EXpert VoIP Test Tools

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Contents

1 Introducing the EXpert VoIP Test Tools

The EXpert VoIP Test Tools provides Ethernet testing capability on the EXFO's platform. It runs as a separate software application on the platform and operates independently of the modules installed on the platform. The EXpert VoIP Test Tools application generates a single VoIP call from the platform to another EXFO's platform supporting the application or any IP phone. The call can either be "live", "stored audio" or "synthesized media". Several signaling technologies are supported including SIP, SCCP, H.323, and H.248 to support the majority of VoIP environments currently used in the industry. Resulting quality metrics include MOS, R-factor as well as numerous RTP loss metrics.

The EXpert VoIP Test Tools application runs on the following EXFO's platforms: FTB-200v2, FTB-1, FTB-1v2, FTB-1v2 Pro, FTB-2, FTB-2 Pro, and FTB-4 Pro.

Conventions

Before using the product described in this guide, you should understand the following conventions:

WARNING

Indicates a potentially hazardous situation which, if not avoided, could result in *death or serious injury*. Do not proceed unless you understand and meet the required conditions.

CAUTION

Indicates a potentially hazardous situation which, if not avoided, may result in *minor or moderate injury*. Do not proceed unless you understand and meet the required conditions.

CAUTION

Indicates a potentially hazardous situation which, if not avoided, may result in *component damage*. Do not proceed unless you understand and meet the required conditions.

IMPORTANT

Refers to information about this product you should not overlook.

2 *Getting Started*

Signal Connection

The EXpert VoIP Test Tools uses the 10/100/1000 Mbit/s Ethernet RJ45 port of the EXFO's platform.

Connect the 10/100/1000 Mbit/s electrical signal to be tested to the RJ45 port of the EXFO's platform.

Note: *Refer to the EXFO's platform User Guide for more information.*

Laser Safety Information

The EXpert VoIP Test Tools software is not provided with any hardware components. However, it may be used with your platform or modules which may contain laser components. Refer to the user guides of your platform or modules for further laser safety details and instructions.

Starting the EXpert VoIP Test Tools Application

The EXpert VoIP Test Tools application is pre-installed on the EXFO's platform. If the EXpert VoIP Test Tools is not already installed, refer to the EXFO's platform User Guide for more information on how to install the application.

Note: *The application can be used in trial mode if you do not have a valid license. You can follow the on-screen instructions to activate the trial. If you wish to renew the trial period or purchase a valid license, then refer to EXFO website. For information on how to install and activate software options, refer to the EXFO's platform User Guide.*

To start the EXpert VoIP Test Tools application:

1. **FTB-1**: From **Mini ToolBox**, tap on the **Test Tools** tab.
   FTB-1v2 and **FTB-1v2 Pro**: From **Mini ToolBox X**, tap on the **Test Tools** button.
   FTB-2, **FTB-2 Pro**, and **FTB-4 Pro**: From **ToolBox X**, tap the **Test Tools** button.
   FTB-200v2: From **Compact ToolBox**, tap on the **Test Tools** tab.

2. Select the **EXpert VoIP Test Tools** then tap on **Start**. At startup, the application always loads the previous configuration.

3 *Using the Graphical User Interface*

This chapter describes the graphical user interface of the EXpert VoIP Test Tools application.

Main Application Window

The following main application window is displayed when the EXpert VoIP Test Tools application is started.
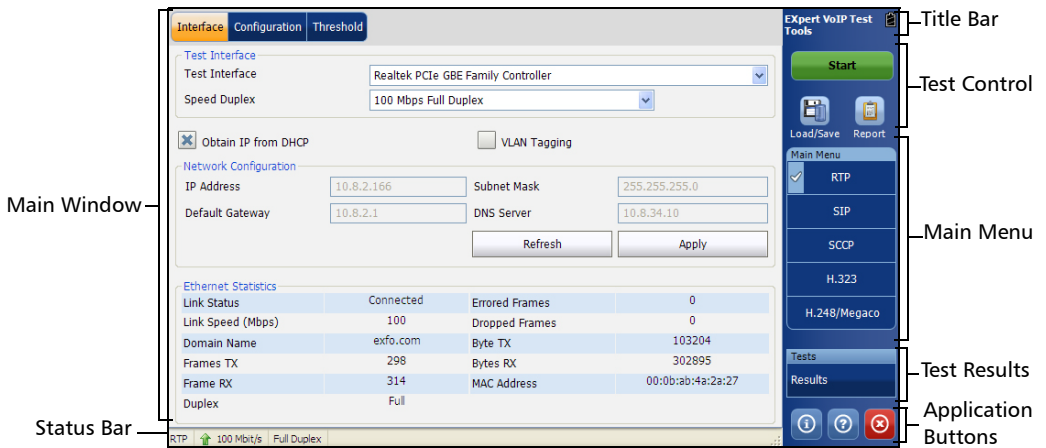
Main Window

The main window allows to setup a test and to view the test status and results.

Status Bar

The status bar displays the following information.

Icon and/or text	Description
Test name	Test name of the selected test.
Link arrow	Green arrow: Ethernet Link up Red arrow: Ethernet Link down.
Interface speed	10 Mbit/s, 100 Mbit/s, or 1000 Mbit/s.
Test status	Current test status.

Title Bar

The title bar displays the software application name and the battery level indicator.

Test Control

Note: *Refer to* Test Control *on page 85 for more information.*

Main Menu

The EXpert VoIP Test Tools main menu allows the selection of a test tool. The selected test tool is the one with a check mark to its left. Refer to *Selecting and Starting a Test* on page 11.

Test Results

Note: *Refer to* Test Results *on page 53 for more information.*

Application Buttons

➤ **About** (i) mainly displays the product version details and technical support information.

Software Options button displays the list of software options enabled. For information on how to install and activate software options, refer to the EXFO's platform User Guide.

Option Name	Description
EXPERTVOIP	Enable "VoIP call test tool" application
EXPERTSIP	SIP call signaling support for EXpert VoIP Test Tools
EXPERTSCCP	SCCP call signaling support for EXpert VoIP Test Tools
EXPERTH.323	H.323 call signaling support for EXpert VoIP Test Tools
EXPERTH.248	H.248/Megaco call signaling support for EXpert VoIP Test Tools

Licence Agreement button displays the details of the product licence agreement.

➤ **Help** (?) displays the help information related to the content of the active main window. It is also possible to navigate through the remainder of the help information.

➤ **Exit** (x) closes the application.

Arrow Buttons

⊼	Moves to the top of the list.
⯅⯅	Moves one page up.
▲	Moves one line up.
▼	Moves one line down.
⯆⯆	Moves one page down.
⊻	Moves to the end of the list.

Keyboard Usage

The GUI pops up different keyboards to modify data. Following are the usual keyboard keys:

➤ Left arrow moves the cursor one position to the left.

➤ Right arrow moves the cursor one position to the right.

➤ Up arrow increases the byte value by one.

➤ Down arrow decreases the byte value by one.

➤ **Del** deletes the value at the cursor position.

➤ **Back** deletes the value preceding the cursor position.

➤ **OK** completes data entry.

➤ **Cancel** closes the keyboard and discards the keyboard entry.

➤ **Previous...** or **Presets** allows the selection of previously configured values. This button is only available for certain fields like IP Address, MAC Address, etc.

Note: *For certain text fields, the GUI pops up or uses the platform's on-screen keyboard. Refer to the platform user guide for more information on how to used it.*

For full keyboard, the **Back**, **Del**, **Shift**, and **Space** bar keys have the same functionality as a regular PC keyboard.

4 *Selecting and Starting a Test*

A test can be created either by selecting the test from the **Test Menu** or by loading a previously saved configuration (refer to *Load/Save* on page 86 for more information).

The EXpert VoIP Test Tools main menu offers the following test tools:

Test Tools
RTP on page 12
SIP on page 13
SCCP on page 15
H.323 on page 17
H.248/Megaco on page 18

RTP

The RTP (Real Time Protocol) network test is a peer-to-peer test that measures the VoIP related parameters by streaming RTP packets between two endpoints.

The RTP network test simulates VoIP traffic by streaming RTP packets between a Controller test set, which initiates the packet stream, and a Responder endpoint, which initiates its own stream to the Controller test set.

The Controller sends an audio packet stream to the Responder. The audio stream is based on the type of traffic parameter (and, optionally, test length parameter). Port selection is determined by the setting of the UDP Port Parameters.

The Responder initiates a packet stream to the Controller. The Controller receives the packets that are sent from the Responder.

To select, configure, and start a RTP Test:

1. From the **Main Menu**, tap **RTP**.

2. From the **Interface** tab, configure the test interface parameters. See *Interface* on page 20. Ensure that the link is up in the status bar before proceeding to the next step.

3. From the **Configuration** tab, configure the RTP test. See *Configuration (RTP)* on page 23.

4. From the **Threshold** tab, enable and configure the threshold values. See *Threshold* on page 52.

5. Press **Start** to start the test.

6. For results, refer to *Summary (RTP)* on page 76, *Controller Statistics* on page 65, and *Responder Statistics* on page 68.

7. If required, tap the **Report** button to generate the report file of the results and statistics. Refer to *Report* on page 88 for more information.

SIP

The SIP (Session Initiation Protocol) test measures the performance of Voice over IP (VoIP) services within a network. The test uses the SIP for call signaling and the RTP for digitally transporting the encoded audio.

The SIP signaling portion of the test can be transported over UDP/TCP. SIP calls transported over UDP/TCP can be routed through a proxy server. The test's RTP functionality conforms to RFC 3550.

The SIP test measures call statistics for a SIP call and initiates a call to or answers a call from a given endpoint. It reports call setup and media path statistics between the test set and a SIP endpoint or another EXpert VoIP SIP enabled test set.

The SIP test operates in two modes.

➤ Network Active Test

➤ Service Active Test

The SIP network active test is performed, if EXFO devices (i.e. EXpert VoIP Test Tools devices) are selected on both ends, that is, on the Controller side and the endpoint. The SIP network active test operates peer-to-peer in UDP communication mode along with third-party SIP proxy servers to set up calls between two endpoints, a Controller, which initiates the call, and a Responder, which is responsible for completing the call and, optionally, echoing media packets sent from the Controller. For the SIP network active test in TCP communication mode third-party SIP proxy servers are mandatory. A SIP test cycle consists of the initiation and termination of an audio session (simulated audio data) between two endpoints for which it reports call setup and audio stream statistics.

The SIP test measures the call statistics. It initiates and answers a SIP call from a given endpoint. It reports call setup and media path statistics between the test set and a SIP endpoint or another EXpert VoIP SIP enabled test set.

To select, configure, and start a SIP Test:

1. From the **Main Menu**, tap **SIP**.

2. From the **Interface** tab, configure the test interface parameters. See *Interface* on page 20. Ensure that the link is up in the status bar before proceeding to the next step.

3. From the **Configuration** tab, configure the SIP test. See *Configuration (SIP)* on page 26.

4. From the **Controller Setup** tab, configure the controller settings. See *Controller Setup* on page 42.

5. From the **Responder Setup** tab, configure the responder settings. See *Responder Setup* on page 45.

6. From the **Threshold** tab, enable and configure the threshold values. See *Threshold* on page 52.

7. Press **Start** to start the test.

8. For results, refer to *Summary (SIP)* on page 81, *Call Details* on page 54, and *Controller Statistics* on page 65.

9. If required, tap the **Report** button to generate the report file of the results and statistics. Refer to *Report* on page 88 for more information.

SCCP

The SCCP (Skinny Client Control Protocol) service active test measures the performance of calls between two endpoints and a SCCP peer endpoint made through the Cisco Call Manager.

In the SCCP test, the Controller acts as a calling endpoint (Caller), and the Responder acts as an answering endpoint (Callee). Both endpoints must be registered in the CallManager. Devices registered in the CallManager are assigned a telephone number associated with the MAC address of the device.

Note: *Each endpoint executing a test against a given CallManager must have a unique MAC address-phone number pair for that CallManager.*

Note: *The Caller and the Callee must be manually registered with the CallManager with respect to their phone type before you configure the test. For executing the call successfully, the Callee's phone number must be a valid number from the CallManager's registration tables, and it should be associated with the Responder's MAC address.*

To select, configure, and start a SCCP Test:

1. From the **Main Menu**, tap **SCCP**.

2. From the **Interface** tab, configure the test interface parameters. See *Interface* on page 20. Ensure that the link is up in the status bar before proceeding to the next step.

3. From the **Configuration** tab, configure the SCCP test. See *Configuration (SCCP)* on page 32.

4. From the **Parameters** tab, configure the call, phone, and media parameters. See *Parameters (SCCP)* on page 48.

5. From the **Threshold** tab, enable and configure the threshold values. See *Threshold* on page 52.

6. Press **Start** to start the test.

7. For results, refer to *Summary (SCCP)* on page 79, *Call Quality Details* on page 62, and *Call Manager Statistics (SCCP)* on page 60.

8. If required, tap the **Report** button to generate the report file of the results and statistics. Refer to *Report* on page 88 for more information.

H.323

The H.323 test is a set-to-service test that measures call statistics for an H.323 call. It either makes a H.323 call to a given destination (directly to an endpoint, or through the gatekeepers and/or gateways that you specify) or answers a call made to it from an endpoint, and reports call setup and media path statistics. The test assumes that you know the destination (specified using an IP address, telephone number, or H.323 ID), and that the destination will answer the call and will not hang up during the configured call duration. The H.323 test supports versions 1 through 4 of the H.323 protocol.

To select, configure, and start a H.323 Test:

1. From the **Main Menu**, tap **H.323**.

2. From the **Interface** tab, configure the test interface parameters. See *Interface* on page 20. Ensure that the link is up in the status bar before proceeding to the next step.

3. From the **Configuration** tab, configure the H.323 test. See *Configuration (H.323)* on page 35.

4. From the **Parameters** tab, configure the local, and media parameters. See *Parameters (H.323)* on page 49.

5. From the **Threshold** tab, enable and configure the threshold values. See *Threshold* on page 52.

6. Press **Start** to start the test.

7. For results, refer to *Summary (H.323)* on page 73, *Call Quality Details* on page 62, and *Call Manager Statistics (H.323)* on page 58.

8. If required, tap the **Report** button to generate the report file of the results and statistics. Refer to *Report* on page 88 for more information.

H.248/Megaco

The H.248, the Media Gateway Control Protocol, is designed for distributed media gateway systems in which one or more centralized gateway controllers (Call Agents) control one or more media gateways.

The VoIP H.248 test measures the performance of the services of H.248 based VoIP networks. It either initiates a call to or answers a call from other endpoints based on a telephone number. Performance is measured for both the call signaling path (H.248 signaling) and the media path (RTP/RTCP). Each set of measurements is performed during a test cycle. A test cycle consists of the initiation and termination of a single, bi-directional RTP audio session between the test set and a third-party device.

To select, configure, and start a H.248/Megaco Test:

1. From the **Main Menu**, tap **H.248/Megaco**.

2. From the **Interface** tab, configure the test interface parameters. See *Interface* on page 20. Ensure that the link is up in the status bar before proceeding to the next step.

3. From the **Configuration** tab, configure the H.248/Megaco test. See *Configuration (H.248/Megaco)* on page 39.

4. From the **Parameters** tab, configure the local parameters. See *Parameters (H.248/Megaco)* on page 51.

5. From the **Threshold** tab, enable and configure the threshold values. See *Threshold* on page 52.

6. Press **Start** to start the test.

7. For results, refer to *Summary (H.248/Megaco)* on page 71, *Call Quality Details* on page 62, and *Call Manager Statistics (H.248/Megaco)* on page 56.

8. If required, tap the **Report** button to generate the report file of the results and statistics. Refer to *Report* on page 88 for more information.

5 *Test Setup*

The test setup offers the following tabs depending on the test:

Tabs	Test					Page
	RTP	**SIP**	**SCCP**	**H.323**	**H.248**	
Interface	X	X	X	X	X	20
Configuration	23	26	32	35	39	<---
Controller Setup		X				42
Responder Setup		X				45
Parameters			48	49	51	<---
Threshold	X	X	X	X	X	52

Interface

Note: *The interface configuration parameters apply globally to all tests.*

From the **Main Menu**, tap on a test and the **Interface** tab.

Test Interface

➤ **Test Interface** displays the network adapter(s) found on the EXFO's platform. The first network adapter is selected by default.

➤ **Speed/Duplex** selects the network adapter speed and duplex mode: **Auto Negotiation** (default), **10 Mbit/s Half Duplex**, **10 Mbit/s Full Duplex**, **100 Mbit/s Half Duplex**, **100 Mbit/s Full Duplex**, **1000 Mbit/s Full Duplex**. Changing this setting restarts the network adapter, a user confirmation is required.

Obtain IP from DHCP

Obtain IP from DHCP check box when selected (default) allows to dynamically obtain an IP address from a DHCP (Dynamic Host Configuration Protocol) server for the Ethernet port.

VLAN Tagging

Only available on FTB-200v2 and FTB-1 platforms, allows to enable or disable VLAN Tagging.

Note: *VLAN Tagging enable/disable will reboot the device. A warning pop-up message is displayed. Press Yes to reboot the platform. The EXpert VoIP Test Tools application has to be manually restarted after the platform's reboot.*

Network Configuration

Note: *Network configuration is only possible when the **Obtain IP from DHCP** check box is cleared. Otherwise, when the **Obtain IP from DHCP** check box is selected, all parameters are not configurable and set to the values obtained through DHCP process.*

➤ **IP Address** allows to enter the IP address for the Ethernet port: **0.0.0.1** to **223.255.255.255**.

➤ **Subnet Mask** allows to enter the Subnet Mask for the Ethernet port: **0.0.0.1** to **255.255.255.255**.

➤ **Default Gateway a**llows to enter a default Gateway address: **0.0.0.01** to **255.255.255.255**.

➤ **DNS Server** allows to enter the DNS Server address: **0.0.0.0** to **223.255.255.255**.

➤ **Refresh**, available when the **Obtain IP from DHCP** check box is selected, refreshes the network configuration parameters.

➤ **Apply** allows to apply the network configuration settings.

Ethernet Statistics

The following Ethernet Statistics are displayed.

➤ **Link Status** displays the status of the link: **Connected** or **Disconnected**.

➤ **Link Speed (Mbps)** displays the speed of the Ethernet connection.

➤ **Domain Name** displays the name of the connected domain.

➤ **Frame TX** displays the total Ethernet frames transmitted.

➤ **Frame RX** displays the total Ethernet frames received.

➤ **Duplex** displays the Duplex mode of the link: **half duplex**, **full duplex**, or **auto**.

➤ **Errored Frames** displays the number of Ethernet frames received with error.

➤ **Dropped Frames** displays the number of Ethernet frames dropped.

➤ **Bytes TX** displays the total data bytes transmitted.

➤ **Bytes RX** displays the total data bytes received.

➤ **MAC Address** displays the MAC Address of the EXFO's platform.

Note: *Bytes TX and Bytes RX indicate the total amount of data that has passed through the Ethernet interface either way. EXpert VoIP Test Tools update the Ethernet statistics every second.*

Configuration (RTP)

From the **Main Menu**, tap **RTP** and the **Configuration** tab.

Call Mode

Select the call mode:

➤ **Controller** establishes a control communications channel with the **Responder**. It also sends the configuration details to the **Responder**. By default, the **Controller** is selected.

➤ **Responder** (default) is the endpoint, which initiates its own stream to the **Controller** test set. When **Responder** is selected, all other configuration parameters are disabled.

Test Length

Select the test length:

➤ **Test Duration (sec)** is the length of time the test will run in seconds: **1** to **600 seconds** (default is **10 seconds**).

➤ **Number of Packets** is the number of RTP packets to exchange between the controller and the responder: **2** to **65535** (default is **500**).

Call Information

➤ **Media Type**

➤ **Live Call** (default) specifies that actual call is established between the Controller and the Responder.

➤ **Synthesized Media** specifies that EXpert VoIP Test Tools application automatically generates a randomly RTP data.

➤ **Codec** represents the audio **Codec** (packet interval, payload size, and IP bandwidth) of the call simulated by the Controller.

For **Live Call** media type: **G.711 μlaw**, and **G.711 aLaw** (default).

For **Synthesized Media** type: **G.711 μlaw**, **G.711 aLaw** (default), **G.722**, **G.723.1**, **G.726**, **G.728**, **G.729**, **G.729A**, **G.729B**, and **G.729AB**.

➤ **IP Media DSCP**[1] (IP Media Differentiated Services Code Point) describes a set of end-to-end QoS (Quality of Service) capabilities. End-to-end QoS is the ability of the network to deliver service required by the specific network traffic from one end of the network to the other end. The accepted range is from **0** (default) to **63**.

Destination IP address

➤ **IP Address**

➤ **Manually Configure IP** allows to configure the destination IP address manually: **0.0.0.1** to **223.255.255.255**.

➤ **Auto Discovery List** allows to select the IP address from a list of IP addresses discovered by the application.

Utilities

➤ **Traceroute** check box when selected (default) calculates the performance of the network path between two endpoints of a running test every time the test is run.

➤ **Capture** check box when selected (cleared by default) captures the RTP packet stream for the running test. Once test is completed, the capture file is available and can be opened in the Wireshark application from the **Load/Save** button; refer to *Packet Capture* on page 87 for more information.

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1. Not supported on FTB-2 and FTB-2 Pro.

➤ **Record** check box when selected (cleared by default) allows recording network VoIP traffic. The recorded file can be listened from the **Load/Save** option; refer to *Record Audio Tab* on page 87 for more information. **Record** is only available for **Live Call** media type.

Capture Filters

Available when the **Utilities** - **Capture** check box is selected.

➤ **IP Address** check box when selected (cleared by default) represents the Peer **IP Address** that acts as a filter that will capture packets having the source or destination IP address matching the specified Peer IP address. The accepted range is from **0.0.0.1** to **223.225.225.225**.

➤ **File Size** represents the maximum size of the file to capture either in megabytes (**1** to **50 MB** (default is 5 MB)) or in seconds (**1** (default) to **3600** seconds).

Configuration (SIP)

From the **Main Menu**, tap **SIP** and the **Configuration** tab.

Peer Device

➤ **EXFO Device**: Select EXFO Device option on both the endpoints (i.e. Controller side and the Responder side) to run the SIP Network Active Test.

➤ **Other Device** (default): Select the Other Device option to run the service active test. In service active test, the test set can either act as controller or responder with the other end being the SIP endpoint. By default, Other Device is selected.

Call Mode

➤ **Controller** when selected (default) establishes a control communications channel with the Responder and also sends the configuration details to the Responder.

➤ **Responder**: If selected all the other configurations are disabled (only for Network Active test). In the Service Active test, the configurations are available at responder side also. The Responder is the endpoint, which initiates its own stream to the Controller test set.

Communication Channel

➤ **TCP** selects the Transmission Control Protocol (TCP) for the transmission of the data on the network.

➤ **UDP** selects the User Datagram Protocol (UDP) for the transmission of the data on the network.

Note: *Communication channel (TCP/UDP) is used for SIP signaling packets only. Voice data will pass through UDP only even though TCP is selected. SIP calls over TCP must be routed through a proxy server.*

Test Length

➤ **Test Duration** (sec) is the length of time for which the test will run in seconds: **1** to **600** seconds (default is **10** seconds). **Test Length** is not available when **Stored Audio File** media type is selected.

➤ **Number of Packets** is the number of RTP packets to exchange between the Controller and the Responder: **2** to **65535** (default is **500**).

Call Information

➤ **Media Type**

➤ **Live Call** (default) specifies that actual call is established between the Controller and the Responder.

➤ **Stored Audio File** specifies that any stored media file on the drive is transmitted from Controller to Responder and the results are recorded.

➤ **Synthesized Media** specifies that the EXpert VoIP Test Tools application automatically generates a randomly RTP data.

➤ **Codec** selects the audio Codec (packet interval, payload size, and IP bandwidth) of the call simulated by the Controller.

For **Live Call** media type: **G.711 μlaw** (default), and **G.711 aLaw**.

For **Audio File** media type: **G.711 μlaw** (default), **G.711 aLaw**, **G729**, **G.729A**, **G.729B**, and **G.729AB**.

For **Synthesized Media** type: **G.711 μlaw** (default), **G.711 aLaw**, **G.722**, **G.723.1**, **G.726**, **G.728**, **G.729**, **G.729A**, **G.729B**, and **G.729AB**.

➤ **Call Type** selects the type of the call. For **Peer Device** set as **EXFO Device**, only **SIP Call** is available. In Network Active test, the supported call type is **SIP Call**.

  ➤ **SIP Call** attempts to establish a single call with a single audio session.

  ➤ **SIP Options Request** attempts a single "options" request method with the specified endpoint. Use this mode to test the availability of the SIP endpoint without making a complete call.

  ➤ **SIP Proxy Registration** attempts a single proxy "register" request method with the specified registrar domain and proxy server.

  ➤ **SIP Call After Proxy Registration** acts as a SIP Call, but includes data acquired during registration with a specified registrar domain and proxy server.

➤ **PRACK** (Provisional Response ACKnowledgement) method is used to acknowledge receipt of reliably transported provisional responses (1xx). The reliability of 2xx, 3xx, 4xx, 5xx, and 6xx responses to INVITE requests are achieved using the ACK method. However, in cases where a provisional response such as 180 ringing is critical in determining the call state, it may be necessary for the receipt of a provisional response to be confirmed. The PRACK method applies to all provisional responses except for the 100 trying response which is never reliably transported.

For Controller Call Mode:

➤ **Disabled** (default) doesn't use the PRACK method.

➤ **Supported** does not insist on usage of reliable provisional responses. However, the UAC supports the reliable provisional responses in case where the UAS needs to send one. The UAC creates a request by inserting a Supported header field into the request.

➤ **Required** insists on reliable delivery of provisional responses. The UAC creates a request by inserting a Require header field into the request.

For Responder Call Mode:

➤ **PRACK** check box when selected (default) allows to respond to Provisional Response ACKnowledgement by providing reliable provisional response messages.

➤ **SIP DSCP**[1] (SIP Differentiated Services Code Point) describe a set of QoS (Quality of service) capabilities. The accepted range is from **0** (default) to **63**.

➤ **IP Media DSCP**[1] (IP Media Differentiated Services Code Point) describes a set of end-to-end QoS (Quality of Service) capabilities. End-to-end QoS is the ability of the network to deliver service required by the specific network traffic from one end of the network to the other end. The accepted range is from **0** (default) to **63**.

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1. Not supported on FTB-2 and FTB-2 Pro.

Destination IP address

Only available when the **Peer Device** is set as **EXFO Device**.

➤ **Manually Configure IP** allows to configure the destination IP address manually.

➤ **Auto Discovery List** allows to select the IP address from a list of IP addresses discovered by the EXpert VoIP Test Tools.

➤ **IP address** allows to either enter the IP address manually (from **0.0.0.1** to **223.255.255.255**) or from a list.

Utilities

➤ **Capture** check box when selected (cleared by default) captures the RTP packet stream for the running test. Once test is completed, the capture file is available and can be opened in the Wireshark application from the **Load/Save** option; refer to *Packet Capture* on page 87 for more information.

➤ **Record** check box when selected (cleared by default) allows recording network VoIP traffic. The recorded file can be listened from the **Load/Save** option; refer to *Record Audio Tab* on page 87 for more information. **Record** is only available for **Live Call** media type.

Capture Filters

Available when the **Utilities - Capture** check box is selected.

➤ **IP Address** check box when selected (cleared by default) uses the Peer **IP Address** that acts as a filter that will capture packets having the source or destination IP address matching the specified Peer IP address. The accepted range is from **0.0.0.1** to **223.225.225.225**.

➤ **File Size** represents the maximum size of the file to capture either in megabytes (**1** to **50 MB** (default is 5 MB)) or in seconds (**1** (default) to **3600** seconds).

Configuration (SCCP)

From the **Main Menu**, tap **SCCP** and the **Configuration** tab.

Call Mode

➤ **Make Call**: The Caller establishes a control communications channel with the Receiver. By default, Make Call is selected.

➤ **Answer Call**: It is the endpoint, which initiates its own stream to the Caller test set.

Test Length

Select the test length. Not available when **Stored Audio File** is selected as the media type.

➤ **Test Duration (sec)** is the length of time the test will run in seconds: **1** to **600 seconds** (default is **10 seconds**).

➤ **Number of Packets** is the number of RTP packets to exchange between the controller and the responder: **2** to **65535** (default is **500**).

Call Information

➤ **Media Type**

➤ **Live Call** (default) specifies that actual call is established between the Controller and the Responder.

➤ **Stored Audio File** specifies that any stored media file on the drive is transmitted from Controller to Responder and the results are recorded.

➤ **Synthesized Media** specifies that the EXpert VoIP Test Tools application automatically generates a randomly RTP data.

➤ **Codec** selects the audio Codec (packet interval, payload size, and IP bandwidth) of the call simulated by the Caller.

For **Live Call** media type: **G.711 μlaw** (default), and **G.711 aLaw**.

For **Audio File** media type: **G.711 μlaw** (default), **G.711 aLaw**, **G729**, **G.729A**, **G.729B**, and **G.729AB**.

For **Synthesized Media** type: **G.711 μlaw** (default), **G.711 aLaw**, **G.722**, **G.723.1**, **G.726**, **G.728**, **G.729**, **G.729A**, **G.729B**, and **G.729AB**.

➤ **SCCP DSCP**[1] (SCCP Differentiated Services Code Point) describes a set of end-to-end QoS (Quality of Service) capabilities. End-to-end QoS is the ability of the network to deliver service required by the specific network traffic from one end of the network to the other end. The accepted range is from **0** (default) to **63**.

Destination Information

Destination Phone Number is the phone number associated with the MAC address for the Callee as specified in the CallManager registration tables. This number is dialed by the Caller at the start of each test. The accepted range is from **1** to **9999999999**. Only available when **Make Call** is selected as the call mode.

Note: *The destination phone number must be assigned by the CallManager administrator in order to ensure compliance with the network standard.*

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1. Not supported on FTB-2 and FTB-2 Pro.

Utilities

➤ **Traceroute** check box when selected (default) calculates the performance of the network path between two endpoints of a running test every time the test is run.

➤ **Capture** check box when selected (cleared by default) captures the RTP packet stream for the running test. Once test is completed, the capture file is available and can be opened in the Wireshark application from the **Load/Save** option; refer to *Packet Capture* on page 87 for more information.

➤ **Record** check box when selected (cleared by default) allows recording network VoIP traffic. The recorded file can be listened from the **Load/Save** option; refer to *Record Audio Tab* on page 87 for more information. **Record** is only available for **Live Call** media type.

Capture Filters

Available when the **Utilities** - **Capture** check box is selected.

➤ **IP Address** check box when selected (cleared by default) represents the Peer **IP Address** that acts as a filter that will capture packets having the source or destination IP address matching the specified Peer IP address. The accepted range is from **0.0.0.1** to **223.225.225.225**.

➤ **File Size** represents the maximum size of the file to capture either in megabytes (**1** to **50 MB** (default is 5 MB)) or in seconds (**1** (default) to **3600** seconds).

Configuration (H.323)

From the **Main Menu**, tap **H.323** and the **Configuration** tab.

Call Mode

➤ **Make Call**: The Caller establishes a control communications channel with the Receiver. By default, Make Call is selected.

➤ **Answer Call**: It is the endpoint, which initiates its own stream to the Caller test set.

Call Information

➤ **Media Type**

 ➤ **Live Call** (default) specifies that actual call is established between the Caller and the Callee.

 ➤ **Synthesized Media** specifies that the EXpert VoIP Test Tools application automatically generates a randomly RTP data.

➤ **Codec** selects the audio Codec (packet interval, payload size, and IP bandwidth) of the call simulated by the Caller.

For **Live Call** media type: **G.711 μlaw** (default), and **G.711 aLaw**.

For **Synthesized Media** type: **G.711 μlaw** (default), **G.711 aLaw**, **G.722**, **G.723.1**, **G.726**, **G.728**, **G.729**, **G.729A**, **G.729B**, and **G.729AB**.

Note: *If the call endpoint does not support the specified codec, the call fails.*

DSCP[1]

IP Media DSCP (IP Media Differentiated Services Code Point) describes a set of end-to-end QoS (Quality of Service) capabilities. End-to-end QoS is the ability of the network to deliver service required by the specific network traffic from one end of the network to the other end. The accepted range is from **0** (default) to **63**.

Call Duration

Call Duration (sec) is the length of time for the H.323 session. The accepted range is from **1** to **3600** seconds (default is **10**). Only available when Make Call is selected as call mode.

Destination Information

➤ **Destination Phone** specifies the destination of the H.323 call. Select the destination phone and its value (a maximum 64 characters are allowed). If gatekeeper details are present, select H.323 ID or E.164 ID otherwise the call will fail.

   ➤ **H.323 ID** specifies the call destination's H.323 ID address. The H.323 ID is a dialing attribute of an H.323 endpoint. For example, if the H.323 endpoint is a softphone, then the H.323 Id will be H.323 extension or H.323 phone number' of the H.323 softphone.

   ➤ **E.164** specifies the destination phone number. The E.164 is an International Public Telecommunications Numbering Plan. The E.164 ID is a dialing attribute of E.164 or an E.164 extension of H.323 endpoint.

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1. Not supported on FTB-2 and FTB-2 Pro.

➤ **Transport Address** specifies the call destination's transport address. Enter the IP address with Q.931 TCP server port number or DNS name with Q.931 TCP server port number of the called H.323 terminal. The TCP server port number is optional. The format used for destination address transport is <IP address or DNS name>[:port]. The default port number is 1720. For example 192.168.1.150 is equivalent to 192.168.1.150:1720, and yourname.com is equivalent to yourname.com:1720.

Utilities

➤ **Traceroute** check box when selected (default) calculates the performance of the network path between two endpoints of a running test every time the test is run.

➤ **Capture** check box when selected (cleared by default) captures the RTP packet stream for the running test. Once test is completed, the capture file is available and can be opened in the Wireshark application from the **Load/Save** option; refer to *Packet Capture* on page 87 for more information.

➤ **Record** check box when selected (cleared by default) allows recording network VoIP traffic. The recorded file can be listened from the **Load/Save** option; refer to *Record Audio Tab* on page 87 for more information. **Record** is only available for **Live Call** media type.

Capture Filters

Available when the **Utilities** - **Capture** check box is selected.

➤ **IP Address** check box when selected (cleared by default) represents the Peer **IP Address** that acts as a filter that will capture packets having the source or destination IP address matching the specified Peer IP address. The accepted range is from **0.0.0.1** to **223.225.225.225**.

➤ **File Size** represents the maximum size of the file to capture either in megabytes (**1** to **50 MB** (default is 5 MB)) or in seconds (**1** (default) to **3600** seconds).

Configuration (H.248/Megaco)

From the **Main Menu**, tap **H.248/Megaco** and the **Configuration** tab.

Call Mode

➤ **Make Call** (default): The Caller establishes a control communications channel with the Receiver. By default, Make Call is selected.

➤ **Answer Call**: It is the endpoint, which initiates its own stream to the Caller test set.

Test Length

Select the test length.

➤ **Media Duration (sec)** is the length of time the test will run in seconds: **1** to **600 seconds** (default is **10 seconds**).

➤ **Number of Packets** is the number of RTP packets to exchange between the controller and the responder: **2** to **65535** (default is **500**).

Call Information

➤ **Media Type**

   ➤ **Live Call** (default) specifies that actual call is established between the Caller and the Callee.

   ➤ **Synthesized Media** specifies that the EXpert VoIP Test Tools application automatically generates a randomly RTP data.

➤ **Codec** selects the audio Codec (packet interval, payload size, and IP bandwidth) used to transmit audio data within the RTP stream.

For **Live Call** media type: **G.711 μlaw** (default), and **G.711 aLaw**.

For **Synthesized Media** type: **G.711 μlaw** (default), **G.711 aLaw**, **G.722**, **G.723.1**, **G.726**, **G.728**, **G.729**, **G.729A**, **G.729B**, and **G.729AB**.

➤ **H.248 DSCP**[1] (H.248 Differentiated Services Code Point) describe a set of QoS (Quality of service) capabilities. The accepted range is from **0** (default) to **63**.

➤ **IP Media DSCP**[1] (IP Media Differentiated Services Code Point) describes a set of end-to-end QoS (Quality of Service) capabilities. End-to-end QoS is the ability of the network to deliver service required by the specific network traffic from one end of the network to the other end. The accepted range is from **0** (default) to **63**.

Destination Information

Destination Phone is the phone number of the Callee: **1** to **99999999999999**. Only available when **Make Call** is selected as the call mode.

Utilities

➤ **Capture** check box when selected (cleared by default) captures the RTP packet stream for the running test. Once test is completed, the capture file is available and can be opened in the Wireshark application from the **Load/Save** option; refer to *Packet Capture* on page 87 for more information.

➤ **Record** check box when selected (cleared by default) allows recording network VoIP traffic. The recorded file can be listened from the **Load/Save** option; refer to *Record Audio Tab* on page 87 for more information. **Record** is only available for **Live Call** media type.

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1. Not supported on FTB-2 and FTB-2 Pro.

Capture Filters

Available when the **Utilities - Capture** check box is selected.

➤ **IP Address** check box when selected (cleared by default) represents the Peer **IP Address** that acts as a filter that will capture packets having the source or destination IP address matching the specified Peer IP address. The accepted range is from **0.0.0.1** to **223.225.225.225**.

➤ **File Size** represents the maximum size of the file to capture either in megabytes (**1** to **50 MB** (default is 5 MB)) or in seconds (**1** (default) to **3600** seconds).

Controller Setup

Note: *Only available when **Controller** is selected as the call mode (see Call Mode on page 26).*

From the **Main Menu**, tap **SIP** and the **Controller Setup** tab.

Controller Parameters

➤ **User Name** is the name used for the SIP URL associated with the Controller. A maximum 50 characters are allowed.

➤ **Signaling Port** is the signaling port of the Controller: **1000** to **65535** (default is 5060).

➤ **Domain Name** is the DNS name of the Controller, specified in either DNS name format or IP address format. A maximum 60 characters are allowed. If the DNS domain or IP address is specified, this value is used as the host portion of the caller's SIP URL, otherwise the IP address of the caller is used.

➤ **Media Mode** defines the flow direction of the RTP packets (media).

  ➤ **Send and Receive** (default) specifies that the RTP packets flows in both directions. For example, if Controller is set in Send and receive media mode then it will send RTP Packets to Responder and also it will receive the RTP packets from Responder.

  ➤ **Receive Only** specifies that the RTP packets flows only in one direction, i.e. from Responder to Controller.

Proxy Settings

➤ **Force Proxy**

If the **Force Proxy** check box is selected, the test routes all requests associated with each call through the specified proxy server. The test ignores any Record-Route or contact information contained in the final response to the initial "SIP invite" request method.

If the **Force Proxy** check box is cleared (default), the test routes subsequent requests in one of the following ways.

➤ Using the Record-Route path, if specified

➤ Directly to the peer Responder via the contact information

➤ **Proxy IP Address/DNS: Port** is the proxy IP address or DNS. IP address can be in a dotted decimal notation (for example, 192.168.34.5) or in a DNS name (for example, cm.acme.com) format. A maximum 60 characters are allowed.

➤ **Authentication User Name** is the user name for the authentication for the Controller, if the proxy server or registration server requires authentication. A maximum 50 characters are allowed.

➤ **Authentication Password** is the password for the authentication for the Controller, if the proxy server or registration server requires authentication. A maximum 64 characters are allowed.

Registration Parameters

Note: Available when **Other Device** is selected as Peer Device (see *Peer Device* on page 26).

➤ **Specify Registrar** specifies SIP Registrar or Registration-Server. When the Specify Registrar check box is selected (cleared by default), SIP registration requests are accepted and processed by the specified SIP Registrar.

➤ **Registrar IP Address/DNS: Port**: Option for the Controller to attempt to register itself every 60 minutes within the specified registrar domain using the specified proxy server. If selected Call Type under Configuration tab is **SIP Call After Proxy Registration** or S**IP Proxy Registration**, the Controller attempts to register itself within the specified registrar domain using the specified proxy server. If this parameter is not provided, it defaults to the Proxy IP address. A maximum 60 characters are allowed.

Responder Setup

Note: *Not available when both **EXFO Device** (see Peer Device on page 26) and **Responder** (see Call Mode on page 26) are selected.*

From the **Main Menu**, tap **SIP** and the **Responder Setup** tab.

Responder Parameters

➤ **User Name** is the name of the Responder, which is used for the SIP URL associated with the Responder. A maximum of 50 characters are allowed.

➤ **Domain Name** is the DNS name used as the host portion of the Responder's SIP URL. A maximum of 60 characters are allowed.

➤ **Signaling Port** is the signaling port of the Responder: **1000** to **65535** (default is **5060**).

➤ **Media Mode** defines the flow direction of the RTP packets (media).

  ➤ **Send and Receive** (default) specifies that the RTP packets flows in both directions. For example, if Controller is set in Send and receive media mode then it will send RTP Packets to Responder and also it will receive the RTP packets from Responder.

  ➤ **Receive Only** specifies that the RTP packets flows only in one direction, i.e. from Responder to Controller.

Proxy Settings

➤ **Force Proxy**

If the **Force Proxy** check box is selected, the test routes all requests associated with each call through the specified proxy server. The Responder ignores any Record-Route or contact information contained in the final response to the initial "SIP invite" request method.

If the **Force Proxy** check box is cleared (default), the test routes subsequent requests in one of the following ways.

➤ Using the Record-Route path, if specified

➤ Directly to the peer endpoint via the contact information

➤ **Proxy IP Address/DNS: Port** is the proxy IP address or DNS. IP address can be in a dotted decimal notation (for example, 192.168.34.5) or in a DNS name (for example, cm.acme.com) format. A maximum 60 characters are allowed.

➤ **Authentication User Name** is the user name for the authentication for the Controller, if the proxy server or registration server requires authentication. A maximum 50 characters are allowed.

➤ **Authentication Password** is the password for the authentication for the Responder, if the proxy server or registration server requires authentication. A maximum 64 characters are allowed.

Registration Parameters

Note: Available when **Other Device** is selected as Peer Device (see *Peer Device* on page 26).

➤ **Register** check box when selected (cleared by default) specifies that the SIP-Responder will register with a SIP Proxy (or SIP Registrar -if specified) during every test run.

➤ **Specify Registrar** specifies SIP Registrar or Registration-Server. When the Specify Registrar check box is selected (cleared by default), SIP registration requests are accepted and processed by the specified SIP Registrar.

➤ **Registrar IP Address/ DNS: Port**: Enter the IP Address to register. If enabled, the Responder attempts to register itself within the specified registrar domain using the specified proxy server. If disabled, the Responder does not attempt to register itself within the specified registrar domain using the specified proxy server. When the Registrar IP Address/ DNS: Port field is selected for editing using virtual keyboard, the Presets button appears on the screen. Press the Presets button to display the list of previously configured values. Select the desired value from the list, if any. Maximum 60 characters are allowed.

➤ **Registration Interval (Sec):** Enter the proxy registration refresh interval in seconds. When the Registration Interval field is selected for editing using virtual keyboard, the Presets button appears on the screen. Press the Presets button to display the list of previously configured values. Select the desired value from the list, if any. The accepted range is from 1 to 65527 seconds. The default value is 3600 seconds.

Parameters (SCCP)

From the **Main Menu**, tap **SCCP** and the **Parameters** tab.

Call Parameters

➤ **Call Manager IP Address/DNS** allows to enter the IP Address or DNS of the Call Manager. IP address can be in a dotted decimal notation (for example, 192.168.34.5) or in a DNS name (for example, cm.acme.com) format. A maximum of 56 characters are allowed.

➤ **Local MAC Address** allows to enter the MAC Address associated with the Callee as specified in the CallManager registration tables. This need not be the Callee's actual MAC address, nor the address of an actual network device. However, it should have a valid MAC Address format. The accepted range is from **00:00:00:00:00:00** to **FF:FF:FF:FF:FF:FF**. The MAC Address must be unique across all tests using the same CallManager.

Phone Type

Phone Type is the Phone Type that the answering endpoint will emulate: **12 S** (default), **12 SP, 12 SP+, 30 SP+, 30 VIP, Cisco IP Phone 7910**, **Cisco IP Phone 7935, Cisco IP Phone 7940**, and **Cisco IP Phone 7960**.

Media Parameters

Media Mode defines the flow direction of the RTP packets (media).

Send and Receive specifies that the RTP packets will flow in both directions. For example, if Controller is set in Send and receive media mode then it will send (will send RTP as Audio Codec is selected) RTP Packets to Responder and also it will receive the RTP packets from Responder.

Parameters (H.323)

From the **Main Menu**, tap **H.323** and the **Parameters** tab.

Local Parameters

➤ **Gatekeeper IP Address/DNS** allows to enter the IP address, or DNS name, or optionally, the UDP RAS port number of the H.323 Gatekeeper. A maximum of 56 characters are allowed.

➤ **Gatekeeper Port** allows to enter the port number of the Gatekeeper: **1000** to **65535** (default is **1719**).

➤ **Local Phone** allows to select the Local Phone number to use when registering with the Gatekeeper then enter the phone number of the selected phone type. A maximum of 64 characters are allowed.

➤ **H.323 ID**: Select the Local H.323 ID address to use when registering with the Gatekeeper. The H.323 ID is a dialing attribute of an H.323 endpoint. For example, if the H.323 endpoint is a softphone, then the H.323 Id will be H.323 extension or H.323 phone number' of the H.323 softphone.

➤ **E.164 ID**: Select the Local E.164 ID address to use when registering with the Gatekeeper. The E.164 is an International Public Telecommunications Numbering Plan. The E.164 ID is a dialing attribute of E.164 or an E.164 extension of H.323 endpoint.

Note: *Gatekeeper details are mandatory to make a call with E.164 ID or H.323 ID.*

Media Parameters

➤ **Media Mode** defines the flow direction of the RTP packets (media).

➤ **Send and Receive** (default) specifies that the RTP packets flows in both directions. For example, if Controller is set in Send and receive media mode then it will send RTP Packets to Responder and also it will receive the RTP packets from Responder.

➤ **Receive Only** specifies that the RTP packets flows only in one direction, i.e. from Responder to Controller or Controller to Responder.

Parameters (H.248/Megaco)

From the **Main Menu**, tap **H.248/Megaco** and the **Parameters** tab.

Local Parameters

➤ **H.248 End Point Name** allows to enter the H.248 End Point Name. The End Point Name should be registered with the end point location (i.e. IP address) and the phone number present in the Call Agent or Media Gateway Controller. For example, if the End Point Name is registered with the abc/1 and abd/1, then you can enter abc and abd in the H.248 End Point Name field. A maximum of 64 characters are allowed.

➤ **H.248 Gateway IP Address/DNS** allows to enter the H.248 Gateway IP Address/DNS name of the Media Gateway Controller or the Call Agent. The Media Gateway Controller should be configured with the end point name, end point location, and the end point phone number.
A maximum of 60 characters are allowed.

Threshold

The threshold is used to specify the Pass and Fail values for the test.

From the **Main Menu**, tap any test and the **Threshold** tab.

VoIP Pass/Fail Thresholds

➤ **Select All** enables all the threshold parameters.

➤ **Maximum Jitter (ms)**: Enter the maximum Jitter value: **0** (default) to **125** milliseconds. The test fails if the average jitter is greater than the configured value.

➤ **Maximum Latency (ms)**: Enter the Maximum Latency value: **0** (default) to **600** milliseconds. The test fails if the average latency is greater than the configured value.

➤ **Maximum Packet Loss (%)**: Enter the maximum Packet Loss percentage value: **0** (default) to **100** %. The test fails if the average packet loss (%) is greater than the configured value.

➤ **Minimum Conversational MOS**: Enter the minimum Conversational MOS value: **0** (default) to **5**. The Conversational Voice Quality Mean Opinion Score (MOS) for the RTP stream going from the Controller to the Responder, if the thresholds setting are at Controller then the RTP stream will flow from Responder to Controller, and if the thresholds setting are at Responder then the RTP stream will flow from Controller to Responder. The test fails if the average MOS is less than the configured value.

➤ **Minimum User R-Factor**: Enter the minimum User R-Factor value: **0** (default) to **100**. The Conversational Voice Quality R factor for the RTP stream going from the Controller to the Responder, if the thresholds setting are at Controller then the RTP stream will flow from Responder to Controller, and if the thresholds setting are at Responder then the RTP stream will flow from Controller to Responder. The test fails if the average R-Factor is less than the configured value.

6   *Test Results*

The rest **Results** menu offers the following Result Summary tabs depending on the test:

Tabs	Test					Page
	RTP	**SIP**	**SCCP**	**H.323**	**H.248**	
Call Details		X				54
Call Manager Statistics			60	58	56	<---
Call Quality Details			X	X	X	62
Controller Statistics	X	X				65
Responder Statistics	X	X				68
Summary	76	81	79	73	71	<---

Call Details

From the **Tests** menu, tap **Results** and the **Call Details** tab.

Test Results

➤ **Endpoint IP Address** displays the IP Address of the endpoint device.

➤ **Peer URL** displays the URL of the endpoint device.

➤ **Response** is the method and response string of the last SIP message received by the test.

➤ **Response Code** is the SIP Response code that the destination returned during the last SIP transaction.

Controller Results

➤ **No of Packets Sent** indicates the number of packets sent to the responder.

➤ **Round Trip Latency** indicates respectively the length of time in milliseconds for the minimum, maximum, and average round-trip latency.

Call Setup and Termination Time (ms)

➤ **Option Response Time** (ms) indicates the time in milliseconds to complete the "Sip options" request method.

Note: *The Option Response Time result is valid only, if SIP Options Request is selected under Call Type in Call Information in the Configuration tab.*

➤ **Initial Response Time** (ms) indicates the time measured in milliseconds from the initial SIP request message sent by the test until the first response is received.

➤ **Call Termination Time** (ms) indicates the length of time in milliseconds to complete the call termination.

➤ **Registration Time** (ms) indicates the time in milliseconds to complete a SIP "register request" method.

➤ **Authentication Time** (ms) indicates the time in milliseconds to complete the proxy authentication during call initiation.

Note: *The Authentication Time result is valid only, if SIP Proxy Registration Only or SIP Call After Proxy Registration is selected under Call Type in Call Information in the Configuration tab.*

➤ **Post Dial Delay** (ms) indicates the time measured in milliseconds from the transmission of the "SIP invite" request method to the reception of first provisional or final response message from the Responder (180 Ringing, for example).

➤ **Call Setup Time** (ms) indicates the length of time in milliseconds to complete the call initiation from the Controller to the Responder (from the reception of "invite" request method to the reception of "ack" response method).

➤ **Post Pickup Delay** (ms) indicates the post pickup delay measured by the test Responder in milliseconds. This is defined as the difference between the transmission of the SIP OK message and the reception of the first media (RTP) packet from the Controller.

➤ **Media Start Time** (ms) indicates the time in milliseconds from dialing to the reception of the first media packet from the called party.

➤ **Call Terminated By** indicates whether the call was terminated by the Controller or the endpoint.

Call Manager Statistics (H.248/Megaco)

Displays the performance statistics of the Call Manager.

From the **Tests** menu, tap **Results** and the **Call Manager Statistics** tab.

Result Summary Metrics

➤ **Peer Phone Number** is the phone number used by the endpoint in Caller mode.

➤ **Peer IP Address** is the IP address of the peer endpoint.

Note: *RTCP (Real-time Transport Control Protocol) provides out of band statistics and control information for an RTP flow.*

➤ **Peer RTP Port** is the UDP port number used by the peer endpoint for media packet reception. The corresponding UDP port number for RTCP is 1 greater than this value.

➤ **Local RTP Port** is the UDP port number used by the test for media packet reception. The corresponding UDP port number for RTCP is 1 greater than this value.

➤ **Local Name** is the endpoint name used in the Media Gateway Controller.

Call Manager Performance Statistics

➤ **Off Hook Time** (ms) indicates the time in milliseconds between the transmission of the Notify (Off Hook) message and the reception of the CreateConnection (dialtone) message from the Media Gateway Controller.

➤ **Post Dial Delay** (ms) indictes the length of time in milliseconds from the start of the call to either the ringing or connection of the called terminal, whatever is earlier.

➤ **Post Pickup Delay** (ms) is the post pickup delay measured by the Caller in milliseconds. This value is the difference between the transmission of the Off Hook Notification message by the test and the reception of the first media (RTP) packet from the endpoint. A negative value indicates that the media arrived early (audio clipping).

➤ **Total Dial Delay** (ms) is the time in milliseconds to complete the call initiation as seen by the Caller. This is defined as the difference between the transmission of the Off Hook Notification message by the test and the reception of the first On Hook Notification Request from the Call Agent.

➤ **Media Start Time** (ms) is the time in milliseconds from the dialing to the reception of the first media packet from the Callee.

➤ **Call Termination Time** (ms) is the time in milliseconds to complete the call initiation as seen by the Caller. This is defined as the difference between the transmission of the response to the Call Agent's Delete Connection command and the reception of the next Off Hook Notification Request from the Call Agent.

➤ **Number of Notified Entities** is the total number of notified entities (Call Agents) that the test endpoint is connected to during the test.

Call Manager Statistics (H.323)

Displays the performance statistics of the Call Manager.

From the **Tests** menu, tap **Results** and the **Call Manager Statistics** tab.

General Results

➤ **Endpoint Display Information** is provided by the peer endpoint for the call.

➤ **Endpoint Phone Number** is the called number used for this call.

➤ **Receiving Audio Codec** is the Audio Codec used for the inbound audio channel from the H.323 endpoint to the Caller.

➤ **Sending Audio Codec** is the Audio Codec used for the outbound audio channel from the Caller to the H.323 endpoint.

➤ **Registration Failure Reason** is the error message received from the CallManager, if the registration is failed.

Call Manager Performance Statistics

➤ **Post Dial Delay** (ms) is the length of time in milliseconds from the start of the call to either the ringing or connection of the called terminal, whatever is earlier.

➤ **Media Setup Delay** (ms) is the length of time in milliseconds for the media to setup after the call is answered.

➤ **Total Call Setup Delay** (ms) is the aggregate amount of time it takes to setup the call. This statistic is a summation of the following delay components: Gatekeeper Discovery Delay, Gatekeeper Registration Delay, Gatekeeper Admission Delay, Post-Dial Delay, Ring Duration, and Media Setup Delay. This result is measured with microsecond accuracy.

➤ **Ring Duration** (ms) is the length of time in milliseconds for a called terminal to answer the call after it rings.

➤ **Media Duration** (sec) is the length of time in seconds for the duration of the test.

Gatekeeper Results

➤ **Gatekeeper Address** displays the Gatekeeper DNS or IP Address through which the call happens.

➤ **Discovery Delay** (ms) is the length of time in milliseconds for the Gatekeeper Discovery Delay.

➤ **Registration Delay** (ms) is the length of time in milliseconds for the Gatekeeper Registration Delay.

➤ **Admission Delay** (ms) is the length of time in milliseconds for the Gatekeeper call Admission Delay.

➤ **Gatekeeper RAS Port** is the RAS port number of the Gatekeeper that was used.

➤ **Local RAS Port** is the RAS port number of the local endpoint.

Call Manager Statistics (SCCP)

Displays the performance statistics of the Call Manager.

From the **Tests** menu, tap **Results** and the **Call Manager Statistics** tab.

Call Statistics

➤ **Assigned Phone Number** is the phone number assigned to the endpoint (caller) by the Call Manager during registration.

➤ **Peer Phone Number** is the phone number used by the endpoint in Caller mode to reach the peer endpoint.

➤ **Peer Display Information** is the peer endpoint's display information.

➤ **Call Manager IP Address/DNS** is the IP/DNS of Cisco call manager which establish the SCCP call signaling.

➤ **Registration Failure Reason** is the error message received from the CallManager if the registration is failed.

Call Manager Performance (ms)

➤ **Registration Time** is the time required to transmit the registration request to the CallManager and receive the registration acknowledgement from the Call Manager.

➤ **Unregistration Time** is the time required to transmit the unregistration request to the CallManager and receive the unregistration acknowledgement from the CallManager.

➤ **TCP Connect Time** is the time required to establish the connection.

➤ **TCP Disconnect Time** is the time required to close the connection.

➤ **Off Hook Time** is the time in milliseconds between the transmission of the Off Hook Message to the CallManager and the reception of the Call State (off-hook) message.

➤ **Post Dial Delay** is the time in milliseconds between the transmission of the last dialed digit to the CallManager and the reception of the Alerting tone message.

➤ **Post Pickup Delay** is the time in milliseconds between the reception of the SCCP Open Receive Channel "Ack" message and the reception of the first media RTP packet.

Call Quality Details

From the **Tests** menu, tap **Results** and the **Call Quality Details** tab.

MOS / R-Factor

➤ **EXFO Conversational MOS** is the Voice Quality Conversational Mean Opinion Score (MOS) score for the incoming RTP stream.

➤ **User R-Factor** is the Voice Quality R-factor for the incoming RTP stream.

RTP Statistics

➤ **Packet Count** is the number of good packets received from endpoint.

➤ **Lost Packets** is the Number of lost packets in the incoming RTP stream.

➤ **Late Packets** is the number of RTP packets that arrived with a higher jitter than the acceptable jitter input parameter value. The test discards all late packets.

➤ **Early Packets** is the number of packets received and dropped by the local endpoint that were too early and could not be buffered within the jitter buffer.

➤ **Total Lost Packets** is the total number of RTP packets that were early, lost and late. This value is calculated using the jitter buffer length input parameter value.

➤ **Total Lost Packets** (%) is the percentage of total lost packets from the number of correct packets and total lost packets. i.e total lost packets/(packet count + total lost packets)* 100.

➤ **Out-of-Order Packets** is the number of RTP packets that were received out of order. This value is calculated using the jitter buffer length input parameter value.

➤ **Duplicate Packets** is the number of duplicate packets received and dropped by the local endpoint.

➤ **No. of Loss Periods** is the count for the number of periods of lost RTP packets. One period of lost packets equals one or more consecutive packets that were lost during the transmission of the packet stream.

➤ **Loss Period Size** (Min) is the size of the smallest period of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result would return 1.

➤ **Loss Period Size** (Max) is the size of the largest period of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result would return 5.

➤ **Loss Period Size** (Avg) is the average size of periods of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result would return 3.

➤ **Loss Type** is the type of packet loss (random or burst) that occurred in the media stream to the Callee. "None" is displayed, when there are no lost packets.

Degradation Factors

➤ **Codec** (%) is the percentage of MOS score degradation that was due to the codec type in the incoming RTP stream.

➤ **Latency** (%) is the percentage of MOS score degradation that was due to the delay in the incoming and outgoing RTP stream.

➤ **Packet Loss** (%) is the percentage of MOS score degradation that was due to the packet loss in the incoming RTP stream.

Jitter (ms)

Length of time in milliseconds of the minimum, maximum, and average jitter of the incoming RTP packets.

Controller Statistics

Displays the result information for packets that travelled from the responder to the controller.

Note: *The **Controller Statistics** tab is only available when **Responder** is selected under the **Call Mode** from the **Configuration** tab.*

From the **Tests** menu, tap **Results** and the **Controller Statistics** tab.

MOS / R-factor

➤ **EXFO Conversational MOS** is the Voice Quality Conversational Mean Opinion Score (MOS) score for the RTP stream from the Responder to the Controller.

➤ **User R-Factor** is the Voice Quality R-factor for the RTP stream from the Responder to the Controller.

RTP Statistics

➤ **Packet Count** is the number of good packets received at Controller.

➤ **Lost Packets** is the number of lost RTP packets going from the Responder to the Controller.

➤ **Late Packets** is the number of RTP packets that arrived with a higher jitter than the acceptable jitter input parameter value. The test discards all late packets.

➤ **Early Packets** is the number of packets received and dropped by the Controller that were too early and could not be buffered within the jitter buffer.

➤ **Total Lost Packets** is the total number of RTP packets that were either lost, early, or late. This value is calculated using the jitter buffer length input parameter value.

➤ **Total Lost Packets** (%) is the percentage of RTP packets that were either lost or late.

➤ **Out-of-Order Packets** is the number of RTP packets that were received out of order. This value is calculated using the jitter buffer length input parameter value.

➤ **Duplicate Packets** is the number of duplicate test packets received and dropped by the Controller.

➤ **No. of Loss Periods** is the count for the number of periods of lost RTP packets. One period of lost packets equals one or more consecutive packets that were lost during the transmission of the packet stream.

➤ **Loss Period Size** (Min) is the size of the smallest period of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result will return 1.

➤ **Loss Period Size** (Max) is the size of the largest period of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result will return 5.

➤ **Loss Period Size** (Avg) is the average size of periods of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result will return 3.

➤ **Loss Type** is the type of packet loss (random or burst) that occurred in the media stream to the Controller. "None" is displayed, when there are no lost packets.

Degradation Factors

➤ **Codec** (%) is the percentage of MOS score degradation that was due to the codec type in the RTP stream from the Responder to the Controller.

➤ **Latency** (%) is the percentage of MOS score degradation that was due to the delay in the RTP stream from the Responder to the Controller.

➤ **Packet Loss** (%) is the percentage of MOS score degradation that was due to the packet loss in the RTP stream from the Responder to the Controller.

Jitter (ms)

Length of time in milliseconds of the minimum, maximum, and average jitter of the RTP packets traveling from the responder to the controller.

Responder Statistics

Displays the result information for packets that travelled from the controller to the responder.

From the **Tests** menu, tap **Results** and the **Responder Statistics** tab.

MOS / R-factor

➤ **EXFO Conversion MOS** is the Voice Quality Conversational Mean Opinion Score (MOS) score for the RTP stream from the Controller to Responder.

➤ **User R-Factor** is the Voice Quality R-factor for the RTP stream from the Controller to Responder.

RTP Statistics

➤ **Packet Count** is the number of good packets received at Responder.

➤ **Lost Packets** is the number of lost RTP packets going from the Controller to Responder.

➤ **Late Packets** is the number of RTP packets that arrived with a higher jitter than the acceptable Jitter input parameter value. The test discards all late packets.

➤ **Early Packets** is the number of packets received and dropped by the Responder that were too early and could not be buffered within the jitter buffer.

➤ **Total Lost Packets** is the total number of RTP packets that were either lost, early, or late. This value is calculated using the jitter buffer length input parameter value.

➤ **Total Lost Packets** (%) is the percentage of RTP packets that were either lost or late.

➤ **Out-of-Order Packets** is the number of RTP packets that were received out of order. This value is calculated using the jitter buffer length input parameter value.

➤ **Duplicate Packets** is the number of duplicate test packets received and dropped by the Responder.

➤ **No. of Loss Periods** is the count for the number of periods of lost RTP packets. One period of lost packets equals one or more consecutive packets that were lost during the transmission of the packet stream.

➤ **Loss Period Size** (Min) is the size of the smallest period of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result will return 1.

➤ **Loss Period Size** (Max) is the size of the largest period of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result will return 5.

➤ **Loss Period Size** (Avg) is the average size of periods of lost RTP packets. For example, if three periods of loss packets were detected, the first with 3 packets, the second with 5 packets, and the third with 1 packet, this result will return 3.

➤ **Loss Type** is the type of packet loss (random or burst) that occurred in the media stream to the Responder. "None" is displayed, when there are no lost packets.

Degradation Factors

➤ **Codec** (%) is the percentage of MOS score degradation that was due to the codec type in the RTP stream from the Controller to Responder.

➤ **Latency** (%) is the percentage of MOS score degradation that was due to the delay in the RTP stream from the Controller to Responder.

➤ **Packet Loss** (%) is the percentage of MOS score degradation that was due to the packet loss in the RTP stream from the Controller to Responder.

Jitter (ms)

Length of time in milliseconds of the minimum, maximum, and average jitter of the RTP packets traveling from the controller to the responder.

Summary (H.248/Megaco)

From the **Tests** menu, tap **Results** and the **Summary** tab.

➤ **Status** displays the Status of the test. Once the test is completed, the Pass/Fail verdict is displayed.

　➤ **Running**: test is running.

　➤ **Stopped**: test is interrupted; test is stopped before the set time.

　➤ **Completed**: test is completed successfully.

➤ **Message** displays one of the following error messages when the test fails.

Call Agent DNS Lookup Failure.
Call Agent host unreachable.
Call Agent port unreachable.
Call Agent Connection Timeout.
Call Agent Disconnection.
No RTP port available.
Endpoint host unreachable.
Endpoint port unreachable.
Call Agent Returned an Error.
No Response To Dialed Number.
Ring No Answer.
Received Busy Signal.
Received Warning Signal.
Received Reorder Signal.
NCS Message Parsing Error.
Unsupported Media Session.
Phone number not configured.
Internal error.
Network error.
Test results not within the Threshold range.

Voice Quality Summary Metrics

➤ **MOS** is the Mean Opinion Score for the RTP stream (packets).

If **Make Call** is selected as **Call Mode**, in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Callee to Caller.

If **Answer Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Caller to Callee.

➤ **R-Factor**: The Voice Quality R-factor for the RTP stream.

If **Make Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Callee to Caller.

If **Answer Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Caller to Callee.

➤ **Jitter** (ms) is the average length of time in milliseconds of the jitter of the RTP packets of incoming RTP stream.

➤ **Latency** (ms) is the average length of time in milliseconds of the round trip latency for the packets traveling between the endpoints.

➤ **Peer Phone Number** is the phone number used by the endpoint.

➤ **Peer IP Address** is the IP address of the peer endpoint.

➤ **Local Name** is the endpoint name used in the Media Gateway Controller.

➤ **Bytes Sent** is tumber of bytes sent from the Caller to the H.248 endpoint, as reported by the H.248 endpoint. This value is taken from the last RTCP report at the H.248 endpoint.

➤ **Packets Sent** is the number of packets sent to the H.248 endpoint. This value is taken from the last RTCP report at the H.248 endpoint.

➤ **Latency** is the length of time in milliseconds for the minimum and maximum latency.

Summary (H.323)

From the **Tests** menu, tap **Results** and the **Summary** tab.

➤ **Status** displays the Status of the test. Once the test is completed, the Pass/Fail verdict is displayed.

    ➤ **Running**: test is running.

    ➤ **Stopped**: test is interrupted; test is stopped before the set time.

    ➤ **Completed**: test is completed successfully.

➤ **Message** displays one of the following error messages when the test fails.

DNS gatekeeper lookup failed.
Gatekeeper discovery failed.
Gatekeeper registration failed.
DNS called terminal lookup failed.
Gatekeeper refused call.
Dial error.
Line busy.
No answer.
Ring error.
Media setup error.
Media error.
Too many answer mode tests.
Unknown error.
Test results not within the Threshold range.

Voice Quality Summary Metrics

➤ **MOS** is the Mean Opinion Score for the RTP stream (packets).

If **Make Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Callee to Caller.

If **Answer Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Caller to Callee.

➤ **R-Factor**: The Voice Quality R-factor for the RTP stream.

If **Make Call** is selected as **Call Mode**, in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Callee to Caller.

If **Answer Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Caller to Callee.

➤ **Jitter** (ms) is the average length of time in milliseconds of the jitter of the RTP packets of incoming RTP stream.

➤ **Latency** (ms) is the average length of time in milliseconds of the round trip latency for the packets traveling between the endpoints.

➤ **Endpoint Phone Number** is the called number used for this call.

➤ **Packets Sent To Endpoint** is the total number of packets sent to the endpoint.

➤ **Network Bandwidth** (Kbps) is the averaged bandwidth consumed audio RTP media streams flowing from the H.323 endpoint. Includes IP and UDP overhead.

➤ **Media Bandwidth** (Kbps) is the averaged bandwidth consumed by audio RTP media streams flowing from the H.323 endpoint. Does not include IP and UDP overhead.

➤ **Packet Loss** (%) is the percentage of lost, late, and early packets traveling on the network.

If **Make Call** is selected as the **Call Mode** in the **Configuration** tab, then the value calculated is from Callee to Caller.

If **Answer Call** is selected as the **Call Mode** in the Configuration tab, then the value calculated is from Caller to Callee.

➤ **Latency** is the length of time in milliseconds for the minimum and maximum latency.

➤ **Traceroute**

Hops to Responder is the number of hops to the peer endpoint. Calculated only if the Traceroute parameter is enabled. See Trace Route on page 37. Hops to Controller is displayed when **Answer Call** is selected as **Call Mode** in **Configuration** tab.

Summary (RTP)

From the **Tests** menu, tap **Results** and the **Summary** tab.

➤ **Status** displays the Status of the test. Once the test is completed, the Pass/Fail verdict is displayed.

➤ **Running**: test is running.

➤ **Stopped**: test is interrupted; test is stopped before the set time.

➤ **Completed**: test is completed successfully.

➤ **Message** displays one of the following error messages when the test fails.

DNS lookup failed.	**Broken pipe.**
No Response from Responder.	**Operation now in progress.**
Responder unreachable.	**Operation already in progress.**
Test Instance not created on Responder.	**Destination address required.**
Connetion refused by Responder.	**Wrong protcol type for socket.**
Incompatible version of test on running on responder.	**Protocol not available.**
	Protocol not supported.
No UDP port available.	**Address family not supported.**
Responder Verifier stopped the test.	**Address already in use.**
Invalid SrcIPAddr parameter.	**Network is down.**
No destination address received.	**Connection reset by peer.**
Error setting socket async option.	**No buffer space available.**
Error setting socket rcvpack option.	**Socket is already connected.**
Wrong ssrc received.	**Socket is not connected.**
Sequence number error.	**Operation timed out.**
Duplicate sequence number received.	**Connection refused.**
Socket busy.	**Write not completed.**
Invalid argument.	**Previous transmit not done.**
File table overflow.	**Send operation timed out.**
Test results not within the Threshold range.	**Interrupted system call.**
Resource temporarily unavailable.	**No such device or address.**
Operation not supported on socket.	**Bad file number.**
Software caused connection abort.	**Permission denied.**
Can't assign requested address.	**Bad address.**

Voice Quality Summary Metrics

➤ **MOS**: The Mean Opinion Score (MOS) is the value calculated for RTP Stream (packets).

If **Controller** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Responder to Controller.

If **Responder** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Controller to Responder.

➤ **R-Factor**: The Voice Quality R-factor for the RTP stream.

If **Controller** is selected as **Call Mode**, in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Responder to Controller.

If **Responder** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Controller to Responder.

➤ **Jitter** (ms) is the average length of time in milliseconds of the jitter of the RTP packets of incoming and outgoing RTP stream.

➤ **Latency** (ms): Average length of time in milliseconds of the round trip latency for the packets traveling between the endpoints.

➤ **Packet Loss** (%) Result: Percentage of lost, late, and early packets traveling on the network.

If **Controller** is selected as **Call Mode** in the **Configuration** tab, then the value calculated is from Responder to Controller.

If **Responder** is selected as **Call Mode** in the **Configuration** tab, then the value calculated is from Controller to Responder.

➤ **Packet Loss** (%) Threshold: The threshold value set from the threshold configuration page.

➤ **No Packets Sent**: The number of RTP packets that were transmitted to other peer.

Latency

Length of time in milliseconds for the minimum and maximum latency.

Traceroute

➤ **Hops to Controller** is the number of hops from the Responder to the Controller.

➤ **Hops to Responder** is the number of hops from the Controller to the Responder.

Summary (SCCP)

From the **Tests** menu, tap **Results** and the **Summary** tab.

➤ **Status** displays the Status of the test. Once the test is completed, the Pass/Fail verdict is displayed.

➤ **Running**: test is running.

➤ **Stopped**: test is interrupted; test is stopped before the set time.

➤ **Completed**: test is completed successfully.

➤ **Message** displays one of the following error messages when the test fails.

DNS lookup failed.
Host Unreachable.
No Response.
Request Refused.
Registration Failure.
Reset Received.
Line busy.
Reorder.
No RTP port.
Disconnect.
Bad Caller Address.
No Answer.
Test results not within the Threshold range.

Voice Quality Summary Metrics

➤ **MOS** is the Mean Opinion Score for the RTP stream (packets).

If **Make Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Callee to Caller.

If **Answer Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Caller to Callee.

➤ **R-Factor** is the Voice Quality R-factor for the RTP stream.

If **Make Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Callee to Caller.

If **Answer Call** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Caller to Callee.

➤ **Jitter** (ms) is the average length of time in milliseconds of the jitter of the RTP packets of incoming and outgoing RTP stream.

➤ Latency (ms) is the average length of time in milliseconds of the round trip latency for the packets traveling between the endpoints.

➤ **Packet Loss** (%) - **Result** is the percentage of lost, late, and early packets traveling on the network.

➤ **Packet Loss** (%) - **Threshold** is the threshold value set from the threshold configuration page.

If **Make Call** is selected as **Call Mode** in the **Configuration** tab, then the value calculated is from Callee to Caller.

If **Answer Call** is selected as **Call Mode** in the **Configuration** tab, then the value calculated is from Caller to Callee.

➤ **Packet Sent** is the number of Packets transmitted to other peer.

➤ **Latency** is the length of time in milliseconds for the minimum and maximum latency.

Traceroute

Hops to Responder is the number of hops from the Controller to the Responder. **Hops to Controller** is available when **Answer Call** is selected as **Call Mode** in the **Configuration** tab.

Summary (SIP)

From the **Tests** menu, tap **Results** and the **Summary** tab.

➤ **Status** displays the Status of the test. Once the test is completed, the Pass/Fail verdict is displayed.

   ➤ **Running**: test is running.

   ➤ **Stopped**: test is interrupted; test is stopped before the set time.

   ➤ **Completed**: test is completed successfully.

➤ **Message** displays one of the following error messages when the test fails.

Call initiate timeout.	Remote RTP port unreachable.
Call initiate failure.	Remote RTCP port unreachable.
Call terminate timeout.	Options request timeout.
Call terminate failure.	Options request failure.
Registration timeout.	Stream Negotiation Failure.
Registration failure.	Network Error.
Proxy authentication failure.	Invalid Session Description.
Proxy authentication timeout.	No Media was received.
Proxy dns lookup failure.	No Answer.
Proxy host unreachable.	Proxy TLS connect error.
Proxy port unreachable.	No Custom File.
Call redirected by proxy.	Local URL unavailable.
Proxy failure server.	Caller resource unavailable.
Proxy failure global.	Responder resource unavailable.
Parse error.	Test terminated redirect loop detected.
Endpoint DNS lookup failure.	Test redirect to null destination.
Endpoint unreachable.	Test results not within the Threshold range.
Endpoint port unreachable.	
Internal error.	
No RTP port available.	

Voice Quality Summary Metrics

➤ **MOS** is the Mean Opinion Score (MOS) for the audio RTP stream (packets).

If **Controller** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Responder to Controller.

If **Responder** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Controller to Responder.

➤ **R-Factor** is the Voice Quality R-factor for the RTP stream.

If **Controller** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Responder to Controller.

If **Responder** is selected as **Call Mode** in the **Configuration** tab, then the value is calculated for RTP Stream (packets) from Controller to Responder.

➤ **Jitter** (ms) is the average length of time in milliseconds of the jitter of the RTP packets of incoming and outgoing RTP stream.

➤ **Latency** (ms) is the average length of time in milliseconds of the round trip latency for the packets traveling between the endpoints.

➤ **Packet Loss** (%) - **Result**: Percentage of lost, late, and early packets traveling on the network.

If **Controller** is selected as **Call Mode** in the **Configuration** tab, then the value calculated is from Responder to Controller.

If **Responder** is selected as **Call Mode** in the **Configuration** tab, then the value calculated is from Controller to Responder.

➤ **Packet Loss** (%) - **Threshold**: The threshold value set from the threshold configuration page.

➤ **Packet Sent** is the number of Packets transmitted to other peer.

➤ **Latency** is the length of time in milliseconds for the minimum and maximum latency.

7 *Test Control*

This chapter describes the test control buttons available on the top-right navigation bar of the application.

Start/Stop Button

The **Start/Stop** button allows to manually start or stop the test.

➤ **Start**: Tap the **Start** button to start the test. **Start** is available when the test is not running.

➤ **Stop**: Tap the **Stop** button to stop the test. **Stop** is available when the test is running.

Load/Save

Configurations Tab

The **Configuration** tab allows to save, load, and delete a test configuration

Tap the **Load/Save** button, and the **Configuration** tab.

Restore Factory Default restores the configuration parameters of all the tests to factory default setting.

To save a configuration:

1. Select the media where the file will be saved: **Internal Flash** (default) or a removable drive (USB media for example) if present.

2. Type the name of the file to be saved (**File Name**) if needed.

3. Tap on the **Save Config** button.

4. Tap **OK**.

To load a configuration:

1. Select the media from where the file will be loaded: **Internal Flash** (default) or a removable drive (USB media for example) if present.

2. Select the file from the list.

3. Tap the **Load Config** button.

Note: *Configuration file has a limited backward compatibility (Typically the backward compatibility period is one year or three software releases).*

To delete a configuration file:

1. Select the media where the file is located: **Internal Flash** (default) or a removable drive (USB media for example) if present.

2. Select the file from the list.

3. Tap the **Delete Config** button.

4. Tap **Yes** to confirm the deletion.

Record Audio Tab

The Record Audio tab allows to listen or delete an audio file.

Tap the **Save/Load** button, and the **Record Audio** tab.

Delete Audio File button deletes the selected audio file from the list.

Listen Audio File button plays the selected audio file from the list.

Packet Capture

The Packet Capture tab allows to open or delete a captured packet file.

Tap the **Save/Load** button, and the **Packet Capture** tab.

Delete Capture File button deletes the selected packet file from the list.

Open Capture File button opens the selected packet file from the list.

Report

Config/Save Tab

The **Config/Save** tab allows to configure the report parameters and generate/save the report.

Tap the **Report** button, and the **Config/Save** tab.

The report contains all information about the current test including its setup and results.

➤ **Report Content** parameters are used to identify the report and are not mandatory. Up to 50 characters are allowed for each parameter.

   ➤ **Report Header** could be the company name.

   ➤ **Report Title** could be the name of the product, name of test, test number, etc.

   Restore Default reverts all parameters back to the default values.

➤ **Save Report**

   ➤ **File Name** is the name of the report to be generated. By default, the name of the report will contain the name of the test, the date (YY_MM_DD), and time (HH_MM_SS).

   ➤ **Media Selector** is the media where the report file will be saved: **Internal Flash** (default), or **Removable Disk (USB)**. **Removable Disk (USB)** is available only when there is a removable disk/key connected to the EXFO's platform USB port.

   ➤ **Display Report after Saving** check box when selected (default) automatically displays the report once it is generated.

Note: *Once generated, the report can be opened from the Open Tab on page 89.*

   ➤ **Turn on Report Generation Prompt** check box when selected (default) displays a pop-up every time a test case is stopped or completed to ask if a report generation is desired.

➤ **Format** is the file format for the report: **PDF** (default), **Text**, **HTML**, and **CSV**.

➤ **Logo** check box when selected (default) allows to include a logo to the report. Only available with the PDF, and HTML file format. Select the logo picture that will be displayed on the report.

➤ **Save** button generates and saves the report on the selected media.

Open Tab

Report files can be opened from this page.

Tap the **Report** button and the **Open** tab.

To open a saved report:

1. Select the media containing the file to open: **Internal Flash** (default), or **Removable Disk (USB)** if present.

2. Select the report file to open from the list.

3. Tap the **Open** button.

8 *Troubleshooting*

Solving Common Problems

Before calling EXFO's technical support, please read the following common problems that can occur and their respective solution.

Problem	Possible Cause	Solution
SIP call not working.	Signaling port may not be correct.	Check whether signaling port is configured correctly.
SCCP call not working.	SCCP Call Manager may not be available.	Check whether Call Manager is up.
H.323 call not working.	H.323 ID of destination phone may not be correct.	Check whether H.323 Gateway is up. Check whether the destination H.323 Endpoint is registered to Gateway.
H.248/Megaco call not working.	H248 endpoint name may not be correct.	Check whether the H.248 endpoint name is configured correctly.

Contacting the Technical Support Group

To obtain after-sales service or technical support for this product, contact EXFO at one of the following numbers. The Technical Support Group is available to take your calls from Monday to Friday, 8:00 a.m. to 7:00 p.m. (Eastern Time in North America).

Technical Support Group

400 Godin Avenue	1 866 683-0155 (USA and Canada)
Quebec (Quebec) G1M 2K2	Tel.: 1 418 683-5498
CANADA	Fax: 1 418 683-9224
	support@exfo.com

For detailed information about technical support, and for a list of other worldwide locations, visit the EXFO Web site at www.exfo.com.

If you have comments or suggestions about this user documentation, you can send them to customer.feedback.manual@exfo.com.

To accelerate the process, please have information such as the name and the serial number (see the product identification label), as well as a description of your problem, close at hand.

A   *Glossary*

Acronym List

?	Help

A

AC	Alternating Current
ARP	Address Resolution Protocol
ASCII	American Standard Code for Information Interchange

B

bps	Bit Per Second

C

CE	European Conformity

D

DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Server

F

FCC	Federal Communications Commission

G

GUI	Graphical User Interface

I

ID	Identification
IEEE	Institute of Electrical & Electronics Engineers
IP	Internet Protocol
ISO	International Organization for Standardization

L

LAN	Local Area Network
LED	Light-Emitting Diode

M

m	Minute
m	Meter
MAC	Media Access Control
Mbps	Megabit Per Second
MOS	Mean Opinion Score
ms	millisecond

P

PRACK	Provisional Response ACKnowledgement

Q

QoS	Quality of Service

R

RFC	Request For Comments
RJ-45	Registered Jack 45
RMA	Return Merchandise Authorization
RTP	Real Time Protocol
RX	Receive

S

SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

T

TCP	Transport Control Protocol
TTL	Time To Live
TX	Transmit

U

UAC	User Agent Client
UAS	User Agent Server
UDP	User Data Protocol
URL	Uniform Resource Locator

μs	microsecond
USB	Universal Serial bus

V

VLAN	Virtual Local Area Network
VoIP	Voice over Internet Protocol

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