

VoIP calls and analysis using SIP

The following procedure provides instructions for using a T-BERD / MTS 5800 or a T-BERD / MTS 6000A with a MSAM to place and receive VoIP calls using SIP call control. Please read the entire procedure before beginning the test.

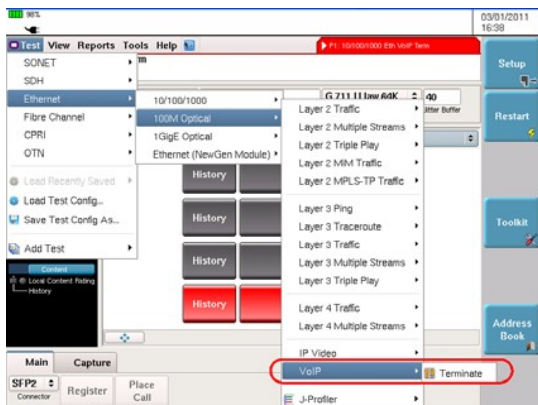
Topics discussed in this procedure include the following:

- “Configuring the test” on page 1
- “Registering with the server” on page 3
- “Placing and receiving calls” on page 3
- “Analyzing audio packets” on page 4

Configuring the test

Before you begin testing, make sure the test settings on the T-BERD / MTS 8000 match the settings of the line that you are testing.

Step 1 Use the Test Menu to select the VoIP application. For example, **Test>Ethernet>1GigE Optical>VoIP>Terminate**.



- Step 2** Select the **Setup** soft key, and then select the **Ethernet/IP** tab, and then, at a minimum, specify the following settings:
- Encapsulation - Select VLAN, Q-in-Q, or None.
 - VLAN ID - Specify the VLAN ID (if Encapsulation is VLAN).
 - Source IP and Gateway - IP address of the Source and the Gateway.
- Step 3** Select the **VoIP** tab, and then in the panel on the left side of the tab, select **General**.
- Step 4** Set the **Call Control Standard** to **SIP**, and then, specify the following settings:
- Source Alias - Enter the source phone number alias.
 - Outbound Alias - Select how to dial the destination: Dial by Phone Number or Dial by Name/URI/Email.
 - Dest. Phone Number - If you selected “Dial by Phone Number” for Outbound Alias, enter the destination phone number.
 - Dest. Name/URI/Email - If you selected “Dial by Name/URI/Email” for Outbound Alias, enter the destination name/URI/Email
 - SIP Vendor - Specify the vendor.
 - 100Rel Usage - Specify whether 100rel is required, supported, or disabled. 100 Rel provides reliable provisional response messages by appending the 100rel tag to the value of the required header of initial signaling messages.
- Step 5** In the panel on the left side of the tab, select **Proxy**, and then, at a minimum, specify the following settings:
- Proxy IP - The IP address of the proxy server.
 - User Name/Password - The user name and password used to access the proxy.
 - Proxy Port - The port number of the proxy.

Step 6 In the left panel on the side of the tab, **Audio/Codec**, and then, at a minimum, specify the following settings:

- Primary Codec - The codec type to be advertised/supported for receiving audio packets.

NOTE: The codec on the receiving and transmitting end should match in order for the call to be successful.

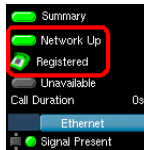
- Transmit Source - Select the transmit source: Voice conversation (transmits and receives live voice), IP voice announce (the unit repeats a sequence of words including the calling party's IP address), Tone (transmits the specified frequency).
- Silence Suppression - Specify whether silence suppression is supported.

Step 7 In the left panel on the side of the tab, select **QoS** and then specify the following thresholds: MOS Scaling, Jitter Threshold, Delay Threshold, Loss Threshold, and Content Threshold.

Step 8 Press the **RESULTS** key to return to the test menu.

Registering with the server

Before placing or receiving calls, you must register with the Proxy server. The instrument automatically deregisters and registers with the server after a change in call settings. After successful registration, the "Register" button will turn yellow and change to "Registered" and the Stack status indicator in the LED panel updates.



Placing and receiving calls

After specifying configuration settings and registering with the server, you can place a VoIP call.

- Step 1** To place a call, select the **Place Call** action button. The button label changes to Hang Up. After the call is connected, the Call status in the LED panel will update and the timer begins counting.
- Step 2** To receive a call manually, press the **Receive Call** soft key when an incoming call is indicated.
- To answer calls automatically, select **Auto Answer Enabled** from the VoIP General Settings menu. When a call is received, the instrument answers the call with a voice announcement, and then basic call statistics are provided.
- While the call is up, observe the Transport and Content result categories.
- Step 3** Select the **Hang up** action button to end the call.

Analyzing audio packets

Assuming the instrument is equipped with the capture option, audio packets captured with the VoIP application and viewed using Wireshark or on an offline PC. When capturing packets in the VoIP application, it is recommended that you do not save the captured packets until the call is ended (the phone is on hook).

Related information

Use this Quick Card in conjunction with the *MSAM, CSAM, and T-BERD / MTS 5800 Ethernet Testing Guide* that shipped with your instrument.

Technical assistance

For the latest TAC information, go to www.jdsu.com or contact your local sales office for assistance. Contact information for regional sales headquarters is listed at the bottom of this Quick Card.

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