or multiplay scenario. Normally these will be “system test”, “integration test” or “POC lab” groups within NEMs, or “service validation” and “pre-deployment test” within CSPs.

**Equipment needed**

**N2X Equipment**
- Two N5551A/B “Tahu” Tri-rate Ethernet ports

**N2X Software Licenses**
- E7881B Packets and Protocols License
- N2X System Release 6.11 beta or later
- N5588A SIP license
- E7887A DHCP license

**Device under test**

This test will be of most interest if the DUT has some SIP awareness, although this is not strictly required. A registrar must be present in the SUT if AOR calling is to be used.

---

**Simple back to back setup scenario**

**Port Setup**

1. Click Ports from the main window top tool bar. In Port Selection dialog, select one test port, change its type to “Ethernet”. Select another test port. Change its type to “Ethernet”. Click “Ok”.

2. Click Physical Layer from the main window side tool bar.
3. Configure the ports selected
4. In the Physical Configuration dialog, change the Media Type to SPF if using optical connection, or RJ45 if using copper wire.
   
   *Note: For SPF, Step 5 and 6 must be performed. For other media type, skip step 5 and 6.*
5. Click Turn All Laser Off
6. Click Turn ALL Laser On
7. Close physical layer configuration

---

![Port and Port Type Selection](image-url)
Adding SIP emulation devices, establishing and terminate a call:

1. Select any port and click on “New” button on the right most corner of setup-emulation window.
2. The following screen will appear.
3. From SIP folder select SIP IPv4 device.
4. Choose the pool size to add by manipulating count value and click “OK”.
5. Properties dialog appears.
6. Configure general properties as shown below by clicking on General Tab. Configure Tester IP and SUT IP. In back to back mode SUT IP is configure as tester IP of the other device with which we want to establish a call.

Figure 2. Breadth of protocol coverage

Figure 3. Parametric definition of addresses makes large configurations easy
7. Click on SIP session tab to configure SIP specific properties as follows.

**TIP:** a single SIP device is capable of simulating multiple SIP subscriber’s by increasing the pool count.

8. Apply the configuration and click “OK”.

9. Follow steps 1 to 8 for the other port to add another SIP device (see Figure 5B).

![Figure 4. Configuring SIP specific properties](image1)

![Figure 5A. Add another SIP device](image2)

![Figure 5B. Configuring second SIP devices specific properties](image3)
10. Now enable both the devices. Click the device check box to enable the devices.

*Note that the device’s default state is “No Session”*

11. Configure the capture so that we can see the SIP signaling call flow and this can be used to debug any SIP signaling related problems.

   a. Click on the “Capture” button on the side bar.
   b. Tick the “Enable” check box for the ports you want to capture
   c. Click on “Tx & Rx Emulation”.
   d. Click “Start All Enabled” button to start the capture. See Figure 7.
   e. This will open the wire shark software and to capture the packets.

---

*Figure 6. Enable the device*

*Figure 7. Enable start capture*
12. Now we are ready to establish the call.

13. Select any of the sip device and click on “Action SIP IPv4”

14. And click on Establish to set up a call. The device selected will now act as caller user agent i.e user agent client (UAC) and the other device will act as callee user agent i.e If the call is established the device state will change to “Established”, if there is some delay in call setup users may also see a transition state “Establishing”

15. See the wire shark output to look at the signaling call flow.
To view SIP emulation statistics click any of the device and click “Results” button in setup emulation window. The statistics show cumulative stats for the entire device. Click on each measurement category and observe the description appearing at the bottom of the window to know about what they mean. Hold the mouse cursor on any individual stats field to get a tool tip explaining what that filed means. (Figures 11A, 11B, and 11C show statistics of the caller device after First establish)
To see the per instance level measurements click on the “Details” button next to Results. This window captures per instance level stats. Users can select columns to be displayed by clicking on column button on top corner of the details window (see Figure 12).

Figure 12. Customizing the stats view

Figure 13A. Per instance level stats

Figure 13B. Per instance level stats
18. Terminating call is also similar to establishing a call, go to Action SIP IPv4 or right click on a device to see terminate command. Click on “terminate” command to tear down a call. Users may select any of the devices to terminate a call.

Registration scenario and call using AOR:

For this exercise we have used Open source SIP proxy (called OPEN Ser) installed on a Linux Desktop as SUT. These in real test scenario can be replaced with SBC (session border controller) having SIP registration service or connected to an external SIP registrar. Because of limitation in h/w resources we have single port connected to the Linux box hence in the following exercise we will add both caller and callee SIP device pools on the same port. This in real test can be on different port as well.

Steps: Adding the sip device is similar as explained previously. These steps will only talk about the differences.

1. Same steps 1 to 8 explained above needs to be followed for caller and callee SIP devices.

2. Configure SIP general tab configuration as shown in following screen shot. Observe that the SUT IP is the IP of the machine where OPEN Ser proxy is installed.
3. Configure SIP session tab as shown below. Observe that call using AOR is enabled, sip proxy is enabled. Sip proxy is optional and is provided to match SIP word terminology. To configure the proxy, the SUT IP must be over rid with proxy IP. Also observe that this screen capture all media types supported by our SIP emulation. For successful call establishment the media types on both caller and callee device should be same.

4. Configure the capture as shown in the previous exercise.

5. Enable the device.

6. Invoke REGISTER command as shown in Figure 17.
7. Observe the wire shark capture for registration call flow. Observe the contact binding.

8. Open per instance detail window to observe the registration status of each instance in a device (similar to step 18 of call establishment exercise).

9. Steps one 1 to 8 should be followed for both caller and callee device.

10. Now establish the call using AOR.

11. Select any of the SIP device and send the establish command (similar to step 13 of call establishment exercise).

12. Observe the wire shark capture for the call flow between UAC->Proxy->UAS.
SIP IPv4 with DHCP client testing:

Most of the steps are similar as explained above (in Registration scenario) the differences are explained below. With this device since we don’t know the device IP until the device is enabled and get an IP address, we cannot do calls using static IP to SIP IPv4 with DHCP device. For this caller needs to know the AOR (address of record) of the SIP DHCP device and SIP DHCP device should have registered with a registrar. However we can call normal SIP IPv4 device from SIP IPv4 with DHCP client device in which case we need to configure the static IP address range in SIP session configuration SIP of DHCP device.

The following steps assume that calls are being established between two SIP DHCP devices.

1. Choose the appropriate SIP device (DHCP or PPPoE) as explained in step 2 (in b2b call establishment).

2. Need not configure the tester IP address in general tab.

3. DHCP Client configurations, the default configuration are sufficient to for the device to get an IP address. For advanced configuration please refer N2X help or consult DHCP emulation expert.

4. Execute the DHCP start command for the device to discover an IP address and bind to it. (Please refer DHCP help to know more about other DHCP related command). For SIP testing purpose basic IP address discovery is sufficient. See Figure 21.

5. SIP session configurations are as explained previously.

6. Register the clients with third party registrar (Open Ser) refer the Register scenario explained above to know about the proxy configuration.

7. Make sure that call using AOR is enabled.

8. Perform call establishment as shown in Figure 22.

Figure 21. IP address discovery

Figure 22. Call establishment
NAT Simulation:

This feature is to simulate the subscribers behind NAT/Firewall. The typical use scenario is an enterprise network protected by a NAT or Firewall. When NAT is enabled, SIP emulation uses the NATED IP configured as L3 (IP layer) source address and the tester IP configured as the internal IP for populating SIP header values. If a SBC (session border controller) supports NAT traversal then it can identify that a device is behind a NAT/firewall by examining the SIP packets headers and the L3 source address and typically maintain a pinhole for routing packets to devices behind NAT/Firewall. Following steps explain configuration of NAT in back to back scenario. Most of the steps are similar to call establishment scenario explained above only NAT specific configuration are shown in Figures 23 and 24.

1. General tab configuration of SIP device 1/port1 and device 2/port2.
2. Session tab configuration of sip device1/port and device2/port2.
3. Call flow capture in Wire shark.

Figure 24A. Session configuration of sip devices

Figure 24B. Session configuration of SIP devices. Simulating clients behind a NAT
Scale testing:

The steps are same as that of call establishment scenario explained above. The differences in configuration are explained here.

1. Setting the pool size. This is same as show previously in step 2 of call establishment scenario.

2. Make sure that all IP addresses are set to incrementing mode (where ever applicable).
   a. For calls using static IP addresses (Not using AOR) the remote host IP configuration needs to in incrementing mode.
   b. In back-to-back mode the SUT IP has to be in incrementing mode. This is not required if test is against an actual DUT /proxy.
   c. In NAT simulation scenario the NAT’ed IP has to be in incrementing mode.

3. Use of action control. Refer the Action Control section below to see how to use action control.
   a. If we do not use action control then all signaling may not go through the completion because of resource capacity constraints (i.e embedded CPU processing power). With Action control we have tested successful call establishment rate of 20 calls/seconds which good enough for most practical scenarios.
**Action Control:**

This feature is to control the rate of particular command type per specified time interval in scaled testing scenarios. This allows us verify/determine load capacity rating of SUT as specified by their data sheets (if any).

1. Select a SIP device.
2. Click on “Action SIP IPv4”
3. Click on “Action Control” to see Figure 26a, 26b, and 27.
   *Note: Be sure to control the rate so the DUT is not overleaded*
4. Configure command for which the action control to be applied and click on start to start generating the commands at the controlled rate.

---

**Figure 26A. Action control to verify/determine load capacity rating of SUT**

**Figure 26B. Action control to verify/determine load capacity rating of SUT**

**Figure 27. Configure action control command**
Resource Priority:

Refer to RFC 4412 for details. This feature can be used to assign priorities to the calls. This is basically to test if the SBC (SUT) can route the calls properly based on this priority or not.

Following screen shows the priority domain and levels supported by the N2X SIP emulation. This can be configured in sip session tab of device properties as shown in Figure 28.

Session Timer:

Refer RFC 4028 for details. This configuration enables us to configure the session refresh interval and who is supposed to take the refresher role (UAC or UAS).

Minimum Interval and desired interval:

Minimum interval is a UAS specific value i.e when UAS receives a INVITE request with Session-Refresh header value, the UAS compared it with this value. The values received is less than the value configured in minimum interval then the UAS responds 422 “Session interval too small” with Min-Se header set to this value. (RFC Default is 90 seconds, N2X defaults this value to 300 seconds)

Desired interval is a UAC specific value and is used to set Session-Refresh header in session refresh INVITE request. (RFC Default is 90 seconds, N2X defaults this value to 300 seconds) (See Figure 30).
**Delays in INVITE accept:**

This is UAS specific feature and is a time in seconds by which the UAS delays accepting the call. This is implemented to simulate the real time user behavior of accepting a call. If the caller (UAC) terminates the call before this timer expires (in UAS) then the call terminate command will generate the CANCEL request.

**SIP RFC Timer configuration:**

The SIP emulation provides options for user to configure base values of RFC 3261 timers i.e T1, T2 and T4.

---

![Figure 31. Invite accept](image1)

![Figure 32. Configure base values of RFC 3261 timers](image2)
VoIP Quick Test application:

The VoIP quick test application first configures the SIP emulation and traffic, then it initiates the call establishment, once the signaling is over (i.e. once the call is established) it starts the traffic.

Disclaimer: At the time of this writing the quick test module is still under development.

Quick test Configuration, Back-to-Back SIP/VOIP Call scenario:

- Click on Application in N2X main window.
- Go to tools->protocols->SetUp VoiceCall to open following screen (Figure 33)

1. Configure the Test Session by Clicking on the Configure Session Tab
2. Make sure the correct version is selected (6.11 System Release instead of latest)

Configure Tool Tab:
1. Click on Configure Tool Option.
2. Click on “Add Subscriber Pool” button to add N number of subscribers in a Subscriber Pool
3. Make sure a test session is connected before clicking on “Add Subscriber Pool” Button.
 ► Add Subscriber Pool Dialog

1. Select the Source Port and Sink Port to select after clicking on the “Interface Configuration Tab” of Add Subscriber Pool Dialog
2. Enter the number of Subscribers to configure in the Subscriber Pool
3. Currently only “IPV4 Version” is Supported and Port Type will be only “Ethernet”
4. To configure VLANs, click on the checkbox “VLAN Configuration”
5. Help Button:

 ► Address Configuration Tab:

1. Uncheck the “Default” CheckBox
2. Configure the Tester IP Address, Increment, Repeat
3. Configure the Static IP Address, Increment, as the SUT Address.

Figure 37. VLAN Configuration

Figure 38. Help menu

Figure 39. Configuration Tab
**SIP Configuration Tab:**

1. By clicking in the “Remote AOR” checkbox the remoteUser address will be remoteuser@domainB

2. By deselecting the “Remote AOR” CheckButton the remoteuser address will be remoteipaddress@domainB

3. Make Sure the SUT Address is same as Remote User IP Address, Increment, Repeat if the “Remote AOR” is unchecked.

4. Click “OK” Button

5. Users may edit the subscriber pool by clicking on “Edit Subscriber Pool”... button by selecting the row to be edited

6. Change the number of subscribers in pool from 1 to 11

7. Click “OK” Button

8. Remove subscriber pool by clicking on “Remove Subscriber Pool” button
Call Setup Tab:

1. Make Sure there were at least two Subscriber Pools are Added by “Add Subscriber Pool Button”

2. Click On “Add Call” Button and Select the Media Payload Type from the Media Payload Type Combo box

3. Click on Select Button and Select the Caller and Callee Subscriber Pools

4. Click “OK” Button

5. Multiple subscriber pools may be selected by clicking on “Add Call Options” and then clicking “OK”

6. Users may Enable/Disable the checkbox to select the caller and callee subscriber pools to be taken for the test run or not.

7. Users may Edit the Call Properties by “Edit Call” Button

8. Users may remove the call by selecting the row in the table and “Remove Call” Button
Test Control Tab

Users may optionally run the traffic by setting up the traffic configurations.

Set up Traffic Measurements:

Before starting the traffic, configure the N2X measurements to display the Packet Loss, Latency, Latency Variation, and MOS Estimate statistics:

1. From the left most panel on the N2X main screen, click on “Realtime” results, then bring up the “Setup Measurements” window by clicking the “Setup” button as shown in Figure 46:

2. From the “Measurements” tab, select the measurements:
   - Rx Packet Loss
   - Average Latency
   - Average Latency Variation
   - Maximum Latency Variation
   - Maximum Sampled Latency Variation
   - MOS
3. From the “Streams” tab, hi-light all the traffic streams and click the add button:

4. Change the measurement display mode from instantaneous to cumulative:

   Note: the MOS Estimate is not available in instantaneous mode.

► Start Test

1. Click on “Start Test” button to start the test with the current configuration

2. Click on “Stop Test” button to stop the Test

Figure 47. All the streams are selected

Figure 48. Changed measurement display mode
The Following is the TestLog for Back-To-Back RTP Traffic Generation Through QuickTest

06:09:28 - AgtTsuWaitForStopTest - Traffic and Measurement System stopped
06:09:28 - AgtTsuRemoveAllStreamGroups - removed 0 stream groups
06:09:28 - AgtTsuRemoveAllTraffic - removed 0 profiles
06:09:28 - Waiting for up to 60 seconds for the Routing Engine to stop...
06:09:28 - Routing Engine stopped
06:09:28 - Removing SIP/DHCP/IGMP/PPPOE related sessions from 2 ports in the test session.
06:09:28 - RemoveAllEmulationSessions - Removed SIP related sessions from ports: 104/3 104/4
06:09:28 - Add the Sip Device
06:09:28 - Creating the Sip Device
06:09:28 - Configuring General Tab
06:09:28 - Configuring the Sip Device
06:09:28 - Add the Sip Device
06:09:28 - Configuring General Tab
06:09:28 - Configuring the Sip Device
06:09:28 - Enabling Sip Device
06:09:28 - Enabling Sip Device
06:09:28 - Establishing a call
06:09:34 - Session handle 11 of emulation name sipIpv4 custom state Established count: 1
06:09:36 - All Sip instances are not established properly CustomCount:0,PoolSize:1
06:09:36 - Setting up the stream
06:09:36 - AgtTsuRemoveAllStatistics - removed 0 Statistic objects
06:09:36 - Statistics list = AGT_TEST_PACKETS_TRANSMITTED AGT_TEST_PACKETS_RECEIVED
06:09:36 - Setting up the stream
06:09:36 - AgtTsuRemoveAllStatistics - removed 1 Statistic objects
06:09:36 - Statistics list = AGT_TEST_PACKETS_TRANSMITTED AGT_TEST_PACKETS_RECEIVED
06:09:36 - AgtTsu: state changed to TEST_RUNNING
06:09:36 - AgtQtl: state changed to TEST_RUNNING
06:09:36 - AgtSga: state changed to TEST_RUNNING
06:09:36 - Starting traffic generator and measurement system...
06:09:37 - - test state = AGT_TEST_STOPPED
06:09:37 - - test state = AGT_TEST_STARTING
06:09:38 - Traffic and Measurement System started
06:09:39 - Measurements (1): TxPkts = 34.0000000000, RxPkts = 34.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:39 - Measurements (1): TxPkts = 34.0000000000, RxPkts = 34.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:40 - Measurements (2): TxPkts = 67.0000000000, RxPkts = 67.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:40 - Measurements (2): TxPkts = 67.0000000000, RxPkts = 67.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:41 - Measurements (3): TxPkts = 100.0000000000, RxPkts = 100.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:41 - Measurements (3): TxPkts = 100.0000000000, RxPkts = 100.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:41 - Measurements (3): TxPkts = 100.0000000000, RxPkts = 100.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:42 - Measurements (4): TxPkts = 134.0000000000, RxPkts = 134.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:42 - Measurements (4): TxPkts = 134.0000000000, RxPkts = 134.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:43 - Measurements (5): TxPkts = 167.0000000000, RxPkts = 167.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:43 - Measurements (6): TxPkts = 200.0000000000, RxPkts = 200.0000000000, Lost = 0.0, Tput(%) = 100.0
06:09:43 - Stopping test...
06:09:44 - - test state = AGT_TEST_RUNNING
06:09:45 - - test state = AGT_TEST_STOPPING
06:09:46 - - test state = AGT_TEST_STOPPING
06:09:47 - AgtTsuWaitForStopTest - Traffic and Measurement System stopped
06:09:47 - **** STOP requested (lib) ****
06:09:47 - AgtTsu: state changed to TEST_STOP_PENDING
06:09:47 - AgtQtl: state changed to TEST_STOP_PENDING
06:09:47 - AgtSga: state changed to TEST_STOP_PENDING
06:09:47 - AgtTsuWaitForStopTest - Traffic and Measurement System stopped
06:09:47 - AgtTsu: state changed to TEST_STOPPED
06:09:47 - AgtQtl: state changed to TEST_STOPPED
06:09:47 - AgtSga: state changed to TEST_STOPPED
06:09:47 - Terminating the call
06:09:47 - AgtTsu: state changed to TEST_STOPPED
06:09:47 - AgtQtl: state changed to TEST_STOPPED
06:09:47 - AgtSga: state changed to TEST_STOPPED
► **Registration**
To enable the registration in SIP emulation click on “External SIP Registrar” option as below and specify the address and expiry time.

► **Traffic Priority**
User can decide VLAN and IP Priority for the RTP Traffic.
1. Click on Traffic Configuration Tab to configure the IP Priority (Qos) for the RTP traffic
2. Users may select “Raw”, “Diff-Serv”, TOS radio button and based on the user selection the Rawpriority, Diff-Serv, TOS frames will be enable

---

**Figure 49. External SIP Registrar**

**Figure 50. VLAN and IP Priority for the RTP Traffic**

**Figure 51. Traffic Configuration Tab**
Remove all doubt

Our repair and calibration services will get your equipment back to you, performing like new, when promised. You will get full value out of your Agilent equipment throughout its lifetime. Your equipment will be serviced by Agilent-trained technicians using the latest factory calibration procedures, automated repair diagnostics and genuine parts. You will always have the utmost confidence in your measurements.

Agilent offers a wide range of additional expert test and measurement services for your equipment, including initial start-up assistance, onsite education and training, as well as design, system integration, and project management.

For more information on repair and calibration services, go to:

www.agilent.com/find/removealldoubt

Agilent Open

www.agilent.com/find/open

Agilent Open simplifies the process of connecting and programming test systems to help engineers design, validate and manufacture electronic products. Agilent offers open connectivity for a broad range of system-ready instruments, open industry software, PC-standard I/O and global support, which are combined to more easily integrate test system development.

LXI

www.lxistandard.org

LXI is the LAN-based successor to GPIB, providing faster, more efficient connectivity. Agilent is a founding member of the LXI consortium.